Proceedings of the 3rd International Conference & Exhibition on

Underwater Acoustic Measurements: Technologies and Results

21st June to 26th June 2009

Nafplion, Peloponnese - Greece

Edited by

John S. Papadakis
&
Leif Bjørnø

Volume I, II & III
Cover drawing: “Sea shell” symbolizing Sound Propagation in the Sea

by Yorgis Androulakis, IACM-FORTH
Preface

The Proceedings at hand, include the papers presented in the 3rd International Conference and Exhibition on Underwater Acoustic Measurements: Technologies and Results, held in Nafplion, Peloponnese, Greece, from 21 to 26 June, 2009.

The 1st and 2nd Conference with the same title were held at the Foundation for Research and Technology Hellas (FORTH) in Heraklion, in 2005 and 2007 respectively.

Early in 2007, when it was clear that the 2nd International Conference and Exhibition on Underwater Acoustic Measurements: Technologies and Results was going to be a success in number of participants and quality of presentations, we have decided that we would continue organizing this conference every two years, in the last week of June, in Greece. During the conference days in June 2007, at the suggestion of many participants, we have decided that we would be moving the venue of the conference in different places in Greece so the participant will also have the opportunity to visit famous sites of ancient culture and civilization. That is the reason why the 3rd International Conference and Exhibition on Underwater Acoustic Measurements: Technologies and Results is held in Nafplion. The venue of the 4th International Conference and Exhibition on Underwater Acoustic Measurements: Technologies and Results will be decided in June 2009 in Nafplion with the help of the participants.

The main goal of this conference is to act as an “umbrella” for a number of more specialized symposia, each organized by a scientist actively involved in one of the “hot topics” of current and future interest in underwater acoustics. Therefore, the “backbone” of this conference, consists of the Structured Sessions, most of them organized as mini-symposium by an eminent scientist with a central position in R & D in a well-defined topic area of underwater acoustics. For the selection of these hot topic areas, and the identification of the key scientists in the on-going research, we have received substantial advice and support from the members of the Scientific Committee of this conference to whom we are very grateful. We are particularly grateful to the Structured Sessions organizers who responded so positively to our invitation, and took it up to themselves to make this goal materialize. Finally, the sessions with contributed papers are an integral and important part of this series of Conferences.

Among the important news at this conference is the creation of a freely accessible database at FORTH. This database comprises all papers printed in these Proceedings together with all papers printed in the Proceedings of the 1st and 2nd Underwater Acoustics Measurements Conference. This database can be reached via the internet. All papers can be accessed on the address: 


Therefore, we recommend that all references to papers presented at this – and the earlier – conferences shall indicate the above address, where the papers can be found.

The local organization has been in the hands of the experienced team of conference organizers at FORTH, and we will use this opportunity again to thank the team of scientists, technicians and secretaries for an extremely well-done job.

Our gratefulness also goes to our Sponsors and Supporters for their willingness to help by lending the name of their organization and for the economical support given to the
conference, without which, this type of international conference hardly would have been possible. We should mention that the preparation and printing of these proceedings is fully financed by ONR and ONRG.

All names on members of the Scientific Committee, the Organizers of Structured Sessions, the Plenary Lecturers, the Sponsors and Supporters and the members of the Local Organizing Committee are given on the very first pages after this preface. We are grateful to them for their contribution to the success of this 3rd conference, and in particular to all the speakers for their great and important work prepared and presented in this conference. The success of the conference depends more than ever on the quality of these presentations.

It is our hope that these Proceedings of the 3rd International Conference on Underwater Acoustic Measurements: Technologies and Results will stimulate the international interest in measurements in all aspects of underwater acoustics, and that they will become a source of inspiration to everybody in the international Underwater Acoustics Community

Nafplion, Peloponnese, June 2009

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Plenary Lecture
HYPOTHESES ON THE ACOUSTICS OF WHALES, DOLPHINS AND PORPOISES IN BUBBLY WATER

Timothy G. Leighton, P. R. White, D. C. Finfer

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Abstract: The use made of acoustics for communication and echolocation by cetaceans is well-known. We are also familiar with the ability of gas bubbles in the ocean to complicate and confound human attempts to achieve these tasks for ourselves. Some cetaceans must deal with bubbles as a result of their location (for example as occurs with those species restricted to coastal regions): others actively generate bubbles to aid their feeding. Data is scarce as to what extent, if any, cetaceans have exploited the acoustical effects of bubbles, or undertake tactics to compensate for their deleterious effects. The absence of data provides a fruitful opportunity for hypothesis. Having evolved over tens of millions of years to cope with the underwater acoustic environment, cetaceans may have developed extraordinary techniques from which we could learn. This paper outlines some of the possible interactions, ranging from the exploitation of acoustics by humpback whales (Megaptera novaeangliae) in bubble nets to trap prey, to techniques by which coastal dolphins (e.g. of the genus Cephalorhynchus) could successfully echolocate in bubbly water. These hypotheses are then used to develop practical sonar technology for use in bubbly waters.

Keywords: Bubble, ondontocete, cetacean, humpback whale, dolphin, sonar
1. INTRODUCTION

Observations of cetaceans in bubbly water have stimulated thought on the extent to which these animals either exploit the potent acoustical properties of bubbles, or compensate for the deleterious effects bubbles might have, when attempting communication or echolocation in bubbly waters. Certainly humans are aware of the deleterious effects bubbles can have on sonar, to the extent where there exists no satisfactory man-made sonar for use in bubbly coastal waters or ship wakes. Some cetaceans must deal with bubbles as a result of their location (for example as occurs with those species restricted to coastal regions): others actively generate bubbles to aid their feeding. Having evolved over tens of millions of years to cope with the underwater acoustic environment, cetaceans may have developed extraordinary techniques from which we could learn. This paper outlines some of the possible interactions, ranging from the exploitation of acoustics by humpback whales (*Megaptera novaeangliae*) in bubble nets to trap prey, to techniques by which coastal dolphins (e.g. of the genus *Cephalorhynchus*) could successfully echolocate in bubbly water. A sonar system is developed, not by mimicking the echolocation signals of ondontocetes, but by asking what system could be designed to overcome the deleterious effects of bubbles on sonar if it must operate in bubbly water. Whether any species of ondocetes use such a system is uncertain.

2. THE BUBBLE NETS OF HUMPBACK WHALES

Several species of cetacean use bubble nets to assist in the catching of prey, including the short-beaked common dolphins (*Delphinus delphis*) and the Bryde’s whale [1]. The most famous bubble nets are those used by humpback whales (*Megaptera novaeangliae*), although the mechanism by which they trap prey has never been conclusively proven. The hypothesis that these nets may be used to generate a ‘wall of sound’ to trap prey was first proposed in 2004 [2, 3]. It had been known for decades that humpback whales, either singly or in groups, sometimes dive deep and then release bubbles to form the walls of a cylinder, the interior of which is relatively bubble-free. The prey are trapped within this cylinder, for unknown reasons, before the whales ‘lunge feed’ on them from below. When the whales form such nets, they emit very loud, ‘trumpeting feeding calls’. Leighton *et al.* [2, 3] showed how a suitable void fraction profile would cause the wall of the cylinder to act as a waveguide, creating a ‘wall of sound’ with a relatively quiet interior at the centre of the cylinder (Figure 1(a)). They hypothesized that any prey which attempted to leave the trap would enter a region where the sound is subjectively loud and furthermore could excite swim bladder resonances [2, 4-6]. In response, the prey would school, and be trapped ready for consumption (the bubble net turning the ‘schooling’ survival response into an anti-survival response). Whilst forming an attractive hypothesis, however, it is clear that the attenuation of the sound by the bubbly water will require considerably more acoustic power to be projected into the net (e.g. using multiple sources) than would be the case were such attenuation not to occur.

The circular geometries modelled by Leighton *et al.* [2, 3] were based on historical photographs (e.g. Figure 1(b)) and the frequent description in the literature of humpback bubble nets as ‘circular’, or as bubble ‘rings’ [7-16]. The authors were then alerted (by Dr Simon Richards) to high-quality photographs showing the development of spiral bubble nets. The authors hypothesized [16, 17] that such nets would allow the formation of a ‘wall of sound’ with greatly reduced problems of bubble attenuation, whereby refraction in the bubbly
layer, and reflection from it during propagation in the bubble-free arm of the spiral, generate a wall of sound (Figure 2).

![Figure 1: (a) Plan view (from [2,3]) of four whales insonifying an annular bubble net (having 20 m mean diameter and a wall width of 4 m). Here the bubbles are driven in stiffness-controlled mode such that the sound speed decreases linearly from 1500 m/s at the walls (i.e. the sound speed in bubble-free water), to 750 m/s at the cloud midline (corresponding to a void fraction there of ~ 0.01%). The rays are coloured blue, and the locations of the inner and outer walls of the net are shown in red. Computed ray paths, where each whale launches 281 rays with an angular extent of 10°, and then refract. (b) Aerial view of a humpback bubble net (photograph by A. Brayton, reproduced from reference [18]; the author has obtained permission from the publisher to use this image but has been unable to contact the photographer).](image1)

![Figure 2: Panels (a) and (b) show photographs (by Tim Voorheis www.gulfofmaineproductions.com, taken in compliance with United States Federal regulations for aerial marine mammal observation) of the formation of a spiral bubble net. Superimposed upon the photograph in (a), schematic ray paths in white show the refractive path in the bubbly arm of the spiral, whilst the yellow rays show the reflective path in the bubble-free arm of the spiral, which reinforces the attenuated sound field in the bubbly water by partial transmission (producing the red ray at A, the pink ray at B, and the orange ray at D). In (c) the spatial features of the net in (b) have been transposed into a ray tracing model (see [16, 17]), with a putative sound speed profile based on Wood’s equation: the region free of sound rays in (c) is coincident with the location in (b) where the whales rise to catch the herded prey.](image2)
Fig. 3: (a) A simple scale model spiral net of 0.3 m outer diameter, with a closed centre, made from expanded polystyrene in water. The base of the spiral is fixed to an upturned aquarium, such that all of the expanded polystyrene except the top 10 cm is submerged. The spiral is 0.6 m tall and a 1.57 m length of expanded polystyrene (of 7 mm thickness) was required to complete the two full revolutions of the spiral. (b) Measured acoustic field in horizontal plane in demonstration spiral bubble net of expanded polystyrene (1:100 scale, so that the Blacknor Technology sound source projected a 375 kHz tone-burst into the open end of the spiral). The white line shows plan view position of spiral. Data only exists for the discrete measurement points shown as black dots: between these the colour indicates an interpolation and so, whilst visually appealing, cannot include the zero-pressure at the spiral wall. Colour scale: rms sound pressure level (dB re 1 Pa) at each measurement location, time-averaged over the entire 2 ms window from the start of one tone-burst signal to the start of the next (these tone-burst signals are characterized by an ~8 μs free-field duration of a 375 kHz basic frequency sinusoid), so that all the reflections within the spiral were included in the calculation. See references [16, 17] for details.

Of course there is a range of possible explanations for why the prey become trapped by the net, and it is possible that different mechanisms work for different species (e.g. an acoustical swim bladder resonance may operate for some fish, whilst for other creatures (such as krill) a tactile or mechanical effect may dominate).

Testing the proposal would require field trials beyond the current (and likely future) means of the authors. In the meantime the evidence to support this proposal is indirect. The location where the whales surface in Figure 2(b) is the location where the sound field amplitude in Figure 2(c) is modelled to be low (and where we might expect prey to congregate), but this may be coincidence. More photographic data, preferably correlated with undersea measurements of the distributions of bubbles and prey, would be welcome (noting that the visual impression of bubble concentration may be dominated by the presence of large bubbles, and underestimate the presence of smaller bubbles which can often have a more potent effect on the sound speed [19]). Record of what proportion of nets are, and are not, made with feeding calls, and whether this correlates with the species of prey trapped in the net, would provide a useful guide to the prey-specific effectiveness of the various mechanisms by which the net might operate. Tank tests can provide provocative measurements of sound fields (Figure 3), but need to be interpreted with care. Indiscriminate bubble generation may place bubbles at the correct location, but use bubbles of the incorrect size (scaled for the insonification frequency, which is in turn scaled for the net size), and so

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**References:**

[16, 17]
provide a bubble net with refractive acoustic properties which differ from those found in the field \[16, 19, 20\]. This is particularly a problem when scaled-down nets are created, and it was to avoid misleading results from this effect that expanded polystyrene was used in the test of Figure 3. This is because it removes the refraction element from the propagation and concentrates on the reflection components. That this then produces a spiral with a quiet centre is not unsurprising, given the geometry of polystyrene \[16\]. However despite the inability to provide conclusive evidence, this hypothesis has proved popular: The authors were last year informed (by S. Robinson, of NPL) that the narration for a National Geographic documentary \[21\] refers to bubble netting in the following terms: “Humpbacks demonstrate a high level of coordination, such as the strategy the use to maximize fish catches. It is called bubble netting. Up to 20 of them work together to harvest a swirling ball of fish. One whale encircles them in a ring of bubbles. Simultaneous high pitched calls from others below may form a wall of concentrated sound within the bubbles, packing the bait ball tighter. At the surface the bubbles and noise enclose the writhing fish. The whales lunge up together, scooping in hundreds of kilos of food in their accordion-like mouths”. Whilst a BBC documentary in 1993 on humpback bubble netting did not mention the acoustics \[22\], a more recent one focused on the calls of the humpbacks during bubble netting \[23\].

Humpback whales are not the only marine mammals to make bubble nets. However, for smaller echolocating mammals, the bubbles present a potential nuisance to feeding not present for a larger mammal which lunge feeds, a topic which is explored in the next section.

3. ECHOLOCATION IN BUBBLY WATER

The attenuation caused by bubbles to the calls of humpback whales was mentioned in the preceding section. However when human sonar is used in bubbly water at the higher frequencies exploited for echolocation by odontocetes, the ability of bubbles to generate clutter can become overwhelming. Video images of dolphins using bubble nets in conjunction with the herding of fish stimulated the deduction that, since the best man-made sonar would not function in such an environment, either the dolphins had such a functionality in their sonar, or they were ‘blinding’ their own sonar during this hunt \[2, 4\]. Given the restrictions which did not allow the authors to make field measurements or undertake experiments with odontocetes, it was proposed that one way of investigating this conundrum was to determine whether it would be possible for a human to devise a sonar which could operate with enhanced efficiency in bubbly water \[2, 4\]. One such solution was proposed, Twin Inverted Pulse Sonar (TWIPS), whereby the signal contains two pulses, one the inverse of the other \[2, 4\]. The pulses could be tonal, chirps, pseudorandom sequences etc., limited only by the requirements to excite nonlinearities in the bubbles, and to have sufficient fidelity that one is the inverted mimic of the other when they reach the bubbles (which places further requirements that the interpulse time not be so short that the pulses overlap, or too long that changes in the environment degrade the mimicry) \[24\]. The success of such a sonar would not of course prove that odontocetes use it, but would open up the possibility that such solutions exist.

TWIPS works in the following manner (see reference \[25\] for details). The echoes of the two pulses are added to form \(P_+\) (which enhances even powered nonlinearities in the scatter and suppresses the odd-powered nonlinearities, including the linear scatter). This \(P_+\) therefore can be used to enhance the scatter from bubbles \[26\]. The echoes of the two pulses are also subtracted one from the other to form \(P_-\) (which enhances odd powered nonlinearities (including the linear components) in the scatter and suppresses the even-
powered nonlinearities). This $P_-$ can be used to suppress some of the bubble scatter, but not all of it. Further enhancement and suppression can be found through ratios of these so-called “TWIPS1 parameters”. Such so-called “TWIPS2” functions therefore feature ratios such as $P_+ / P_-$ and $P_- / P_+$. Although such ratios are susceptible to noise, they provide a number of attractive features:

- Advantages in detection through enhancement of nonlinear scatterers and suppression of linear ones, and vice versa;
- Advantages in classification, since a feature which is strong in $P_+ / P_-$ but disappears in $P_- / P_+$ is likely to be a nonlinear scatterer (e.g. a bubble in sonar applications);
- Advantages in classification, since a feature which is strong in $P_+ / P_-$ but disappears in $P_- / P_+$ is likely to be a linear scatterer (e.g. a solid target in sonar);
- TWIPS2 automatically removes the need for range correction, appropriate application of which depends on knowledge of the environment, specifically whether the scenario is reverberation-limited or noise-limited – TWIPS2 does away with the need to make that decision.

The TWIPS hypothesis was tested through simulation [27, 28] and experimentation in a test tank, where TWIPS has indeed been shown to work [29-32] (Figures 4 and 5).

![Fig. 4: (a) Photograph looking down into the water of an underground water tank, 8 m × 8 m × 5 m deep, in which a rigid frame holds 4 transducers in a Maltese Cross. A target (T) is aligned on the horizontal acoustic axis, 2.00 m from source. Also on the acoustic axis, a hydrophone (P) is placed in front of the source faceplate (the cable to the hydrophone is marked C). The photograph is taken just as hose (H) begins to feed bubbles through a nozzle (G) into otherwise bubble-free water. (b) Photograph from the top of the water column, showing the scaffolding bar at the top of the frame which holds the source. That bar is at a depth in the water of 2.03 m, and its length is 0.8 m.](image-url)
Fig. 5: The output of the TWIPS2 function $P_r^2 / P_r$ for an interpulse time of 100 ms, produced by stacking 100 consecutive echo time histories (see ref. [32] for details). In each of these figures, the target is located between 2.75 and 3.75 ms, and the bubble cloud between 1.5 and 2.5 ms. The echo from the back wall of the tank occurs at around 6.75 ms and of course can also be treated as a secondary target for TWIPS to enhance. Panels (a) and (b) show the case with the target present, and panels (c) and (d) show the case with the target absent. Panels (a) and (c) are produced using standard sonar processing. In panel (b) the same data as for (a) has been reprocessed using TWIPS. In panel (d) the same data as for (c) has been reprocessed using TWIPS.

Given that TWIPS can be made to enhance target detection in bubbly water in a test tank, primarily through clutter reduction, the question remains as to whether odontocetes employ something like this.

As stated earlier, there is no direct evidence for this. The following discussion of the hypothesis can therefore be treated as nothing more than speculation designed to promote discussion. Features of interest include the following:

(i) Some species of odontocete have been observed transmitting at very high source levels [33]. Source levels of 228 dB re 1 μPa @ 1 m peak-to-peak (~126 kPa 0-pk) have been recorded from *Tursiops gilli* (Pacific bottlenose dolphin), *Tursiops truncatus* (Atlantic bottlenose dolphin), *Pseudorca crassidens* (False Killer whale), although these are not members of the shallow-water species which have been identified with the recording of multiple pulses [32]. Furthermore the source of such multiples has not definitively been shown to be the animal’s emission at source, as opposed to surface reflections (although of course TWIPS could function using surface reflections if these resembled an inversion of the direct pulse with sufficient fidelity). Furthermore, the peak frequency of the emission of these three high-amplitude species is, at $\geq$100 kHz [33], higher than would be optimal for generating nonlinearities in an oceanic bubble population [19, 25]. Measurements to date suggest that the peak frequencies are too high, and the source levels too low, to give strong evidence of the likelihood of TWIPS being used by those species for which there have been greater or lesser suggestions of multiplies pulses [32]: *Cephalorhynchus commersonii* (Commerson’s dolphin, 120-134 kHz, 50 Pa 0-pk), *Cephalorhynchus hectori* (Hector’s dolphin, 112-130 kHz, 18 Pa 0-pk), *Neophocaena phocaenoides* (Finless porpoise, 128 kHz, no data on SL), *Phocoena phocoena* (Harbour porpoise, 120-140 kHz, 63 kPa), and *Phocoenoides dalli* (Dall’s porpoise, 120-160 kHz,
158 Pa 0-pk). The main drawback in this assessment is the difficulty in making measurements from creatures using narrow beams, let alone in bubbly water in the wild. As such there is no evidence of twin inverted pulses being generated at sufficiently high amplitudes, let alone at the low kHz frequencies which are optimal for generating nonlinearities in a wide distribution of bubble sizes. However such measurements have not been conducted on wild animals in bubbly water, and it is possible that the animals could adapt for those conditions when faced with them.

(ii) What facility is offered to odontocetes if the animal is sensitive to frequencies greater than twice the upper frequency content of its own echolocation emissions? Whilst a mismatch of this sort can in some animals indicate the requirement to hear environmental dangers (such as the echolocation emission of a predator), for those animals which themselves generate the highest frequencies they are likely to encounter, is the purpose of hearing more than an octave above their maximum emission frequency indicative of the requirement to detect nonlinearities? Whilst careful study of individual animals has produced valuable audiograms [34] (and for example show a harbour porpoise which would have trouble hearing the second harmonic of its peak frequency), the dataset is from those species which emit multiplies pulses is sparse. It would be interesting to process the artificial TWIPS returns through a filter based on such an audiogram, although of course the primary evidence would be the detection in the wild of high amplitude multiple pulses in a bubbly environment.

(iii) Dolphin test tanks can present acoustic environments very different from those found in the wild: the authors are not aware of any published data on whether odontocetes alter or adapt their emissions when their environment contains bubble clutter.

(iv) Whilst the majority of acoustic examinations of odontocetes have focused on free-ranging species such as *Tursiops truncatus*, those species which are restricted to shallow waters [32] may be more appropriate adapted to the acoustics of shallow water environments. Such adaptation may have developed through both evolutionary and cultural means [32].

(v) Tests on the ability of cetaceans to detect phase and second harmonic components would be very interesting. There is evidence that some bats have this capability [35].

Twin pulses have been detected from some odontocetes, and the phase of the second pulse has been shown to be an inverse of the first pulse. This second pulse has been explained away in terms of the second pulse originating from a surface reflection [36]. Whilst possible in specific circumstances, such suggestions should be critically and quantitatively examined against the feasibility of producing the observed fidelity of the second pulse, e.g. in duplicating the amplitude of the first pulse. Indeed the amplitude degradation that has been observed in surface reflections and cunningly exploited to estimate the range to animals [37]. It should be noted that, if twin inverted pulses of identical high amplitudes could be generated at range from a source using surface reflections, they could be used as an effective TWIPS source in exactly the same way as when the source produces the multiples directly (as was done in Fig. 5 for a man-made source, and which is not an unfeasible process given that phase inversion might be expected as a result of reflections off internal air sacs [38]).

4. CONCLUSIONS

The hypothesis that humpback whales enhance the capability of bubble nets by forming a ‘wall of sound’ has not been proven, but has proven to be sufficiently persuasive to be included in television wildlife documentaries. TWIPS has been shown to work in a test tank,
enhancing the detection of a metal target in bubbly water through clutter reduction. TWIPS can be seen as the first stage of clutter reduction, after which other techniques (e.g. target characterization through resonant scattering; SAS or SAR) can be employed, provided that the frequency ranges for these is appropriate for that required to make TWIPS operable in the bubble population under examination. TWIPS can not only enhance detection under appropriate circumstances, but also allows classification, since an item which disappears when the echoes from the two pulses are added, can be identified as a linear scatter. Conventional sonar cannot do this. Furthermore, not only will the TWIPS principle work for a wide range of incident acoustic pulses (chirps, pseudorandom sequences etc.), it will also work for EM signals (Radar, Lidar, THz radiation, Magnetic Resonance Imaging) in order to discriminate between linear and nonlinear scatterers and, furthermore, between those nonlinear targets which scatter the second harmonic and those which scatter the third harmonic. A range of applications can be found by pairing off a given radiation with target field which contains scatters for the various linear and odd and even harmonics. Lidar for example will scatter nonlinearily off combustion products, whilst second harmonic radar imaging has already been investigated. Some technologies (e.g. MRI) already employ multiple pulses, and might be enhanced if attention was paid to the phase of those pulses. Since practical application needs high amplitude signals close to the source and the clutter, bi-static modalities may be preferred in some circumstances (e.g. with satellite or aerial detection systems).

Alternatively acoustic waves could be used in combination with EM signals (e.g. whereby a hand-held or AUV sonar distinguishes the solids from the bubbles, whilst the EM classifies the solids in terms of rocks, metals, or circuitry). Differentiation of the echoes (with the associated conversion between odd and even harmonics) may be used to create further distinguishing methods. TWIPS-like methods offer a range of possibilities, from cryptography and communications (where exploitation of the nonlinearity inherent (or even hidden) in the harmonics of signals could be exploited) or ultrasonic surgery (where the linear scattering from large bubbles can be used to distinguish them from the nonlinear scattering of smaller bubbles, a process which may be important in the ultrasonic treatment of tumours) [24].

The question of whether TWIPS or some other nonlinear technique is used by odontocetes to suppress bubble clutter is unanswered. The authors have proposed two tests, but these have been unfunded: (i) determine if high amplitude twin inverted pulses are generated in nature; (ii) construct a source capable of delivering such signals in the test tank and TWIPS process them after filtering through an audiogram. A third test (examining whether wild animals which habitually encounter bubbly water through bubble netting or shallow-water environments adapt their echolocation signals suppress bubble clutter) would not be legal under UK law. If a conclusion must be drawn from the sparse data currently available, it is that the low signal amplitudes detected to date from creatures associated with multiple pulses in the wild are too low to excite significant bubble nonlinearities in a population covering a wide span of radii, and this would preclude TWIPS from being used by odontocetes.

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REFERENCES


A B Wood Medal Lecture
Using ocean ambient noise for passive acoustic imaging

Karim G. Sabra

Abstract: The random nature of noise and scattered fields tends to suggest limited utility. Indeed, acoustic fields from random sources or scatterers are often considered to be incoherent, but there is some coherence between two sensors that receive signals from the same individual source or scatterer. An estimate of the Green's function (or impulse response) between two points can be obtained from the cross-correlation of ambient noise recorded at these two points. Recent theoretical and experimental studies in ultrasonics, civil engineering, underwater acoustics and seismology have investigated this technique in various environments and frequency ranges. These results provide a means for passive tomography of the ocean environment using only the ambient noise field, without the use of active sources. The coherent wavefronts emerge from a correlation process that accumulates contributions over time from noise sources whose propagation paths pass through both receivers. We will examine the background physics of extracting these coherent structures and present experimental results confirming these theoretical arguments. Further we will present experimental results such as using noise for time synchronization and localization of unconnected acoustic receivers, and for constructing passive tomographic images of the environment.
Structured Session 1

Vector Sensors

Organizer: Tuncay Akal
Characterization of the Near Scattered Acoustic Vector Field

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Abstract: In this study, we investigate the properties of the scattered acoustic vector fields generated by simple geometric objects, including the infinite rigid plate, disk, and sphere. Analytical solutions are derived from acoustic target strength scattering models in the near field region. Of particular interest is the understanding of the characteristics of energy flow of the scattered acoustic vector field in the near to far-field transition region. We utilize the time and space separable instantaneous active and reactive acoustic intensity to investigate the relative phase properties of the scattered field. Numerical results are presented for the near region scattered acoustic vector field of simple objects in both two and three dimensions.
DIRECTION OF ARRIVAL ESTIMATES WITH VECTOR SENSORS: FIRST RESULTS OF AN ATMOSPHERIC INFRASOUND ARRAY IN THE NETHERLANDS

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Abstract: The Royal Netherlands Meteorological Institute has continuously operated an outdoor atmospheric infrasound array containing 37 pairs of particle velocity sensors (Microflown) and 6 pressure sensors in the north of the Netherlands in the fall of 2008. As initial results, we detected transients caused by distant aircrafts and calculated their Direction of Arrival (DOA). A nearby sound-source, probably an agricultural vehicle passing on the nearby road or field, could be tracked. Furthermore, we compare DOA estimates using the amplitudes of the vector components of the particle velocity measured at single stations with those of classical beamforming and discuss the prospects for underwater applications.

Keywords: atmospheric infrasound, Microflown, direction-of-arrival estimate, vector sensor, localization
1. INTRODUCTION

In this paper, we present results of atmospheric infrasound measurements acquired with an array located in the North of the Netherlands. Atmospheric infrasound is being measured as part of the global network of the Comprehensive Nuclear-Test-Ban Treaty (CTBT). Arrays of micro-barometers for measuring infrasound that are currently deployed typically have an aperture in the order of 0.5 to 3 km. Arrays of this size are difficult to realise and maintain.

Recently developed acoustic vector-sensors [1], devices that measure individual components of vector quantities, can provide a solution to this problem. A measurement of the three components of particle velocity at one position namely enables the calculation of the Direction of Arrival (DOA) of waves, even those with wavelengths either much larger or much shorter than the array aperture. Based on theoretical studies [1],[3] it is expected that vector-sensor arrays can be much smaller than conventional arrays while the detection performance and resolution in DOA estimation can be retained.

For the same reason, vector-sensors are of interest to underwater applications. Large arrays are difficult to handle or cannot be deployed from Autonomous Underwater Vehicles (AUVs) where the space is often limited. Unfortunately, there is a limited availability of vector-sensor data for underwater applications. For this reason, the experiences with atmospheric vector sensors are relevant to underwater applications as well. Here, we study the results of an experiment conducted by the Royal Netherlands Meteorological Institute (KNMI) in the framework of the astronomical Low Frequency Array (LOFAR, www.lofar.org). The aim of the experiment is to investigate, among others, the performance of particle velocity sensors for determining the DOA of atmospheric infrasound.

![Fig. 1: A photograph of a measurement station. The Microflown probes are oriented perpendicular and are mounted on the electronic box. The coloured wires lead to the connectors for the signal cables. A probe is 1/2 inch wide.](image)

2. EQUIPMENT

The array consists of 6 pressure sensitive microphones (Infineon, SMM 310) and 72 commercially available particle velocity sensors, called Microflows. [1]. A Microflow consists of two heated parallel wires. Air moving across the wires will cool the wires, changing their electrical resistivity. The up-wind wire will be cooled more than the down-wind wire, causing a measureable difference in electric resistivity. One Microflow measures the flow of air in one direction with a figure of eight sensitivity.
Each measurement station has 2 orthogonally placed Microflows mounted on an electronics box. Fig. 1 is a photograph showing a station without the protective cover. The installation of each station is based on a housing for a thermometer, as used by the KNMI. It consists of a hard-pvc bottom-plate and two ‘saucers’, with a radius of 13 cm, stacked together with a 2 cm gap between them. The lower saucer has a large hole (7 cm radius) to accommodate the setup. In the following, we will use the phrase ‘EW-flown’ to designate the Microflow with its most sensitive direction oriented EW and ‘NS-flown’ for the other, perpendicularly oriented, Microflow.

| Fig. 2: The array lay-out based on the NORESS array. The axes are scaled by the square root of the distance to the centre (R) for visualisation purposes. The radii of the circles are 2.2m, 4.6m, 9.9m, 21.4m, 45.9m, respectively. The symbols indicate which sensors are present (and functioning) at each station. The array consists of 37 stations and has a NORESS-like lay-out [4]. The NORESS-array geometry is based on 4 concentric rings spaced at log-periodic intervals to create many different inter-station distances, which guarantees an optimal performance of the array in terms of the resolution for DOA. The innermost ring, the A-ring consists of 3 elements and a central element. The B-ring has 5 stations, the C-ring has 7 stations and the D-ring has 9 stations. This gives a total of 25 stations. This array has a fifth ring, containing 11 elements, and an additional station close to the central element, which makes a total of 37 stations. The radius of the E-ring is thus 45.9 metres, making full use of the approximately 100 by 100 metres of grass-land available. The array lay-out is shown in Fig. 2.

The recordings are low-pass filtered, i.e. anti-aliased, and subsequently digitized at 200 Hz using the NI-6225 analogue-digital-convertor. The data are stored on disk in 2 minute segments and the off-line processing is done on tapered time-windows of 512 samples.
3. RESULTS
3.1 Initial results

We analysed the pressure data for transients using the Fisher detector. This detector extracts coherent signals out of the continuous recordings, on the basis of their signal-to-noise ratios. As expected, there were large day-night variations; during the night, the main source of noise, that due to wind, is much lower, resulting in many more detections.

From the detections, we selected data from a – presumed - local sound source, to crudely compute the location of the source; we assumed straight paths from the source at ground level to the receivers. Using the Neighbourhood Algorithm [5], we calculated the source position by maximizing the Fisher-value of the recorded signals. This allowed us to track the sound source, probably an agricultural vehicle working the fields. For the remainder of this manuscript we will focus on Direction of Arrival calculations.

Fig. 3: A schematic of a measurement station defining the angles used. The thin, two-headed red arrows show the sensitive direction of the velocity probes. The DOA is the angle of the incoming sound wave with the North.

3.2 DOA estimation using signal amplitudes instead of phase-differences

Fig. 3 is a schematic drawing showing the orientation of the sensors and an incoming sound wave. The angle $\alpha$ is calculated using the formula

$$\alpha = 0.5 \arctan\left(2 \frac{G_{12}}{G_{11} - G_{22}}\right) \mod \left(\frac{\pi}{2}\right),$$

with $G_{12}$ the cross-correlation between the NS and EW flows, $G_{11}$ the auto-correlation of the NS-flown and $G_{22}$ the auto-correlation of the EW-flown. The correlations are calculated over time-windows of 0.3 seconds (60 samples). After choosing $0 < \alpha < \pi/2$ and using the sign of $G_{12}$ the DOA is known modulo ($\pi$). Addition of a pressure sensor will completely remove this ambiguity.
Fig. 4: DOA results. The red arrows all point to the same direction; namely the direction of arrival of the sound with the maximum Fisher value, as calculated using phased beamforming of 2.5 seconds of data. The hollow black arrows point in the direction as calculated from each 0.3 seconds of amplitude data.

Fig. 4 shows the results of processing of 2.56 seconds of selected data (512 samples). Transient detection using the Fisher detector and classic phased array beamforming over the 2.5 seconds resulted in the filled red arrows. These arrows are plotted, originating from every station with two operating Microflows. The hollow black arrows point in the DOA-estimate as derived using formula (1) for each station for all time-windows of 0.3 seconds within the 2.5 seconds time-interval.

The observations reveal the following characteristics: 1) Only for stations with two functioning Microflows can the DOA be calculated. However, some of the sensors had poor signal to noise ratio or were temporarily not functioning. Therefore, some stations do not show a resolved direction (hollow black arrow). Furthermore, when the signals on the EW- and NS-flown were too different; i.e. the absolute value of their normalized cross-correlation was below 0.3, at least one of the channels is too noisy to produce accurate results. Note that this will prevent DOA-estimates close to either N, E, S or W, because then one of signals is very small compared with the other one. 2) There seems to be a lot of variability in the directions, also per station, compared with the phased array beamforming solution. This can be due to local (wind) noise or because of actual (local) changes in the DOA over the 2.5 seconds, over which the beamforming calculated its (average) DOA estimate. 3) Some stations, notably the ones in the SE-corner seem to have a bias. This is probably due to a constant difference in sensitivity of the EW and NS sensors at those stations. It illustrates that the gauging of the instruments should be done accurately.

4. CONCLUSIONS AND UNDERWATER PROSPECTS
As in air, the directional information of (transient) underwater sound can be retrieved from a single station with a vector sensor combined with a pressure sensor; impossible when using only a single hydrophone.

Combining multiple vector-sensors in an array enhances the resolution in the directional beam pattern due to the cardioid response of vector sensors. Theoretical studies indicate that line arrays of directional sensors can have a directivity index approximately 5 dB larger than that of an identical line array of pressure sensors [6]. In addition, it has been shown that estimating the DOA of transients for sound with wavelengths that are either large or small compared with the array aperture is possible. So, an array of vector sensors can distinguish between ambiguous arrival angles, for instance due to spatial aliasing in a coarse array, because a single vector sensor contains information on the DOA.

Furthermore, vector sensors are able to remove the left-right ambiguity of towed arrays when the particle velocity vector-sensor is combined with a pressure sensor.

As a result of these properties, vector sensors are of special interest for applications on an AUV and for distributed sensor networks for passive monitoring. On AUVs, the space is limited. By using vector sensors, resolution in beamforming can be retained using arrays with short apertures. For distributed sensors, the estimation of the DOA is of fundamental importance in order to combine data acquired at different locations.

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GEOACOUSTIC MATCHED-FIELD INVERSION USING A VERTICAL VECTOR SENSOR ARRAY

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Abstract: Vector sensors measure the acoustic pressure and the particle velocity components. This type of sensor has the ability to provide information in both vertical and azimuthal direction allowing increased directivity. These characteristics have been explored by many authors and most of the studies on vector sensors found in literature are related to direction of arrival (DOA) estimation. However, assembled into an array, a Vector Sensor Array (VSA) improves spatial filtering capabilities and can be used with advantage in other applications such as geoacoustic inversion. In this paper it will be shown that a reliable estimation of ocean bottom parameters, such as sediment compressional speed, density and compressional attenuation, can be obtained using high-frequency signals and a small aperture vertical VSA. The introduction of particle velocity on matched-field processing (MFP) techniques is going to be presented. It will be seen how MFP, usually done with acoustic pressure, can be adapted in order to incorporate the three components of the particle velocity. Comparisons between several processors based either in individual particle velocity components or using all the VSA outputs, are made for simulated and experimental data. The quaternion model, which is founded on hypercomplex algebra, thus more appropriate to represent the 4 dimensional VSA data, is also presented in the MFP context. A novel ray tracing model is used to generate field replicas that include both the acoustic pressure and the particle velocity outputs. The data considered herein was acquired by a four element vertical VSA in the 8-14 kHz band, during the Makai Experiment 2005 sea trial, off Kauai I., Hawaii (USA). The results shows that, when the particle velocity is included it can significantly increase the resolution of bottom properties estimation and in some cases a similar result is obtained using only the vertical component of the particle velocity.

Keywords: Vector Sensor, bottom properties estimation, matched-field processing
1. INTRODUCTION

Acoustic vector sensors measure both the acoustic pressure and the three components of particle velocity. Assembled into an array, a Vector Sensor Array (VSA) improves direction of arrival (DOA) estimation and provides information in both vertical and azimuthal directions [1], providing a higher directivity not possible with arrays of scalar hydrophones of same length and same number of sensors [2]. The spatial filtering capabilities give rise to new applications where VSA’s could be used with advantage over scalar hydrophones, which are appearing in different fields like moving platforms (AUV), port and waterway security [3], underwater acoustic tomography and geoacoustic inversion [4, 5].

The geoacoustic inversion and the introduction of the particle velocity on matched-field processing (MFP) inversion techniques is the objective of this paper. MFP, originally proposed for source localization, could be also applied to estimate ocean bottom parameters based in inversion techniques, usually done with acoustic pressure. However, due to the VSA ability to provide directional information and to its spatial filtering capabilities, the use of VSA could be an advantage in MFP. It will be shown that, using a small aperture vertical VSA and high-frequency signals, a reliable estimation of ocean bottom parameters, such as sediment compressional speed, density and compressional attenuation can be achieved. The classical Bartlett estimator will be adapted in order to incorporate the three components of particle velocity. The Bartlett estimators based in individual particle velocity components or acoustic pressure is compared with others based in all vector sensors output information to achieve the bottom estimation. In this context, a quaternion model, that is a four dimensional hypercomplex number system, is also applied to the VSA data and show interesting results.

A novel ray tracing model – TRACE – developed at SiPLAB is introduced to generate field replicas, which include the acoustic pressure and the particle velocity outputs. This model is used to analyze the real data acquired by the VSA during Makai experiment 2005, off Kauai I., Hawaii (USA) [6]. The simulated and experimental results show that when the particle velocity is included it can significantly increase the resolution of bottom properties estimation. The results are in line with those previously obtained with bottom loss deduced by up/down ratio [7].

2. VSA MATCHED-FIELD PROCESSING

In order to test the use of MFP inversion with the VSA, particle velocity components must be modeled. The horizontal ($v_r$) and vertical ($v_z$) particle velocity components can be calculated from sound pressure as the gradient:

\[ v_r = -\frac{i}{\omega \rho} \frac{\partial p}{\partial r} \quad \text{and} \quad v_z = -\frac{i}{\omega \rho} \frac{\partial p}{\partial z}, \]

where $\omega$ is the circular frequency and $\rho$ is the water density. Then, the $v_x$ and $v_y$ components are calculated projecting the horizontal component ($v_r$) in the azimuthal direction of the source estimated in [2].

Thus, for particle velocity field replicas generation, the TRACE model was developed at SiPLAB [8]. The TRACE model is a ray tracing model, designed to perform two-dimensional acoustic ray tracing in ocean waveguides with flat or variable boundaries, whose properties can be range dependent and can provide different sets of output information like acoustic pressure and particle velocity.

In this study, a half-infinite liquid bottom model with three parameters: sediment compressional speed ($c_p$), density ($\rho$) and attenuation ($\alpha_p$) is used for geoacoustic
inversion. Several data arrangements in Bartlett estimator are tested to perform the bottom parameters matched-field inversion based on vector sensor outputs.

The classical Bartlett estimator can be written as:

$$P_B(f, c_p, \rho, \alpha_p) = \frac{\mathbf{e}^H(f, c_p, \rho, \alpha_p) \cdot \hat{\mathbf{R}}(f) \cdot \mathbf{e}(f, c_p, \rho, \alpha_p)}{\|\mathbf{d}(f)\|^2 \|\mathbf{e}(f, c_p, \rho, \alpha_p)\|^2},$$

(2)

where $\mathbf{d}$ is the acoustic data field received by the four vector sensors at frequency $f$, $\mathbf{e}$ is the model predicted data field, $H$ represents the complex transposition conjugation operator and the correlation matrix estimated for each frequency is given by

$$\hat{\mathbf{R}}(f) = \frac{1}{K} \sum_{k=1}^{K} \mathbf{d}_k(f) \cdot \mathbf{d}_k^H(f),$$

where $K$ is the number of snapshots.

In order to incorporate particle velocity and show the advantage of vector sensors in inverse problems, four different forms to compare data and replica are tested:

1) First, the data $\mathbf{d}$ and replica $\mathbf{e}$ in Eq. (2) are considered as only acoustic pressure or one of the components of particle velocity – named single Bartlett estimators (pressure or individual particle velocity components),

2) VSA estimator – named Bartlett VSA ($p+v$) estimator, considering all components of the four elements VSA. The data and replica vectors are:

$$[p_1, \ldots, p_4, v_{x_1}, \ldots, v_{x_4}, v_{y_1}, \ldots, v_{y_4}, v_{z_1}, \ldots, v_{z_4}]^T,$$

(3)

where $p_n$ is the acoustic pressure and $v_{x_n}$, $v_{y_n}$, and $v_{z_n}$ are the three components of the particle velocity at the $n^{th}$ sensor.

3) VSA estimator – named Bartlett VSA ($v$) estimator, applied to only the particle velocity outputs, where data and replica vectors are:

$$[v_{x_1}, \ldots, v_{x_4}, v_{y_1}, \ldots, v_{y_4}, v_{z_1}, \ldots, v_{z_4}]^T,$$

(4)

and finally

4) Based on quaternion theory – named Bartlett Quaternion estimator, where data and replica vectors are:

$$[q_1, q_2, q_3, q_4],$$

(5)

being $q_n = p_n + v_{x_n}i + v_{y_n}j + v_{z_n}k$ the quaternion at the $n^{th}$ sensor.

Quaternion is a four dimensional hypercomplex number system introduced by Hamilton in 1843 [9], and are an extension of complex numbers to four-dimensional space. A quaternion $\mathbf{q}$ is described by four components (one real and three imaginaries) [10] while here it is considered that the real part is attributed to acoustic pressure and the imaginary parts to the three components of particle velocity.

3. SIMULATION RESULTS

Vector sensor provides the particle velocity fields besides the acoustic pressure, then MFP based on cases 1), 2), 3) and 4) presented in previous section, is applied to invert the sediment compressional speed ($c_p$), density ($\rho$) and compressional attenuation ($\alpha_p$). Simulation results conclude that the field is less sensitive to attenuation and has the higher sensitivity to the sediment compressional speed, therefore $c_p$ and $\rho$ ambiguity surfaces are obtained for different values of $\alpha_p$. To reach the optimal matching and to clearly show the advantage of the MFP based on VSA, the simulated results of Bartlett estimators considering,
single and conjugated VSA estimators, are presented in Fig. 1. The simulations results were obtained for the scenario presented in Fig. 2, with ground truth of the bottom parameters of: $c_p=1575 \text{ m/s}$, $\rho=1.5 \text{ g/cm}^3$ and $\alpha_p=0.6 \text{ dB/} \lambda$, for a frequency of 13078 Hz.

![Fig. 1: Simulations results for frequency 13078 Hz and $\alpha = 0.6 \text{ dB/} \lambda$ for: a) sediment sound speed (with $\rho=1.5 \text{ g/cm}^3$) and b) density (with $c_p=1575 \text{ m/s}$).

The lines for pressure ($p$) and horizontal particle velocity components ($v_x$ and $v_y$) are coincident, Fig 1, since those components mostly depend on low-order modes, thus have little interaction with the bottom. On the other hand, the vertical component ($v_z$) has a higher sensitivity to bottom structure than other components, because it is influenced by high-order modes [4]. The same conclusion is obtained when the VSA Bartlett estimators are compared. The one that considers only particle velocity components has the main lobe narrower with better resolution than the other that includes acoustic pressure. A physical interpretation is that the vertical field carries more information about the bottom structure than pressure and particle velocity is more sensitive to variations to sediment compressional speed than to density.

4. REAL DATA VSA INVERSION RESULTS

A – Experimental setup

![Fig. 2: a) Site bathymetry, with localization of the VSA on September 20th 2005 and the acoustic source TB2 and (b) baseline environment with experiment geometry and measured sound speed profile.]
The data analyzed here were acquired by a four element vertical VSA during 2 hours, on September 20th 2005, within the Makai sea trial that took place from 15 September to 2 October, on the North West coast of Kauai I., Hawai, USA [6]. Fig. 2 a) shows the bathymetry of the area where it can be seen an almost smooth and uniform area of constant depth around 80-100m accompanying the island bathymetric contour surrounded by the continental relatively steep slope to the deeper ocean to the West. Most of the bottom in the area is covered with coral sands over a basalt hard bottom.

The VSA was located at 22.1526 N and -159.7976 W, with a range independent water depth of approximately 104 m and deployed at 79.6 m depth with 10 cm spacing down, between elements. The experimental baseline environment with the sound speed profile is shown in Fig. 2 b). The acoustic source TB2 was bottom moored at 98.2 m depth and 1830 m of range and the signals emitted by TB2 were in the 8-14 kHz band, constituted by LFM's (linear frequency modulated), Multitones and M-sequence. The signal used to invert data was the tone at 13078 Hz.

B – VSA Bartlett estimator

![Fig. 3: Real data ambiguity surfaces for sediment compressional speed during the period of acquisition, for: a) Bartlett vertical component, b) Bartlett VSA (p+v) and c) Bartlett Quaternion.](image)

The MF method based on vector sensor discussed on simulations results is applied to real data, acquired by the VSA. The MF inversion technique was applied considering the sediment sound speed and density for each value of attenuation between 0.1 and 0.9 dB/\(\lambda\), because of its less sensitivity to the MF inversion. To reach the best matching of bottom parameters the attenuation were chosen when the estimator had the maximum value corresponding at 0.6 dB/\(\lambda\). The ambiguity surfaces illustrated in Fig. 3 were obtained for the maximum values of estimator functions along the time showing the stability of the results. Fig. 3 (a) illustrates that the vertical component has a narrow main lobe but for some periods of data acquisition the results are unstable. This may be due to the variability of the water column during the period of acquisition (almost 2 hours) and the results were obtained with a mean sound speed profile. Fig. 3 (c) clearly shows that the estimate of sediment speed is 1580 \(\pm\) 5 m/s, proving that the quaternion model is more appropriate to represent the VSA data than the Bartlett VSA (p+v) estimator, which has a wider main lobe, Fig. 3 (b). These results are in line to those obtained in previous section. For density the uncertain estimation is 1.4 \(\pm\) 0.1 g/cm\(^3\) for the V\(_z\) and Quaternion estimators while for the VSA is 1.4 \(\pm\) 0.2 g/cm\(^3\). One can concluded that the particle velocity information in quaternion model has advantage when bottom parameters estimation based on MF inversion is considered, even at high-frequencies.
5. CONCLUSION

In this paper, the possibility of using a vector sensor array and active signals in the 8-14 kHz band to estimate bottom properties was presented. The proposed geoacoustic inversion based on VSA matched-field shows the advantage of including the particle velocity information in such inverse problems. The classical Bartlett estimator adapted to vector sensor information presented interesting results for bottom parameters estimation and the information given by the vertical particle velocity component and quaternion model provided that sediment compressional speed can be obtained with high-resolution and showed stability along time. The VSA based estimators illustrated that density and attenuation, usually parameters with difficult estimation, presented good sensitivity and the MF based on vector field information can decrease the uncertain estimation of these parameters, demonstrated that the channel signature has sufficient structure in this normally considered high-frequency band for bottom estimation. The particle velocity information enhances the ocean bottom parameters estimation, contributing to a better resolution of these parameters. The usage of VSA with high-frequencies will provide an alternative for a compact and easy to deploy system in various underwater acoustical applications.

6. ACKNOWLEDGEMENTS

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REFERENCES

THE HYDROFLOWN: MEMS-BASED UNDERWATER ACOUSTICAL PARTICLE VELOCITY SENSOR

THE SENSOR, ITS CALIBRATION AND SOME POSSIBLE LOCALIZATION TECHNIQUES

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Abstract: Recent developments in homeland security, harbor and infrastructure protection have increased the interest in vector sensors. The small size, MEMS-based sensors developed by Microflown Technologies BV Inc. are the world’s only commercially available transducers capable of measuring the particle velocity in air (instead of sound pressure). This paper focuses on the development of a new generation, innovative underwater sensor based on the Microflown vector sensors. The technology has a great potential to become a revolutionary underwater acoustic sensor using nanotechnology, and has many applications including autonomous underwater vehicles, underwater acoustic communication systems, floating autonomous systems, seismic towed arrays for underwater oil and mineral prospecting, and harbor and water-side infrastructure protection. The paper will address working principle, calibration methods, sound localization and separation methods and some considerations on these methods.

Keywords: Vector sensor, Hydroflown
1. THE MICROFLOWN SENSOR

The Microflown is an acoustic vector sensor measuring the acoustic particle velocity instead of the acoustic pressure which is measured by conventional microphones, see e.g. [1], [2]. The Microflown sensor measures the velocity of air across two tiny resistive strips of platinum that are heated to about 200°C, (see Fig. 1). A single hot wire (or hot wire anemometer) can also be deployed as a velocity sensor, however the underlying principles of anemometer and Microflown operation are completely different. A single hot wire operates on the cooling down of the wire due to convection, it is based on the measurement of the absolute temperature. It operates from 10 cm/s upwards (in air) for which Kings law applies (the cooling down of the wire is proportional to the square root of the velocity). An anemometer cannot distinguish between positive and negative velocity directions; both will cool down the wire. For lower air velocities (lower than 1 cm/s) the wire will not cool down due to the velocity (other cooling mechanisms become dominant) and Kings law does not apply anymore. Although the wire does not cool down, due to the convection, the temperature distribution around the hot wire will alter.

A Microflown consists of two closely spaced heated wires. It operates in a flow range of 10 nm/s up to about 1 m/s. A first approximation shows no cooling down of the sensors, however particle velocity causes the temperature distribution of both wires to alter. The total temperature distribution causes both wires to differ in temperature. The total temperature distribution is simply the sum of the temperature distributions of the two single wires. Due to the convective heat transfer, the upstream sensor is heated less by the downstream sensor and vice versa. Because of this operation principle, the Microflown can distinguish between positive and negative velocity directions and is much more sensitive than a single hot wire anemometer. Because the sensor measures the temperature difference between the two wires, the sensor is (almost) insensitive to ambient temperature fluctuations.

2. THE HYDRFLOWN SENSOR
The underwater counterpart of the in-air Microflown sensor is termed the Hydroflown sensor. The development of the sensor is in progress. Measurement results are not presented yet. The Microflown and Hydroflown are based on the same working principle, but are designed to operate in different environments.

Under free field conditions the specific acoustic impedance equals the product of the density $\rho$ of the medium and the speed of sound $c$ in the medium. The speed of sound in air is 340m/s and the density of air is 1.3kgm$^{-3}$, the specific acoustic impedance in air equals 440Nsm$^{-3}$ (or Ray). The density of water equals 1000kgm$^{-3}$ and the speed of sound in water 1500m s$^{-1}$. The specific acoustic impedance in water equals 1500kNsm$^{-3}$. The specific acoustic impedance is therefore 3400 times higher in water than in air. The reference sound pressure level that is used for underwater acoustics is 1$\mu$Pa (in air this is 20$\mu$Pa).

<table>
<thead>
<tr>
<th>Speed of sound [m/s]</th>
<th>Air</th>
<th>Water</th>
</tr>
</thead>
<tbody>
<tr>
<td>Density [kgm$^{-3}$]</td>
<td>1.3</td>
<td>1000</td>
</tr>
<tr>
<td>Acoustic Impedance [Ns/m$^{-3}$]</td>
<td>440</td>
<td>1500.000</td>
</tr>
<tr>
<td>Particle velocity due to 1Pa [mm/s]</td>
<td>2.3</td>
<td>0.000667</td>
</tr>
<tr>
<td>Particle displacement 1Pa &amp; 1kHz [nm]</td>
<td>370</td>
<td>0.11</td>
</tr>
<tr>
<td>Specific heat [kJ/kg K]</td>
<td>1</td>
<td>4.18</td>
</tr>
<tr>
<td>Thermal conductivity [W/mK]</td>
<td>0.041</td>
<td>0.6</td>
</tr>
</tbody>
</table>

Since the Microflown is based on a mass flow sensor, the sensitivity is proportional to the product of the density of the fluid and the specific heat. Water is 770 times more dense than air, the specific heat of water is 4.2 times higher than air. The product of the density of the fluid and the specific heat is 3200 times higher in water than in air and the associated particle velocity component of a sound wave in water is 3400 times lower than in air. And therefore the sensitivity of the Microflown related to sound pressure in water is expected almost the same as in air.

2.1. Noise

The noise of an acoustic sensor is an important figure, it limits the measurements for low signals. The lowest sound levels that can be expected are signals that are above the noise level of the sea itself. The dominant noise source that is observed in the sea is generated by the wind. These wind induced noise levels are given by the so-called Knudsen spectra. The lowest level, given as sea state zero is: $\text{Noise}_{\text{re}} = 44 - 17 \log_{10} f + \sqrt{\frac{\text{ref.} \mu \text{Pa}}{\sqrt{\text{Hz}}}}$.

The noise level of a standard Microflown at 1kHz in air is 0dB (ref. 20$\mu$(Pa/$\rho$c)/\sqrt{Hz}).

Motion of submersed objects induce flow noise (for example, towed arrays). The array diameter has a direct correlation with turbulent flow.

The physical dimensions of hydroflown must be different as a standard Microflown element because of the other properties of the medium.

3. CALIBRATION TECHNIQUES

A hydrophone can be calibrated by comparing its output to a reference hydrophone (e.g. in a small water filled chamber). The Hydroflown is a sensor that is designed to be sensitive for
particle velocity. Its sensitivity to particle velocity can be determined with by using a sound pressure sensitive hydrophone if the specific acoustic impedance is known. A free field impedance \( z = \rho c \) is difficult to obtain because of reflections. Apart from that, it is difficult to prove that a sensor is particle velocity of sound pressure sensitive. This is important to be able to prove that the Hydroflown is only sensitive for particle velocity and not for sound pressure.

3.1. Standing wave tube

In a standing wave tube the sound pressure and acoustic particle velocity are related in a relatively simple manner and it is possible if a sensor under test is sound pressure sensitive or particle velocity sensitive. At the liquid surface the sound pressure is (almost) equal to zero and the particle velocity is maximal. If a hydrophone is used as reference it should not be placed at the surface. An air particle velocity sensor can be used as a reference just above the liquid surface. The particle velocity distribution in the tube is given by \( \frac{u_{\text{probe}}}{u_{\text{ref}}} = \cos(kd) \) with \( k \) the wavenumber, \( d \) the depth, \( u_{\text{probe}} \) the probe under test and \( u_{\text{ref}} \) the reference velocity sensor (in air). The sound pressure distribution has a \( \frac{p_{\text{probe}}}{p_{\text{ref}}} = \rho c \cos(kd) \) distribution. It can therefore be proven if a sensor under test is sound pressure sensitive or particle velocity sensitive.

A standing wave tube is quite difficult to build because the walls of the tube must be rigid so that the impedance is much higher than the medium in the tube; this is difficult for a water filled tube.

3.2. Surface boundary technique

In the surface boundary technique a Hydroflown is placed just under the surface and a hydrophone is placed just above the surface. The particle velocity at both sides of the surface is equal. With a calibrated Microflown the sensitivity of the Hydroflown can be calibrated. Once it is proven that the sensor under test is not sensitive for sound pressure, this calibration technique is shown to be more practical. First tests show that it works well up to 1kHz.

The surface boundary technique does not have the condition of a rigid wall (that is required for the standing wave tube technique).

3.3. A tube with two loudspeakers

Another method to find out if a sensor under test is sensitive to sound pressure or particle velocity it is placed in the middle of a tube with two loudspeakers at each side. If the loudspeakers are switched in phase the particle velocity should be zero, if the loudspeakers are switched out-of-phase the sound pressure should be zero. The sensitivity of the device under test cannot be obtained in this set up.

3.4. Moving sensor
It is also possible to move the sensor through a still standing medium. This is not a proper acoustic test but it is important to get a better understanding of the operation principles. If a sound wave passes the sensor under test it will vibrate due to drag with the acoustic disturbance. The sensitivity of a particle velocity sensor will be strongly reduced due to this. With the moving sensor technique this effect is not seen. So it can be used to find the maximal sensitivity.

Fig.1: Left: standing wave tube, middle: surface boundary technique (only the in-air Microflown is seen), right the two loudspeaker setup.

4. VECTOR SENSOR SOURCE LOCALISATION

The in-air Microflown particle velocity sensor is well studied and over a hundred papers have been published in the last 15 years. During these years the near and far field source localisation and surface impedance are studied. Far field source localisation techniques are of primary interest in underwater applications. Some in-air results are summed here. Once the techniques are understood in air, they easily can be transferred into underwater acoustics.

In [4], [5], [7] a technique is presented that localizes a single dominant source in 3D. This is a robust technique that is based on the measurement of 3D intensity. In [7] a helicopter was tracked in 3D at 17Hz using this 3D intensity method.

In [6] and [8] it is shown that it is possible to find two (partial incoherent) sources with a single acoustic vector sensor (AVS, three orthogonal particle velocity and one sound pressure sensor). In [6] it is shown that the maximum number of source that can be located is \(4n-2\), with \(n\) the number of spaced AVS sensors. In this case, source locations were computed using a Music algorithm. In a paper to be published it will be shown that if the source is broad banded, another multi frequency algorithm (PARAFAC) can be used to reach an upper limit of \(8n-2\) sources to be located in 3D. Then 6 sources can be found at maximum with a single AVS sensor.

5. DISCUSSION

Our preliminary experiments show that the Microflown sensor element can measure the acoustic particle velocity in a liquid such as oil. In the following year, the Microflown sensor
design will be optimized to reach a higher signal to noise ratio. At the present, signal processing techniques are tested in air.

Microphones do not have any directionality. Directional systems that are based on microphones make use of a spatial distribution and the directivity is based on phase differences between the sound pressure at the different locations. Because the phase shifts are caused by spatial distribution, the method is depending on the wavelength and thus frequency dependent.

Acoustic vector sensors (AVS) are directional. Because a single AVS measures the sound field in one point, there is limited phase information and the directional information is found in the amplitude responses of the individual particle velocity probes. Benefits of AVS versus arrays of microphones are small size, low data acquisition channel count and no (lower and higher) frequency limit. The bandwidth is limited by the sensors.

If a single source is located with an intensity technique such as done in e.g. [7], flow noise may be reduced. This is because the flow noise can be expected symmetrically spaced around the AVS sensor. In such case, theoretically, the intensity of the flow noise will average to zero.

A spaced array can be combined with a vector sensor. If e.g. a line array is used, the solutions are in a line symmetry. The beamfoming line array produces a slice of the 3D environment and with a cross spectral technique of velocity and sliced pressure, the source in that slice will be found. This may be a solution for the left-right ambiguity.

6. CONCLUSION

It is proven that a standard air type Microflown measures particle velocity in liquids. In the next year this sensor will be optimized for the properties of the medium to improve its self-noise.

Three novel calibration techniques are presented that are able to determine the absolute sensitivity of an acoustic sensor in oil and determine weather the sensor is pressure sensitive or particle velocity sensitive.

Several acoustic vector sensor (AVS) techniques are tested in air. With a single AVS it is proven that two sources in 3D can be separated with a MUSIC algorithm, the theoretic maximum is $4n-2$, with $n$ the number of AVS. In a simulation it is proven that $8n-2$ is the maximum for broad band noise sources.

Flow noise might be reduced in cross spectral based localisation techniques.
7. ACKNOWLEDGEMENTS

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Hydroflown: From a Signal Processing Perspective

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Abstract: Vector sensors offer a more complete description of an acoustic field by measuring both the acoustic pressure and the particle velocity (in contrast to hydrophones which measure the acoustic pressure only). Hence, vector sensor based underwater acoustic systems provide several improvements over conventional systems that incorporate hydrophones. Hydroflown is a novel vector sensor based on the MEMS vector sensors developed by Microflown Technologies BV Inc. which have been successfully implemented in a variety of in-air applications. With this paper, various signal processing issues that arise from adapting the in-air sensor to the underwater environment is investigated in-depth. In addition, several potential applications of underwater vector sensors are presented from a signal processing point of view.
METROLOGICAL ASSURANCE OF THE VECTOR-PHASE
MEASUREMENTS IN HYDROACoustICS

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Abstract: Requirements to the task of necessary accuracy of calibration of through tracts characteristics for hydroacoustic receiving systems are discussed, when the combined receiving systems (CRS) consisting of vector receivers, including complex antennas, are used in water environment. The use of CRS assumes joint processing of signals from the channel of acoustic pressure receiver and three channels of the vector receiver. Thus various characteristics of acoustic field in the same point of space are registered. It imposes rather rigid conditions on determination errors of amplitude-frequency and difference-phase characteristics between channels at their calibration, and also on the frequency and phase characteristics of sound receivers at their graduation. The existing regulated errors for hydrophones and single-channel accelerometers are not always satisfactory for the vector receivers intended for operating in CRS. This paper deals with requirements to errors of amplitude-frequency and difference-phase characteristics for providing for optimum operating of receiving systems consisting of vector receivers for additive or multiplicative processing of signals. It was shown that required precision characteristics for calibration of vector receivers were determined appreciably by algorithms of data processing. For example, if the pressure gradient receiver consists of two mutually spaced hydrophones, task accuracy of amplitude-phase characteristics of through tracts is the highest: it cannot be below 0.1 dB for amplitude and 0.3° for phase. In the case when vector receivers are used for determination of acoustic power flux on background of ocean noises it is possible to limit these accuracies 3-5° u 0.5-0.7 dB. The used methods of calibration of vector receivers in various conditions, which were developed at acoustics department of MSU, are discussed. The approach to tuning in through combined linear acoustic array is given.

Keywords: vector-phase methods, vector receiver, pressure gradient receiver, combined receiving systems, calibration.
1. Introduction

A new research guideline, “the vector-phase methods in acoustics”, was formed at the Acoustics department of Physical faculty of MSU in the beginning of 1960-Th. It was based on simultaneous registration in the given point of space of the field of pressure and its gradient (or oscillation velocity) taking into account phase difference between them. Knowledge of vector characteristics of the sound field gives the considerable extra volume of information, solving the great number of important tasks of modern acoustics. For registration of different characteristics of the acoustic field the combined receiving module (CRM) is used for the same point of space. It includes the sound pressure receiver (omnidirectional hydrophone) and three-component pressure gradient receiver (PGR) with the mutually-orthogonal channels, named a vector receiver (VR) [1]. The feature of such sound receiver is a presence of unified phase center, combined with a mass and geometrical center of the shell receiver. Using the allocated in space CRMs, it is possible to create area-extensive aerial arrays or other constructions named by a combined receiving system (CRS).

2. Calibrating and tuning procedures

The use of CRM (and especially CRS) lays hard enough terms on the error of determination of amplitude-frequency and phase-difference characteristics between channels and also on frequency and phase characteristics of sound receivers at their calibrating. However, SPR and VR register different acoustic characteristics of the sound field and formally must be calibrated in different units of measure. Because of absence of exemplary GPR, calibrating and determination of basic characteristics of GPRs usually is carried out by the method of collation with the exemplary SPR and determining of theoretical dependence between components of the sound field at calibrating, and the sensitiveness of VR channels is presented in sound pressure units of measure. As a result, the error of determination of acoustic wave descriptions by VR will depend also on the environment properties variations. Working with CRS, while calibrating the sound receivers, it is important to include its tuning procedure, with the use of additional coefficients to convert output signals of receiver module to acoustic characteristics of the sound field. It enables to minimize the errors of amplitude- and phase-difference characteristics of through electro-acoustic tracts between any pair of CRS channels. We shall consider the combined receiving system “adjusted” if the gain-phase characteristics of the end-to-end channels of the information transfer of all CRM are identical relative to the plane wave within the limits of the prescribed error.

3. Calibration of the Pressure Gradient Receivers

Because exemplary SPR for calibration VR and theoretical dependence between components of the sound field for the same conditions are used, VR accuracy parameters will be largely determined by the algorithms of data processing. Calibrating of VR can be carried out by the direct comparing to exemplary SPR only when the same plane wave affects both receivers. In this case the interrelation between sound pressure $P$ and oscillation velocity $V$ is expressed by $P/V = \rho c$, where $\rho$ and $c$ – density of environment and speed of sound in water. However in many practical cases we have a complex interference pattern of the sound field, which substantially changes amplitude-phase interrelations between the field components and interrelation between $P$ and $V$ is expressed by $P/V = (\rho c)|\Psi|$, where $\Psi$ - complex coefficient, determining amplitude and phase corrections at calibrating. During the last decades [2] the faculty members proposed a number of methods partly solving these problems, but the main
problem, a receipt of metrology characteristics of VR with the set high enough accuracy, still remain unresolved.

3.1. Vector Receivers calibration based on separately spaced hydrophones

It seems to be the most simple to use a pair of hydrophones spaced by the distance $\Delta$ as exemplary PGR. If every hydrophone is calibrated, it is possible to determine the sensitiveness of such GPR from the equation

$$P_{\Delta} = \frac{P_1 - P_2}{\Delta} = \frac{U_0}{G_p\Delta}, \quad V_0 = \frac{U_0}{\omega p G_p\Delta} = \frac{U_0}{kcpG_p\Delta},$$

where $G_p$ is a sensitiveness of a single hydrophone, $\omega$- circular frequency, $k$ – wave vector, $U_0$ is output voltage of exemplary PGR.

This PGR has fairly narrow frequency range and is very sensitive to variations of the sound pressure receiver (SPR) electro-acoustic parameters. Accuracy of amplitude-phase characteristics setting of through tracts of hydrophones is the most high and hard to achieve and it should not be below 0.1 dB on amplitude and 0.3° on a phase.

3.2. Vector Receivers calibration in not muffled hydropool

In the given conditions it is necessary to know with rather small error distance between the acoustic centers of a source and VR, the type of a source (monopole, dipole), and also to enter amendments for the VR sizes and shape.

3.2.1. Calibration in the near field of a source

It is necessary to conduct measurements in a near field to get rid from multi reflection from the hydropool boundaries. As the “direct” wave considerably exceeds a field of reflected waves in the given conditions, interrelation between sound pressure $P$ and oscillatory speed $V$ is determined by a simple equation:

$$P = \rho c V [1 - (kr_1)^{-2}]^{-1/2},$$

The space position amendment between SPR and VR is calculated by the formula

$$\Delta_r = 10 \log \left( \frac{r_2}{r_1} \right)^2,$$

where $r_1$, $r_2$ are horizontal distances between VR, SPR centers and source centre accordingly. However, in a near field of a source the geometrical and phase centers differ even for the spherical receiver. Therefore, it is necessary to enter the amendment on this distinction [6]. By our estimation the accuracy of above method could not be better than 1.5-2 dB, as a rule. The measurements are recommended to be carried with one-third octave band source of white noise.

3.2.2. Calibration with presence of re-reflected signals

The relation of a square of the module of sound pressure $P$ to the sum of squares of modules of components of oscillatory speed $V_x, V_y$ and $V_z$ at basic frequencies of pool is

$$P^2/(V_x + V_y + V_z)^2 = (\rho c)^2 \varphi^2,$$

where amendment coefficient $\varphi^2 = \varphi (x_1, y_1, z_1, x, y, z, f, a, b, H)$ depends on source coordinates $(x_1, y_1, z_1)$, VR coordinates $(x, y$ and $z)$, on the hydropool sizes of $(a, b, H)$ and on sound signal frequencies of the source. If a sound source is located strictly in the center of hydropool, it is possible to conduct VR calibration by registration of two cuts of a sound field.
with reference SPR and of four cuts with VP graduated channel, the graduated channel is parallel of those two axes on which the field is averaged. The amendment coefficient $\gamma^2$ would be equal to one and the VR calibration more exact with field averaging on three coordinates, but would be considerably more time consuming.

### 3.3. Calibration in a field of a plane standing wave

The no standardized plant UVG-1 intended for calibration of vector receivers and receivers of sound pressure (hydrophones) by an absolute method in a vertically oscillating column of a liquid [3] has been created on acoustics department. The plant includes the working chamber in the form of a vertical pipe piece; the complete set of the acoustic equipment, the block diagram of the plant is presented on Fig. 1.

![Fig.1. The block diagram of plant UVG-1.](image)

The given method of calibration VR is absolute since definition of amplitude of a sound pressure gradient, oscillatory velocity, oscillatory accelerations and other necessary acoustic sizes is carried out theoretically, by recalculation of the measured other physical size with use of known, concrete constants for the given plant, such as density of environment in working
volume and speed of a sound in it. Fig.2 has illustrated enough high accuracy of calibration this plant.

3.4. Vector Receivers Calibration on air in a near field in unadapted rooms

Possibility of the implementation of method [4] should be justified by validity of spherical law of decrease of the sound field in horizontal direction of measurement area. Measuring was carried out by one-third octave band radiation of white noise, as a sound source, similar to monopole, was used. The spherical decrease of the field in the real space was provided in the frequency range 100 – 800 Hz for horizontal distances 15 to 60 cm. For determination of PGR sensitiveness, the levels of electric signal were measured from the proper channel of PGR and exemplary SPR, and theoretical correction on wave sphericity was calculated by taking into account determination of location of the acoustic source center. Errors, caused by the mismatch of acoustic centers of GPR and SPR with the geometrical center of receivers in the near field, did not exceed 1.5 %. At GPR calibrating in the real environment was the total inaccuracy of determination of it’s sensitiveness not higher than $\pm 1.5$ dB, and that of measuring of phase characteristics in the investigated range of frequencies $\pm 100$.

4. Tuning of the Combined Receiving Systems

Most difficulties at tracts tuning occur because of the inaccuracy of sensitiveness determination of sound receivers and phase-difference characteristics of signal through channels. CRS is considered «adjusted», when amplitude-phase characteristics of all CRM are identical within the limits of the set inaccuracy in a plane wave. While solving this problem a mathematical modeling and experimental researches were carried out in the anechoic (free-field) room, and also for the shallow water areas and deep sea.

The conducted analysis proved that influence of imprecision of phases and amplitudes compensation in the separate channels most substantially show up for the additive algorithms of processing, in particular at forming of cardioid directional characteristics. A substantive provision is formulated on this basis, related to CRS tuning: an estimation of tuning quality should be based on the amount of suppression of the determined source by the separate CRM, functioning on $(P+V)$ algorithm, and its noise immunity in relation to the line of noise sources. It should be noted that the indicated requirements to accuracy at the use of VR are substantially weaker, than PGR, which formed by the pair of separate SPR.

The experimental data obtained by us show, that for the CRSs, which have been adjusted by a procedure described above, suppression of the localized source (with SNR over 60 dB on the output of SPR) makes at least 30...35 dB (is actually subject to the quality of VR used). With the source removed, the suppression is mainly subject to surrounding noises. Results of the separate module adjustment based on VR and SPR graduation are notably worse (Fig. 3).
Conclusion

The degree of suppression of the localized source by the separate receiving module based on $P+V$ algorithm and its noise immunity to the line of noise sources is proposed as basis for «quality» estimation of CRS calibrating and tuning. Based on mathematical modeling and experimental researches in the soundproof chamber, for the shallow water areas and deep sea it was shown that:

For the ambient noise suppression the required accuracy of setting the relative variation of SPR and GPR channels should not exceed 0.4 – 0.5 dB for the localized source and 1.0 – 1.5 dB for non-local noise source;

Variation of amplitude-frequency characteristic of tracts should not exceed 0.3 dB on amplitude and 3° – on a phase.

The proposed methods of VR calibration provide the required accuracy of determination of sensitiveness of sound receivers for their use in the CRS receiving modules.

References

USE OF THE SONOGRAPHIC ANALYSIS METHODS WITH THE HIGH RESOLUTION FOR AN INCREASE OF A NOISE STABILITY AND SPACE LOCALIZATION OF THE SEVERAL RADIATING SOURCES THOUGHT THE SINGLE VECTOR SENSOR RECEIVER

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Abstract: Algorithms of a spatial spectrum estimation of the ocean noise by means of the single combined hydroacoustic receiving system (CRS) registering in the small wave sizes area (in a point) the acoustic pressure and projections of its gradient to three mutual-orthogonal directions in space are discussed. It is shown that, despite of “obtuse” enough cosine directional characteristics of the vector channels, due to registration of the acoustic power flux vector, it is possible to receive good (up to degree) the spatial resolution of a local source, and at absence of powerful local sources on water area – the ambient noise spatial spectrum close to real. For reduction of a fluctuating component (usually due to the SNR finite value on an input of CRS) the increase in number of analyzed frequency bands is offered. The same technique in the majority of practical cases allows making a spatial separation of two and more broadband sources of the signal working in crossed frequency bands. For decrease of the analyzed sample volume it is possible to use methods of a digital quadrature low-frequency filtration with decimation, which are usually intended for the preliminary processing signals necessary for realization of algorithms of the spectral and sonographic analysis with the high frequency resolution. For elimination of the signal frequency fluctuations (instability of radiating mechanisms work, a signal propagation fluctuations, Doppler’s effect if the bearings object is moving) it is offered to use the Wigner’s time-frequency transform, or time-frequency transform with use of compensating function of the linearly-frequency-modulated type. Offered decisions are illustrated by results of the experimental researches lead during 2004-2008 on the shallow water areas of Finland Gulf of Baltic Sea and the White Sea.
POTENTIALITIES OF THE COMBINED ACOUSTIC RECEIVERS IN RESPECT TO REGISTRATION OF THE WEAK SIGNALS AND THEIR THRESHOLD LEVEL

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Abstract: In work the basic acoustic characteristics of vector receivers (acoustic pressure gradient receivers) which define their opportunities at registration of weak signals from the noise-reduced or removed marine targets are discussed. Usually at the analysis of such situations two basic characteristics more often consider, namely, sensitivity in free field $G$ which defined by ratio of the idling output voltage $U_x$ to acoustic pressure $P$ in the unperturbed plane wave and threshold level $P_M$. The threshold level usually is considered equal to the signal’s acoustic pressure, at which SNR equals unity in the 1 Hz frequency-band, i.e. to root-mean-square pressure of a sine elastic wave which would create a voltage equal to a voltage on an output of the sensor, generated by sensor’s self-noise. Threshold level $\Delta M$ is the defining acoustic receiver’s characteristic (no sensitivity, as it is often enough believe), when it is applied for registration of the weak signals. Principal cause of existence of the threshold level of the acoustic receiver is thermal noise, because of which on an output of the acoustic receiver random signals are generated, even if any external influences on it are absent. The basic mechanisms of the self-noise field’s generation, determining threshold levels of vector receivers of various designs are analyzed in this work. Situations when one use measurement of an acoustic power flux allows decreasing a threshold level are considered and to register signals, which level less than a threshold level of the separate channel of the vector receiver. The basic received results are illustrated by the data received in natural experiments with real vector receivers of various designs.
STUDY OF ANOMALIES OF THE HIGH-FREQUENCY GEOACoustic Emission Previous Strong EarthQuakes in the Seismoactive zone of Kamchatka

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Abstract: One of mechanisms, which can act as a possible operative harbinger of earthquakes acoustic emission is discussed. On an example of researches of features of geoacoustic emission on Kamchatka in area Avacha Bay and the analysis of its anomalies dated for strong seismic events, it is shown, that there are the abnormal geoacoustic noise caused by pressure, arising at preparation of strong seismic events. Thus high-frequency (4–11 kHz) the range is the most informative at supervision of the cracks formation processes. Work is an integral part of researches on studying features of formation of signals of geoacoustic emission (GAE), previous powerful seismic events with power class KS > 10¹¹ in the seismoactive zone of the peninsula Kamchatka. Novelty of the approach consists that before researches described in given work studying of features of formation of GAE signals was limited from above to frequencies 1–1.5 kHz. Among seismologists there was a firm belief, that more high-frequency range does not represent serious interest because of strong attenuation of GAE signals in a ground. Some statistical regularities of peak-frequency and spatial distributions of GAE signals on an example characteristic such signals previous seismic events with power class KS > 10¹¹ taken place during with 2002 for 2008. Measurement of parameters of signals are discussed was carried out by means of the four-componential combined receiver which allows to register acoustic pressure and three mutual-orthogonal components of its gradient in a wide range of frequencies (10–12000 Hz). The receiving system has been placed at height about 0.5 meters from a bottom of the fresh Lake Mikizha having average depth about 5 meters.
Structured Session 2

Recent developments in Seismic Exploration

Organizer: Gerrit Blacquiere
STATE OF THE ART OF SEISMIC EXPLORATION: AN INTRODUCTION

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Abstract: In this paper an overview will be presented of the state of the art of the seismic echo-acoustic method for the exploration and monitoring of oil and gas reservoirs. The paper is meant as an introduction to the session ‘Recent developments in Seismic Exploration’. Aspects that will be discussed are seismic acquisition both at sea and on land, with amongst others the latest equipment developments and recent acquisition geometries, and seismic data processing with the emphasis on wave-equation methods as opposed to ray-tracing methods. Attention will be paid to the principal problem in seismic exploration, which is to obtain a model of the velocity distribution of the subsurface. Or, in other words: the subsurface image is needed to carry out the processing to obtain the subsurface image. Finally, the Delphi research consortium at Delft University of Technology in The Netherlands will be introduced briefly. This consortium, which already exists for more than 25 years, and which is sponsored by all major oil companies and seismic contractors, has introduced a number of interesting concepts, like the Common Focal Point technology and Surface Related Multiple Elimination. Several of these will be presented in various presentations in the UAM 2009 session on seismic acoustics.

Keywords: Seismic, Exploration, Migration
1. INTRODUCTION

The seismic method is based on the fact that the various geologic structures in the Earth have different acoustic properties like speed of sound and density. As a consequence, these structures reflect acoustic energy. Such energy can be generated by seismic sources at the surface, like dynamite, vibroseis trucks or airguns (at sea). The traveltimes of these reflections, once they have been detected at the surface by geophones (on land) or hydrophones (at sea) contain information on the distance the acoustic waves have travelled.

The principle of the method is very simple. However, since its first practical application in the 1920’s the improvement in image quality has been tremendous. Causes are the introduction of 3D seismic imaging in the 1980’s and even 4D (i.e., repeated 3D) in the 1990’s. Furthermore, the huge increase in the number of recording channels and the corresponding improved spatial sampling has lead to the increased quality. Recent improvements are due to single-sensor recording and to wide-azimuth recording (at sea).

Apart from these improvements in the acquisition of seismic data, also the processing of the data has become much more advanced. This is mainly due to the improvements in computer technology. A major problem of the seismic method, to be tackled in the data processing, is the strongly variable medium to be imaged. Both sound speed and density are subject to large contrasts. This means the acoustic energy changes its direction considerably while propagating into the earth. In order to process the data, and locate all the reflected energy at its correct spatial location, conventionally a proper velocity model is required. However, such a model in itself is one of the results of the processing … Therefore, recently, methods have been developed that do not need a velocity model while still being able to produce proper images.

The Delphi consortium at Delft University of Technology has developed one of these methods, the so call Common Focal Point (CFP) technology. The method aims at retrieving Green’s function from the seismic data. Green’s functions describe the acoustic operators that relate one gridpoint in the subsurface to all points at the surface, i.e., they are one-way operators. This in contrast with the seismic data that contain two-way information: from the surface to the gridpoint in the subsurface and back to the surface again. Going from two-way to one-way simplifies the problem considerably: it is easier to retrieve a proper velocity model from a set of one-way operators than from the seismic data.

In the following sections, the seismic acquisition, the processing, and the Delphi consortium will be discussed in more detail.

2. SEISMIC DATA ACQUISITION

The major improvement in seismic data acquisition that has been realized over the years concerns the amount of data that is acquired in a particular survey and the amount of data that is contained in one shot record (i.e., the number of sensors that are ‘live’ when recording the response of one source). These have increased considerably. As a consequence the aperture could be increased while the spatial sampling distances could be made smaller.

A wider aperture means that the target is ‘illuminated’ by a wider range of angles. Also its response is ‘sensed’ by a wider range of angles. A wider range of ‘illumination-and-sensing’ angles lead to an improved resolution and provides more information on angle- and azimuth-dependent subsurface properties. E.g., it is well-known that reflectivity is an angle-dependent phenomenon – even in the simple case that the reflecting boundary is the interface of two
homogeneous isotropic materials – and in many cases it is azimuth dependent – in the case the properties of the earth materials are anisotropic, which they often are. In addition, the collection of more data has lead to an improved signal to noise ratio. Also processes like velocity model building benefit from larger amounts of data. Typical numbers for current state-of-the-art seismic surveys are (order of magnitude): 100,000 live recording channels, recording 20 s per shot at a temporal sampling interval of 2 ms. A seismic survey may contain many tens of terabytes of data. The author is not aware of any other echo-acoustic measurement system (either in non-destructive testing, or in medical imaging, or in sonar applications) where data is collected with such large numbers of detectors. This makes the seismic imaging the most advanced echo-acoustic imaging technology currently available.

3. SEISMIC DATA PROCESSING

3.1. Introduction

A seismic data processing chain consists of many processes and make takes several weeks to be completed. A good general reference is [1]. Many of these processes aim at improving the signal-to-noise ratio. Another category aims at removing the multiply reflected energy. In particular the water surface is a ‘mirror’ which reflects almost all reflected energy back into the earth. This may cause a train of multiples which may interfere with low-amplitude primary reflections [2,3]. A very important category aims at finding the velocity model, i.e., the value of the speed of sound at every subsurface gridpoint. Such a velocity model is required in the seismic migration process. This process puts all reflected energy at its correct position, i.e., at the subsurface location of the corresponding reflector. The result, the migrated image, is the end product of the seismic method [4]: it is the basis for the geologic interpretation of the subsurface. In practice it is common to define the required velocity model in terms of layers / structures, each with a particular velocity or velocity gradient. However, it is important to realize that this particular choice of a velocity-model representation imposes constraints on the solution that may lead to inaccurate results. It has therefore been proposed to interchange the order of the processing: first image the subsurface and then find the corresponding velocity model. This seems contradictory at first, but it has become possible with the CFP (common focal point) technology developed in the Delphi Consortium at Delft University of Technology [5]. This will now be discussed in more detail.

3.2. CFP technology

3.2.1. Data matrix

Seismic data can be conveniently arranged in the so called data matrix P (the letter p refers to pressure field or potential field) [4]. It has two dimensions, each column corresponds to a particular source coordinate and each row corresponds to a particular detector coordinate. Each matrix element represents a recorded signal (i.e., seismic trace), or, in the temporal frequency domain, a single frequency component. The matrix elements are then complex numbers. The data matrix is visualized in Figure 1.
**Fig 1: Data matrix \( \mathbf{P} \). The columns represent the source coordinate and the rows represent the receiver coordinate.**

If all elements \( \mathbf{P} \) are filled it means that each source response has been detected at all detector positions. This situation is obtained if the detector-array doesn’t move. In practice, this is not often the case. E.g., in marine seismic configuration, the hydrophone arrays are towed behind a vessel. In that case the data matrix contains (many) zero elements. The acquisition configuration determines how the data matrix is filled, e.g., see Figure 2.

**Fig. 2: Data matrix \( \mathbf{P} \). Not that it is not completely filled. For each source there are at most four detectors. \( x_r \) and \( x_s \) represent the receiver and source coordinate respectively.**

Within the data matrix, a column represents a shot gather, a row represents a detector gather, and the diagonal represents zero-offset data, i.e., data obtained by a configuration where the source and the detector are co-located. An example of the zero-offset configuration is an echo sounder where the transducer acts both as a source and as a detector.
3.2.2. Seismic migration as a double focusing process

The seismic migration process (implicitly) carries out a double focusing process: each voxel of the image is obtained by 1. focusing the sources at the location of that voxel and 2. focusing the detectors at the location of that voxel as well. In other words: the sources focus the sound at one point and the detectors sense the echo of that same point. This focusing process corrects for the effects of geometrical spreading as well as for the involved travel-times from the sources or detectors to the focus location. As a consequence, if a reflection is detected at time \( t = 0 \) s in the double-focused result, this means that there is indeed a reflector present at the focus location. The strength of the echo is a measure of the reflectivity. If nothing is detected at time \( t = 0 \) s, there is no reflector present at the focus location, i.e., the reflectivity is zero. This double focusing process is repeated for each voxel of the image, until the complete 3D image has been obtained.

Note that focusing becomes beamforming if the focus is located at infinity or if the focus distance is much larger than the array size (or the length of the track).

A focusing step could be carried out physically, e.g., by summing time-delayed signals from the detectors. The time delays should correspond to the time that is required for the signal to travel from the focus point to the detectors. In a similar way the sources could be physically focused: by applying particular time delays to the sources in an array it is possible to create a wave front that focuses at the desired location. This technique is applied in kidney stone lithotripsy, where focused shock waves are used to break up the stone in smaller parts.

A major disadvantage of ‘physical’ focusing, is that a separate echo-acoustic experiment is required for each voxel of the image. This would be time consuming and expensive.

Fortunately, the focusing process can be carried out off-line in a computer. Now that we have organized the measured data in the data matrix \( P \), it is clear that the two involved focusing steps (one for the sources and one for the detectors) can be written as a vector-matrix operations: focusing of the detectors is a row-vector times the data matrix; focusing of the sources is the data matrix times a column-vector; focusing of the detectors as well as the sources is a row-vector times the data matrix times a column vector (Figure 3). The vectors are the focus operators that contain particular traveltime and amplitude corrections such that a nice focus is created. Note that a traveltime correction may be implemented in the temporal frequency domain as a simple phase shift. Its value is \( \omega \tau \), where \( \tau \) is the traveltime correction to be applied and \( \omega \) is the radial frequency, hence it is a frequency dependent phase shift. The implementation corresponds to a complex multiplication with \( e^{j\omega \tau} \). Note that a more general phase shift, \( e^{j\phi} \), \( \phi \) representing phase, may represent correction for multiple traveltimes, e.g., as required to include the effects of multi-

paths in the focusing.
3.2.3. Common Focal Point technology and its benefits

As mentioned before, there exist many migration algorithms that focus echo-acoustic energy and result in images, using a velocity model of the subsurface. The result of such algorithms is that both the sources and the detectors are focused. What makes the CFP (Common Focal Point) technology different is that the intermediate result is examined as well, i.e., the result of a single focusing process, either at the source side or at the detectors side (Figures 4 and 5), rather than the usual (implicit) double focusing result. Furthermore, no velocity model is required.

The intermediate result, called CFP gather, may be interpreted as follows. When the detectors have been focused, this means that a virtual detector has been created, located at the focus point, at some distance away from the detector array; however, the sources are still at their original positions. Similarly, when the sources have been focused, a virtual source has been created at the focus point, at some distance from the source array, while the detectors are still at the original positions.

Fig. 4: Focusing in detection. The focus operator is applied to the detectors. The result is a CFP gather. In this case a virtual detector has been created at the location of the focus point whereas the sources are still at their original position.
Fig. 5: Focusing in emission. The focus operator is applied to the sources. The result is a CFP gather. In this case a virtual source has been created at the location of the focus point whereas the detectors are still at their original position.

The step that now follows is crucial and very important for appreciating the benefits of the so-called CFP domain.

To carry out the focusing process in a computer, a focus operator is required, i.e., the vector containing the traveltime and amplitude corrections corresponding to the focus location. This operator is computed on the basis of a first guess of the velocity model representing the situation (e.g., water with a constant speed of sound and information on the geometry of the experiment). If this model is correct, then a perfect focus operator can be computed: it correctly describes the right traveltimes and amplitudes from the focus point to the locations of the detectors (or sources). This man-made operator is then applied to the measured data, resulting in a CFP gather. As mentioned, this CFP gather represents the situation with a virtual source at the focus location and the detectors still at their original locations (or vice versa). In the CFP gather the traveltimes from the focus location to the detectors will be visible as an echo-acoustic ‘event’ (provided the focus location has been chosen such that it is on top of a reflector). In conclusion: the traveltimes in the focus operator and the traveltimes of the CFP gather (which is the result of the action of the focus operator on the data) are exactly the same! This is called the ‘principle of equal travel time’

Now assume that the model is not correct (and as it was a first guess, it is likely to be erroneous). In such a case the traveltimes of the focus operator are not correct. If this operator is then applied to the data, e.g., at the source side, a virtual source is created which is not focused well and which is not located exactly at the intended focus position. As a consequence the traveltimes in the CFP gather are different from the traveltimes of the focus operator. Although at first sight this may seem to be a disadvantage, it is the strong property of the CFP method: from the difference between the traveltimes in the CFP gather and the traveltimes in the focus operator it is possible to retrieve corrections that have to be applied to the focus operator such that it improves. In this (iterative) way, the focus operator is optimized and a good-quality, well-focused image is obtained.

One additional remark: after the focus operator has been obtained using the CFP procedure discussed above, we do have a perfect focus operator, but we do not have an improved velocity model (e.g., water with the right velocity profile). This is because the operator is updated directly in the iterative process and not in an indirect way via a model update. This means that although the image is well-focused, it may be spatially distorted because the location of the focus point is unknown. To solve this, in the Delphi project at Delft University of Technology tools are being developed to estimate models on the basis of a set of focus operators via inversion [6].
A second possibility that the CFP domain offers, is to quantify the quality of the focus after the optimum focus operator has been determined. This quality depends on the availability of enough data, i.e., the available source locations (source array configuration), the available detector locations (detector array configuration), the number of ‘life’ detectors for each source, etc. In short: the focus quality depends on how the data matrix has been filled. A badly designed and conducted experiment does never deliver a good result, despite a proper data processing.

4. DELPHI CONSORTIUM AT DELFT UNIVERSITY OF TECHNOLOGY

Much of the university research on the seismic echo acoustic imaging method is organized around so-called consortia. The most well-known consortia are SEP (Stanford Exploration Project), CWP (Centre for Wave Phenomena, Colorado School of Mines) and Delphi (Delft University of Technology). Such consortia are funded by the industry. Typically, the major oil companies, the large national oil companies and geophysical contractors are amongst the sponsors.

The Delphi consortium was founded in the early eighties by Prof. A.J. Berkhout and consists of three interrelated programmes: acquisition & preprocessing, multiple removal & structural imaging, and dynamic reservoir characterization & flow dynamics. Together these programmes cover the complete seismic imaging and characterization flow. The session ‘Recent developments in Seismic Exploration’ is organized by people from the Delphi consortium. More information can be found in [5].

5. ACKNOWLEDGEMENTS

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AN R-BASED OVERVIEW OF THE WRW CONCEPT

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Abstract: The WRW model serves the geoscientists' community as a global language dedicated to deliver a better insight into the seismic reflection experiment. The model formulates such an experiment as a multiplication of matrices. In today's numerical modeling, wave equation-based techniques such as implemented in the WRW model are used more and more, while it becomes clear that ray tracing methods do not have the required accuracy. The reflectivity matrix \( R \) is undoubtedly one of the most essential parts of the WRW model since it contains the angle dependent reflectivity information of the subsurface structures. It is this information that is to be retrieved from the seismic experiment. This paper provides insight in the formation process of this matrix. Different properties of the reflectivity matrix are investigated through the numerical modeling of three different cases. Moreover, the accuracy of the WRW approach is compared to that of ray tracing. The numerical results highlight the superiority of the WRW approach, i.e., the wave theory based approach, particularly in the case of a laterally variant reflector.

Keywords: WRW, reflectivity matrix, forward modeling

1. INTRODUCTION

The main goal of the seismic reflection experiment is to obtain reflectivity information of subsurface structures. This can be done by means of structural imaging and reflectivity inversion processes on the acquired seismic data. Therefore, the forward modeling algorithm has a crucial impact on the inversion output. Among numerical modeling techniques, wave equation based modeling algorithms such as wave field extrapolation, and ray tracing are well
known. The WRW model, as introduced in [1], provides an efficient way to describe and handle mathematically the inversion parameters.

In the ray tracing approach, each ray path represents a seismic wave-front from a source to a receiver [2]. As the ray reaches an interface a part of it is transmitted and the rest is reflected depending only on the local reflectivity properties and the angle of incidence. As a result, the effect of irregularities on the reflectors cannot be properly taken into account [3].

On the other hand, according to the wave field extrapolation approach, wave propagation can be described by the contribution of secondary sources placed along the spherical wave front (Huygens’ Principle). Therefore, in order to obtain the reflectivity information at a certain grid point on the reflective interface, the effect of neighboring grid points is also taken into account. Fig. 1 schematically represents the difference between the two methods. Wave field extrapolation based modeling can be described by means of the WRW model [1].

In this paper the WRW model will be analyzed with special attention given to the reflectivity matrix. 2D numerical examples will be presented and the results obtained from the two methods will be compared. The R matrices of the examples shall give an insight into the reflection mechanism and its impact on the final dataset. Finally, the deficiencies of the ray tracing approach with respect to the WRW approach will be discussed.

2. THE WRW MODEL

The key feature of the WRW model is that it describes the seismic data in terms of matrix operators in the frequency domain [1]. According to this model, each monochromatic component (single frequency) of the primary wave field \( P(z_0,z_0) \) that is recorded at the surface \( z_0 \), can be described in the space-frequency domain by:

\[
P(z_0,z_0) = D(z_0) \sum_{m=1}^{M} [W(z_0,z_m)R(z_m,z_0)]S(z_0).
\]

In eq. 1, \( z_m \) denotes the steps in depth of the algorithm, thus, the depth levels that are investigated for reflective boundaries. The designation ‘WRW model’ stems from the two propagation matrix - operators \( W \), and the matrix operator \( R \), whose functions are explained below. It is worthwhile to mention that the WRW concept serves also as the vehicle for the CFP technology [4]. Fig. 2a shows a schematic representation of the WRW model.

The lateral coordinates \( x \) and \( y \) and the frequency \( \omega \) have been left out for convenience.

![Fig. 1: Different approaches to the reflection problem. (a) Wave theory (b) Ray tracing](image)
The matrix operators in eq. 1 have the following meaning:

- \( S(z_0) \): source matrix. It contains the amplitude and phase of the source wavelet at the frequency under consideration. One column represents one source (array) and determines its position in space.
- \( W(z_m, z_0) \): forward wave field propagation matrix. Each column contains a discrete version of the Green’s function that describes the wave propagation from one point (one lateral location) at the surface \( z_0 \) to many points at depth level \( z_m \).
- \( R(z_m, z_m) \): reflectivity matrix. It describes the conversion of an incident wave field into a reflected wave field, as will be further explained below.
- \( W(z_0, z_m) \): forward wave field propagation matrix. It describes the wave propagation from one point at the depth level \( z_m \) to many points at the surface \( z_0 \).
- \( D(z_0) \): detector matrix. It contains the detector wavelet. One row represents one detector (array) and determines its position in space.

It follows that the element \( P_{ij} \) of the data matrix \( P(z_0, z_0) \) corresponds to the configuration: source at the location \( j \) and detector at the location \( i \). Therefore, one column of the data matrix represents a common source gather (shot record) and one row represents a common receiver gather. Other data gathers, such as CMP gathers or common offset gathers, can also be identified in the data matrix. This equation is valid for stationary acquisition geometries and stationary parts of non-stationary acquisition geometries.

3. REFLECTIVITY MATRIX

In the previous section, the general description of the WRW model was reviewed, in which the reflectivity matrix \( R \) was introduced as a matrix in the space-frequency domain containing operators that convert the incident wave field into the reflected wave field. These operators can be derived from the reflectivity operator in the wavenumber-frequency domain.

In the case of a horizontal reflector between two homogeneous media it is given by [5]:

\[
R(k_x, z_1, z_2, \omega) = \frac{\rho_2 k_{x,1} - \rho_1 k_{x,2}}{\rho_2 k_{x,1} + \rho_1 k_{x,2}},
\]

(2)

\[
k_{x,1,2} = \sqrt{k_{x,2}^2 - (k_{1,2} \sin \alpha_{i,t})^2},
\]

(3)

where \( \rho_{1,2} \) denotes the density of the two media and \( k_x \) is the \( z \)-axis component of \( k \) - the wavenumber (as shown in eq. 3). By \( \alpha_{i,t} \) we denote the angle of incidence and angle of transmission respectively.

The reflectivity matrix \( R \) is a convolution matrix based on the operator \( R \) (eq. 1): the multiplication by \( R \) in the \( k_x - \omega \) domain can be expressed in terms of a matrix multiplication by \( R \) in the \( x - \omega \) domain. Fig. 2b represents schematically the formation of a convolution matrix from an operator. Each row of the reflectivity matrix \( R(z_m, z_m) \) contains an operator in the space-frequency domain which corresponds to a certain grid point at depth \( z_m \).
Fig. 2: (a) The WRW model, after the formulation of equation 1. For each reflection, a WRW-term is added to the total expression for the measured pressure $P(z_0, z_0)$. (b) Forming a convolution matrix

In the simple case that all grid points at a certain depth have the same reflective properties (i.e. no lateral changes), the reflectivity matrix has a Toeplitz structure.

The angle dependent properties of the reflectivity operator are obtained from eq. 2 and 3. From eq. 3 it follows that each value of the reflectivity operator in the wavenumber-frequency domain represents reflectivity for a certain angle of incidence [6]. Two extreme cases are:

- Only diagonal elements of the reflectivity matrix are filled with non-zero values. In this case, the operator is constant over a range of wavenumbers, thus angles (angle independent reflector). In this case the grid points are treated as point diffractors.
- Constant values along the columns of the reflectivity matrix. In this case, the reflectivity operators in the wavenumber domain are non-zero for $k_x = 0$ only. Therefore, the grid points on the reflector reflect only the horizontal plane waves. One should note though that this is not a realistic situation.

In general, the matrix $R$ is completely filled with various complex values. Note that the diagonal elements represent $x = 0$ (i.e. the grid-point under consideration). From Fourier theory it follows that this corresponds to a summation over the wavenumbers (or angles). Therefore, the diagonal elements of the reflectivity matrix represent angle-averaged reflectivity. Angle-averaged reflectivity is the result the many migration algorithms deliver, hence the importance of the diagonal elements of $R$. The off-diagonal elements of $R$ contain information related to the angle-dependency of the reflectivity.

4. NUMERICAL EXAMPLES

The main aim of this section is to illustrate - through numerical examples - the properties of WRW wave equation modeling compared to those of ray tracing, as well as to analyze angle dependent reflectivity. Therefore, a number of simple, 2-D numerical examples have
been modeled. The velocity model consists of a single horizontal reflector at the depth of 100 m that forms the boundary between two media. Different cases will be considered with different medium properties. In all cases, the velocity is laterally invariant whereas density may vary along the reflector. The acquisition geometry consists of a 1.5 km long aperture with 5 m receiver spacing and a single source in the middle of the array. Medium properties for the different cases are listed in Table 1.

<table>
<thead>
<tr>
<th>Configuration</th>
<th>( \rho_1 ) (kg/m(^3))</th>
<th>( \rho_2 ) (kg/m(^3))</th>
<th>( c_1 ) [m/s]</th>
<th>( c_2 ) [m/s]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Case 1</td>
<td>1500</td>
<td>2200</td>
<td>2000</td>
<td>3500</td>
</tr>
<tr>
<td>Case 2</td>
<td>2000</td>
<td>1000</td>
<td>2000</td>
<td>2500</td>
</tr>
<tr>
<td>Case 3</td>
<td>1000-3500</td>
<td>1000-3500</td>
<td>2000</td>
<td>2200</td>
</tr>
</tbody>
</table>

*Table 1: Medium properties of different cases studied*

4.1. **Case 1 – Homogeneous media**

As shown in Table 3.1, the medium properties in case 1 refer to the simple situation of two homogeneous layers with layer 2 having a higher acoustic impedance than layer 1. Therefore, all the grid points on the reflector have the same impedance contrast, hence, the same reflectivity operator in the wavenumber domain (homogeneous reflector). Fig. 3a and 3b show the shot records obtained from WRW and ray tracing modeling, respectively. Due to the simple properties of the reflector, no significant difference can be observed in the modeled shot records between the two methods. Fig. 3c displays the absolute value of the reflectivity operator in the wavenumber-frequency \((k_x, \omega)\) domain. As illustrated in this figure, three different areas are clearly separated by dipping boundaries.

- The central turquoise area corresponds to reflection angles of incidence from 0° (i.e. vertical incidence) up to the critical angle.
- The second, red, area consists of post-critical angles up to 90°. For these angles total reflection occurs, thus, the reflectivity values are close to 1.
- The last, blue, area corresponds to reflectivity over 90° and refers to the evanescent part of the reflected wave field. More information on evanescent wave fields can be found in [5].

The angle dependent reflectivity can be better observed in Fig. 3d in which the real part and the absolute value of the reflectivity are depicted as a function of angle of incidence. The real part of the reflectivity reaches its minimum for 0° angle of incidence and its maximum value at critical angle. It collapses after the critical angle, while the absolute value remains 1 till 90°. Fig. 3e shows the reflectivity matrix \(R\) with dominant diagonal elements which represent angle-averaged reflectivity, as mentioned before.

4.2. **Case 2 – Homogeneous media & polarity reversal effect**

In this case, the bottom layer has lower acoustic impedance than the upper layer. Therefore, one expects to have a negative reflection coefficient. However, in this specific situation, the angle dependency plays an essential role in the simulated output.
Fig. 3: Shot records obtained by the WRW modeling (a) and the ray tracing approach (b) for case 1. Different forms of the R operator follow: (c) in the $k_x - \omega$ domain, (d) versus angle and (e) the actual image of the reflectivity matrix as used in the WRW formulation.

It even causes the reflection coefficient to change its polarity as the angle of incidence increases. This is illustrated in Fig. 4a and 4b for the WRW model and the ray tracing method respectively. For narrow offsets (small angles), the reflector has a negative polarity. The amplitude goes smoothly to zero for further offsets and becomes positive again for larger offsets (larger angles), with a reversed, positive polarity. However, the range of offsets with very low amplitude is different for the two simulated results. The WRW modeling result displays a wider range of offsets with very low amplitude level. As illustrated, the reflectivity is negative for small angles; it becomes zero for a certain angle and then steeply reaches the critical angle. There is only a small difference between the angle at which zero reflectivity occurs and the critical angle.

4.3. Case 3 – Media with random density along the lateral direction

In this case the density is randomly varying between 1000 kg/m$^3$ and 3500 kg/m$^3$ along the top and the bottom of the boundary but the velocity remains laterally invariant in each layer. Therefore, the wave propagation in the upper medium remains simple. Because of the random density effect, low impedance contrasts may occur along the reflector. These produce the no-reflectivity gaps on the event in the case of ray-tracing, and the diffraction patterns in
Fig. 4: Shot records obtained by the WRW modeling (a) and the ray tracing approach (b) for case 2. Different forms of the R operator follow: (c) in the $k_x - \omega$ domain, (d) versus angle and (e) the actual image of the reflectivity matrix as used in the WRW formulation.

the case of wave equation modeling. By comparing Fig. 5a and 5b, it can indeed be seen that diffraction patterns due to the irregularities in the reflector are not included in the simulated result obtained by ray tracing, since each grid point on the reflector is treated individually. This example illustrates that the wave equation approach is clearly preferred in this case.

5. DISCUSSION AND CONCLUSION

In this paper, the reflectivity matrix in the general context of the WRW model is studied and different examples are provided to highlight its properties as well as its effect on the final shot record. The general frame of this study consists of the modeling of a horizontal reflector based on two different approaches, the wave equation approach (using the aforementioned reflectivity matrix) and the ray tracing approach. Differences between the outcome of these two methods comprise the limitations of ray tracing, while pointing out the virtues of the WRW concept. As shown in Fig. 1(a), the essence of the wave equation approach is that the reflectivity matrix acts as a spatial convolution operator on the downgoing wave field. On the contrary, ray tracing only applies an amplitude coefficient on each ray. Fig. 5a and 5b illustrate characteristically the outcome. The $R$ matrix, applied on the wave field in the spatial domain, produces a main uninterrupted event with diffraction patterns due to the random lateral density variations that are present in this case. On the other hand, the ray tracing
technique fails to model these diffraction patterns, as it is based on the local impedance contrast at each grid point. Hence, non-realistic gaps are produced on the reflection event.

In conclusion, the distinguishing features of the wave theory modeling (WRW approach) appear when there is irregularity or variability in the subsurface structures, especially in the lateral direction. This situation happens often in reality and ray tracing is unable to invert correctly for amplitudes during the migration process. This example illustrates once more the importance of wave equation based modeling.

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HIGH RESOLUTION RECONSTRUCTION OF IRREGULARLY SAMPLED, ALIASED MEASUREMENTS

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Abstract: In many multi-channel measurement situations it is not possible to acquire data with sufficiently dense sampled arrays. Furthermore, the measurement locations are not always regularly sampled or may have holes, e.g. due to obstructions. Reconstruction algorithms are used to transform the measurements into a regularly sampled, aliasing free dataset. Usually, in these reconstruction algorithms, the assumption is made that the measurements can be efficiently described in a suitable transform domain. Typical examples of such domain transforms are the Fourier transform and the generalized Radon transform. However, due to the non-ideal sampling a straightforward transformation is not possible. Therefore, the reconstruction process is defined as an inversion problem, for which extra constraints have to be included to make the solution unique. In this paper we will give an overview of typical transforms and the involved constraints that can be used to reconstruct measurements onto a user-defined spatial grid.

Keywords: Sampling, Reconstruction, Fourier transform, Radon transform, Inversion, Sparseness, High-resolution

1. INTRODUCTION

In many multi-channel measurement situations it is often not possible to acquire data with sufficiently dense sampled arrays. Furthermore, the measurement locations are not always regularly sampled or may have holes, e.g. due to obstructions. In the field of oil and gas exploration using acoustic signals this is often the case. The objective of exploration seismics is to image the subsurface of the earth based on acoustic reflection measurements made at the
surface. A source at the surface emits an acoustic wavefield which propagates through the subsurface. Any inhomogeneity in the earth will cause a part of the downgoing energy to scatter back to the surface, where a line or a grid of receivers is positioned (see Figure 1a). The recordings consist of primary and multiple reflections and have, for simple structures, a more or less hyperbolic nature (see Figure 1b). Based on wave theory, these reflection measurements can be transformed into an image of the subsurface. Before such an imaging process several preprocessing algorithms have to be applied to the data, e.g. to remove noise and other unwanted energy. These algorithms are usually designed for regularly sampled, alias-free data. Therefore, reconstruction algorithms are used to transform the measurements into a regularly sampled, aliasing free dataset. Usually, in these reconstruction algorithms, the assumption is made that the measurements can be efficiently described in a suitable transform domain. Typical examples of such domain transforms are the Fourier transform and the generalized Radon transform. However, due to the non-ideal sampling a straightforward transformation is not possible. Therefore, the reconstruction process is defined as an inversion problem, for which extra constraints have to be included to make the solution unique. In this paper we will give an overview of typical transforms and the involved constraints that can be used to reconstruct measurements onto a user-defined spatial grid.

2. LEAST-SQUARES FOURIER RECONSTRUCTION

For seismic records reconstruction in the time domain is not necessary. For this reason only spatial reconstruction is addressed here. A popular choice is to use the spatial Fourier transform as the basis for data reconstruction. Thus, we want to describe our dataset in terms of plane wave components. The input dataset (space-time recording, such as shown in Figure 1b) is transformed to the space-frequency domain with the Fast Fourier Transform (FFT). Here the inversion can be calculated for each frequency separately. The inversion problem we want to solve is formulated in vector notation as:

$$p = A\tilde{p} + n$$

with the elements defined as:

$$p_n = p[n\Delta x], \quad A_{mn} = \frac{\Delta k}{2\pi} e^{-j mn\Delta x}, \quad \tilde{p}_m = \tilde{p}[m\Delta k]$$

Fig.1: a) Seismic acquisition in the marine case with a source that emits sound waves and a receiver cable with an array of hydrophones. b) Seismic reflection paths have a hyperbolic nature, as can be observed in the field measurement shown on the right. Blue lines represent primary reflections and the red line is a multiple reflection.
Here $\mathbf{p}$ is the monochromatic data vector in the spatial domain and $\mathbf{\tilde{p}}$ the model vector in the wavenumber domain for one frequency. $\mathbf{A}$ is the inverse Fourier transform matrix, $\Delta x$ the space sampling and $\Delta k$ the wavenumber sampling. Energy outside the spatial bandwidth used in the inversion is accounted by the noise term $\mathbf{n}$. After the data is reconstructed for each frequency, the space-frequency matrix is transformed back to the space time domain and the reconstruction is evaluated there. For uniform and aliasing-free sampling without gaps the total number of traces ($N_o$) and the total number of wavenumber components ($N_k$) are equal. In our case the data vector has less entries (reduced by the number of gaps) than the model vector. Trying to solve the system of equation (1) without any restriction would produce an infinite number of solutions. Thus, certain constraints need to be used. Usually, the solution which has the minimal model norm is chosen. The resulting constrained system has only one solution.

In general the objective of Fourier reconstruction is to minimize the following quantity:

$$J = \frac{1}{c^2} \left\| \mathbf{W}^{1/2} (\mathbf{p} - \mathbf{A} \mathbf{\tilde{p}}) \right\|_2^2 + \frac{1}{\sigma_p^2} \left\| \mathbf{\tilde{p}} \right\|_2^2,$$

where $\left\| \mathbf{\tilde{p}} \right\|_2^2$ is the $L_2$ norm of $\mathbf{\tilde{p}}$. The noise covariance matrix can be expressed [1] as $\mathbf{C}_n = c^2 \mathbf{W}^{-1}$, where $c$ is a constant. The data weighting matrix $\mathbf{W}$ is a diagonal matrix with the diagonal elements defined as $W_{nn} = \Delta x_n$ and is normalized such that $\sum \Delta x_n = 2\pi / \Delta k$.

The minimum of equation (3) is obtained by the least-squares estimator:

$$\mathbf{\tilde{p}} = (\mathbf{A}^H \mathbf{W} \mathbf{A})^{-1} \mathbf{A}^H \mathbf{W} \mathbf{p}.$$  

In our case the inverse problem is ill-conditioned due to the lack of information in the missing traces. Thus, the inversion has to be regularized. Here we use damped least squares minimization:

$$J = \frac{1}{c^2} \left\| \mathbf{W}^{1/2} (\mathbf{p} - \mathbf{A} \mathbf{\tilde{p}}) \right\|_2^2 + \frac{1}{\sigma^2_p} \left\| \mathbf{\tilde{p}} \right\|_2^2,$$

where $\sigma_p^2$ is the a priori model variance [2]. The second term in equation (5) is the restriction on the Euclidean model norm. The minimum of this objective function is derived as:

$$\mathbf{\tilde{p}} = (\mathbf{A}^H \mathbf{W} \mathbf{A} + \lambda \mathbf{I})^{-1} (\mathbf{A}^H \mathbf{W} \mathbf{p}),$$

with the damping term $\lambda = c^2 / \sigma_p^2$. Here the estimator $\mathbf{\tilde{p}}$, which solves equation (5), is the model vector that explains the data best for certain wavenumbers (band limitation) and coevally has the smallest values. This solution is often referred to as Fourier reconstruction with minimum norm (FRMN). The influence of the constraint can be varied via the damping term $\lambda$. One should realize that by taking this solution actually means assuming a minimum energy norm on the model space parameters. After the model vectors have been estimated the data are generated on the uniform grid. We refer to [3] for a detailed discussion of inverse theory. Also see [1] and [2] for further information on FRMN.
In Figure 2c an example of FRMN is shown for the case of two plane wave events (Figure 2a), which have been severely decimated in their spatial sampling (Figure 2b).

3. FOURIER RECONSTRUCTION WITH SPARSENESS

As can be observed in Figure 2c, a proper reconstruction quality of the minimum norm least-squares inversion process is obtained for small gaps only. For the large gaps, the minimum norm constraint enforces the reconstructed wavefield to have small values inside the gaps. Note, however, that the data match is close to perfect (i.e. the reconstructed data resembles the original data at the measurement locations). Thus, to improve the reconstruction quality for the larger gaps, the minimum norm assumption needs to be replaced by another constraint. A popular choice for this is a sparseness constraint. The rationale behind this constraint is that we assume that we need only a limited number of non-zero model domain components (in our case Fourier components) to describe our input data.

For Fourier reconstruction using sparse inversion (FRSI), one possibility is to minimize the following objective function:

\[
J = \frac{1}{c} \| p - A \hat{p} \|^2_p + \sum_k \ln(1 + \frac{\hat{p}_k^2}{\sigma^2_p}),
\]

where the second term represents the Cauchy weighting (see e.g. [4]). Note that this weighting is data-dependent, such that solving equation (7) becomes a non-linear problem, which is typically carried out by an iterative solver (e.g. a Conjugate Gradient scheme). In each iteration, the model space parameters of the previous iteration are used to construct the least-squares solution:
\[ \tilde{p} = (A^H A + S)^{-1} A^H p, \]  

(8)

where \( S \) is a diagonal matrix with the elements defined as:

\[ S_{kk} = \frac{1}{\lambda + \frac{\bar{p}_k \bar{p}_k}{\sigma^2_p}}. \]  

(9)

Figure 2d shows the result for of this approach applied to the decimated input data in Figure 2b. Note that the reconstruction has greatly improved compared to the FRMN result of Figure 2c. Furthermore, note that indeed the Fourier domain has become much sparser: only the wavenumber components belonging to the two plane events have been emphasized.

4. EXTENSION TO PARABOLIC RADON

From Figure 1b it is clear that in seismic measurements the assumption that the observed responses can be represented by a small number of plane wave components is not true in general. Especially because of the curved nature of the reflection events, a different representation can be more effective. Therefore, it has been proposed to use parabolas as the basis functions. If the seismic measurements are sorted in the so-called common midpoint (CMP) gather domain as a function of offset (see e.g. [5]), the resulting measurements can be approximated by parabolas with the apex at the zero offset. As a result the parabolas can be described by two parameters: apex time \( \tau \) and curvature \( q \). The advantage of a parabolic description is that the reconstruction problem can still be defined in the frequency domain (see [6]). Basically, all of the previous expressions can be used again, except that the transform matrix \( A \) needs to be redefined as follows:

\[ A_{nn} = e^{-j \omega_m \Delta q x_n^2}, \quad \tilde{p}_m = \tilde{p}[m \Delta q], \]  

(10)

where \( \Delta q \) represents the sampling of the curvature axis. Note that the parabolic Radon transform is not an orthogonal transform, even if the input signal is regularly sampled in \( x \).

Figure 3 demonstrates the least-squares inversion with a minimum norm constraint, as given by equation (6) for the case of a parabolic transform operator. Although the two events (Figure 3a) map into two small areas in the parabolic Radon domain (Figure 3c), typical smearing effects can be observed due to the fact that the input data is bounded spatially (i.e. edge effects). When imposing a sparseness constraint in the frequency-curvature domain, similar as defined in equation (7), a much better resolution in the Radon domain is obtained, as visible in Figures 3e and 3f.

One application of the high-resolution parabolic Radon transform is to separate events with (slightly) different curvatures. This is desired if we want to remove multiple reflections from the seismic measurements. The multiple reflections have travelled more into the shallow part of the earth, where propagation velocities are low. Therefore, the multiples can be recognized as events with a stronger curvature compared to primary events at a similar arrival time, which have travelled in the deeper part of the earth where velocities are usually higher. In Figure 4 such an application is demonstrated on a synthetic data gather from an earth model with three reflecting boundaries. The three primary events are indicated in Figure 4a with the arrows. After the high-resolution Radon transform, each event is focused in a very
small area (Figure 4b), after which the undesired multiple events, with the larger curvatures, can be removed by muting (Figure 4c). Finally, the result is transformed back into the space-time domain, yielding the desired primary reflections (Figure 4d).

Fig. 3: Demonstration of the least-squares parabolic Radon transform using a minimum norm and a sparseness constraint. a) Input data in the space-time domain. c) Transformed data in the curvature-apex time domain using a minimum norm constraint. e) Result using a sparseness constraint. The bottom row represent the temporal Fourier transforms of the pictures from the top row. Note that the sparseness constraint reduces the smearing. (Figure from M. Schonewille)

Fig. 4: Example of using the high-resolution parabolic Radon transform to separate primaries and multiples. a) Input data gather after an overall curvature correction. The arrows point at the primaries. b) The high-resolution parabolic Radon transform. c) Result of muting the area with the multiple events. d) Reconstructed primaries. Note that the proper primary reconstruction.

5. SPARSENESS IN CURVATURE AND TIME

If the spatial sampling becomes too poor, the sparseness constraint in the curvature domain alone is not enough to recover the measurements properly. In that case an additional constraint is included: the data is also sparse in the time domain, meaning that events have an impulsive character. However, to exploit this property, the inversion process needs to be carried out in the time domain, meaning that a decomposition of the large inversion problem into smaller sub-problems per frequency is not possible. First steps in this direction were
already taken by [7] and followed-up by [8] and [9]. In [10] this method was used to help reconstructing a severely aliased data that is involved in 3D multiple removal. The following example is from this application, where a well sampled dataset with 10 hyperbolic events (Figure 6a) is severely decimated (Figure 6b). This decimated data becomes the input for an iterative hyperbolic reconstruction process. The initial estimate of the hyperbolic model space (Figure 6d) is obtained by simple summing over all possible hyperbolic trajectories. The resulting estimate has a poor resolution. After applying the sparse inversion process, the estimate of the model space becomes more and more sparse (Figure 6e-g). Finally, the data can be reconstructed with high accuracy to the original dense sampling (Figure 6c).

![Figure 6](image)

**Fig. 6:** a) Original data with 10 hyperbolic events. b) Decimated input data. c) Reconstructed data with a time domain sparse hyperbolic transform. d) Initial transform domain. e-g) Transform domain after 1, 10 and 20 iterations respectively. (Figure from E.J. van Dedem).

6. CONCLUSIONS

Spatially irregularly sampled and aliased measurements can often be reconstructed with a good quality by choosing the proper transform domain and including a suitable constraint on the model space parameters. In practice, a sparseness constraint is very powerful. The downside of such constraint is that the inversion procedure becomes nonlinear and need to be solved iteratively. Moreover, for some situations also a sparseness constraint in the time domain needs to be included, which means that the inversion problem cannot be solved independently per frequency.

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ACQUISITION ANALYSIS AND DESIGN USING FOCAL BEAMS

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Abstract: The quality of the images to be obtained with the seismic method for the exploration of oil and gas strongly relies on the acquisition of the seismic data. Aspects that play a role are: -1- the dimensionality of the survey; either 2D (measurements along a line), 3D (measurements at the surface), or 4D (repeated measurements at a surface); -2- the spatial sampling of detectors (geophones, hydrophones) and sources (vibroseis, dynamite, air-gun), both in the in-line and cross-line directions; -3- the aperture, i.e., the size of the survey, -4- the azimuth distribution, i.e., the illumination and sensing angles.

Note that together, these aspects determine the involved number of recording channels. A compromise must be made here, such that the required information is obtained at minimum costs and with available recording instrumentation. To design the optimum acquisition geometry, it is required that as much information about the subsurface is included as is available. This is because the subsurface structure, and in particular, the spatial velocity distribution of the subsurface earth layers, determines the illumination and sensing properties of a certain acquisition geometry.

In this paper the focal beam method is presented to solve this problem. In the focal beam method, the image to be obtained with a particular seismic acquisition geometry is computed beforehand, taking into account the expected velocity distribution of the subsurface. This predicted image is the product of the focal source beam and the focal detector beam, describing the imaging power of the sources and detectors respectively. Because the contributions of the sources and detectors are assessed separately, the link with the acquisition geometry is made.

Keywords: Seismic, Acquisition, Survey design
1. INTRODUCTION

The quality of a seismic image is expressed in terms of resolution and reliability of the (relative) amplitudes, i.e., reflection strength. It strongly relies on the acquisition geometry of the seismic survey, i.e., the position and number of sources and detectors.

Ideally, the surface would be fully covered with densely spaced sources and detectors, i.e., their spatial sampling distance is satisfying the Nyquist-Shannon sampling theorem. The response of every single source would be recorded by all detectors. In that case all wavefields involved would be properly measured and the seismic image quality would be limited by the laws of physics only. Unfortunately, such an ideal acquisition set-up is too expensive. Therefore, more affordable alternatives are used in practice. Consequently, the aperture is limited, i.e., only a limited area is covered, and/or the Nyquist criterion is not fulfilled and/or the response of a particular source is not recorded at all detector locations, i.e., the detector configuration depends on the actual source position. A typical example of the latter is a marine survey, where a seismic vessel is towing both the source (airgun array) and the detectors (certain number of seismic streamers).

The problem to be solved is therefore summarized as follows: what is the most affordable acquisition configuration that delivers the required subsurface image quality.

Aspects that play a role are: -1- the dimensionality of the survey; either 2D (measurements along a line), 3D (measurements at the surface), or 4D (repeated measurements at a surface); -2- the spatial sampling of detectors (geophones, hydrophones) and sources (vibroseis, dynamite, air-gun), both in the in-line and cross-line directions; -3- the aperture, i.e., the size of the survey, -4- the azimuth distribution, i.e., the illumination and sensing angles.

Note that together these aspects determine the involved number of recording channels. As mentioned, a compromise must be made here, such that the required information is obtained at minimum costs and with available recording instrumentation. To design the optimum acquisition geometry, it is required that as much information about the subsurface is included as is available. This is because the subsurface structure, and in particular, the spatial velocity distribution of the subsurface earth layers, determines the illumination and sensing properties of a certain acquisition geometry with respect to the target area in the subsurface.

In this paper the focal beam method is presented as an acquisition geometry analysis tool. In the focal beam method, the image to be obtained with a particular seismic acquisition geometry is computed beforehand, taking into account the expected velocity distribution of the subsurface. This predicted image is the product of the focal source beam and the focal detector beam, describing the imaging power of the sources and detectors respectively. Because the contributions of the sources and detectors are assessed separately, the link with the acquisition geometry is made.

2. THE WRW MODEL FOR SEISMIC MEASUREMENTS

The WRW model describes the seismic data in terms of matrix operators in the frequency domain [1]. According to this model, each monochromatic component (single frequency) of the primary wave field \( P(z_0, z_d) \) that is recorded by detectors at level \( z_d \) due to sources at level \( z_s \) in the case of a single reflector at depth level \( z_m \) can be described in the space-frequency domain by:
The designation ‘WRW model’ stems from the two propagation matrix operators $W$, and matrix operator $R$, whose functions are explained below. In practice the subsurface contains many reflectors at different depth levels: in that case the total response is considered to be the summation of the responses of the individual reflectors.

The lateral coordinates $x$ and $y$ and the frequency $\omega$ have been left out for notational convenience.

The matrix operators in eq. 1 have the following meaning:

- $S(z_s)$: source matrix. It contains the amplitude and phase of the source wavelet at the frequency under consideration. One column represents one source (array) and determines its position in space.
- $W(z_m, z_s)$: forward wave field propagation matrix. Each column contains a discrete version of the Green’s function that describes the wave propagation from one point (one lateral location) at the surface $z_s$ to many points at depth level $z_m$.
- $R(z_m, z_m)$: reflectivity matrix. It describes the conversion of an incident wave field into a reflected wave field, for more information see [2].
- $W(z_d, z_m)$: forward wave field propagation matrix. It describes the wave propagation from one point at the depth level $z_m$ to many points at the surface $z_d$.
- $D(z_d)$: detector matrix. It contains the detector wavelet. One row represents one detector (array) and determines its position in space.

It follows that an element $P_{ij}$ of the data matrix $P(z_d, z_s)$ corresponds to one frequency component of the seismic response as measured by a detector at the location $i$ due to a source at location $j$. Therefore, one column of the data matrix represents a common source gather (shot record) and one row represents a common receiver gather. Other data gathers, such as CMP gathers or common offset gathers, can also be identified in the data matrix. This model is valid for stationary acquisition geometries and stationary parts of non-stationary acquisition geometries.

3. IMAGING AS A DOUBLE FOCUSING STEP

The purpose of the seismic method is to obtain an image of the subsurface in terms of the seismic reflectivity. Such an image can be obtained from the seismic data by a double focusing process [1] according to:

$$ P(z_d, z_s) = D(z_d) W(z_d, z_m) R(z_m, z_m) W(z_m, z_s) S(z_s). $$

(1)

Here matrices $D(z_d)$ and $S(z_s)$ aim at inverting $D(z_d)$ and $S(z_s)$ respectively; the superscript $H$ denote the hermitian (i.e., the complex conjugate transpose). $W(z_m, z_d)^*$ is a row vector – the dagger symbol indicates a row vector – that contains a time-reverse Green’s function at detector level $z_d$, related to subsurface point $(x_k, y_k, z_m)$. It acts as a focusing operator at the detector side such that $W(z_m, z_d)^* W(z_d, z_m) = I_k(z_m, z_m)$, a row vector with
element $k$ equal to 1, other elements are zero. Likewise $W(z_{s}, z_{m})_{k}$ is a column vector containing similar information related to source level $z_{s}$, such that $W(z_{m}, z_{s})W(z_{s}, z_{m})_{k}^{*} = I_{k}(z_{m}, z_{m})$, a column vector with element $k$ equal to 1, other elements are zero. It acts as a focusing operator at the source side. Substituting equation (1) into equation (2) yields:

$$W(z_{m}, z_{s})_{k}^{*} D(z_{d})^{H} D(z_{d}) W(z_{d}, z_{m}) R(z_{m}, z_{m}) W(z_{m}, z_{s}) S(z_{s}) S(z_{s})^{H} W(z_{s}, z_{d})^{*}.$$  \tag{3}

After some matrix manipulation is becomes clear that this expression equals $< R_{m}(z_{m}, z_{m}) >$, i.e., an estimate of diagonal element $k$ of reflectivity matrix $R(z_{m}, z_{m})$. This diagonal element contains the angle-averaged reflectivity information of subsurface location $(x_{k}, y_{k}, z_{m})$. Repeating this procedure for all subsurface locations retrieves an 3D reflectivity image of the subsurface. Hence, the imaging of the subsurface can be obtained via a double focusing process: focusing in sensing and focusing in illumination. Algorithms that carry out this imaging task are called depth migration algorithms. In practice, for reasons of efficiency, they are implemented in such a way that they carry out the double focusing process in an implicit way.

In seismic acquisition, the source and detector distributions are coarse. This means that the use of the hermitian is not a very accurate approximation to a proper inversion. From equation (3) it follows that this directly influences the quality of the estimate for the reflectivity.

### 4. IMAGING AS A PRODUCT OF FOCAL BEAMS

To assess the quality of a particular seismic acquisition geometry more closely, it is worthwhile to analyse the two focusing steps, focusing in detection and focusing in emission, individually. Furthermore, it is not desired to model complete 3-D seismic data sets in order to evaluate equation (2). Therefore, the concept of focal beams has been introduced by [3; 4]. The focal source beam is defined by:

$$W(z_{s}, z_{s}) S(z_{s}) [S(z_{s})^{H} W(z_{s}, z_{s})_{k}^{*}].$$  \tag{3}

Basically this equation contains the right part of equation (2). The expression between brackets [ and ] is called the source focussing operator. It acts on the sources. Then the emitted signal is extrapolated to all depth levels. The downward extrapolation process is shown in Figure 1, where several snapshots are shown. The complete beam is shown in Figure 2.

Similarly the focal detector beam is given by:

$$[W(z_{m}, z_{d})_{k}^{*} D(z_{d})^{H}] D(z_{d}) W(z_{d}, z).$$  \tag{4}

It turns out that an element by element multiplication of the focal beams results in the depth-migrated image of a point diffractor [5]. See Fig. 3, where the product of two beams is shown. Notice the improvement in resolution of the product in comparison with the resolution of each of the two beams. This illustrates a well-known property that is often exploited in the design of echo-acoustic systems: the properties of the source(array) are used.
to enhance the properties of the detector(array) and vice versa, e.g., think of the crossed-array configuration often encountered in sonar systems.

Another interesting property of the beams is that they can be used for an analysis of the angles involved in illumination and sensing. To that end the beams are transformed to the linear Radon domain at focal depth. An example is shown in Figure 4. Ideally, the beam should illuminate or sense the target point uniformly over a wide range of angles. In practice – again due to the practical and commercial restriction of seismic acquisition – illumination and sensing are usually far from uniform and the range of angles involved is limited.

Fig 1: The signal of a focal source array at the surface is downward propagating. Three snapshots are shown. The snapshot at the right is at focus time.

Fig 2: Focal beam (energy)

Fig 4: Focal beam: horizontal cross-section at focal depth in Radon domain. The axes correspond to the angles of illumination (or sensing) in the x- and y-direction respectively.
Fig. 3: The migrated image of a point diffractor at the focus point (right) is the product of the focal source beam (left) and the focal detector beam (middle). This results corresponds to the resolution to be obtained with the acquisition configuration.

5. EXAMPLE: ‘IDEAL’ ACQUISITION GEOMETRY VERSUS REGULAR MARINE STREAMER ACQUISITION

The focal beam analysis was used to analyse two acquisition geometries. Geometry 1 is a ‘full geometry’, where the complete aperture is densely sampled with sources as well as with detectors. It can be considered as the ideal geometry. Geometry 2 is a practical acquisition geometry as used in marine seismic surveys, where a seismic vessel is towing twelve streamers of 8 km length, with 100 m streamer separation. The sail line separation is 600 m. The inline spatial detector and source sampling interval is 50 m. The result is shown in Figure 5. Here the product of the focal source beam and the focal detector beam is shown at the target depth. This corresponds to the lateral resolution. The differences between the geometry 1 and 2 are modest. The best resolution is provided by geometry 1, but the resolution of geometry 2 is sufficient. A few minor side lobes are present. This means that both geometries are capable of producing a structure image of the subsurface. However, the results of the Radon domain analysis are very different for the two geometries. Geometry 1 is illuminating-and-sensing the subsurface target point with a wide range of angles and quite uniformly. Geometry 2 on the other hand, is only capable of transmitting-and-sensing a limited range of angles. This means that from geometry 2 it is not possible to retrieve the full angle-dependent reflectivity or other azimuth-dependent effects (e.g., anisotropic speed of sound). This is one of the reasons of the increasing popularity of so-called wide-azimuth acquisition where the seismic vessel towing the streamers is accompanied by one or more shooting vessels that illuminate the subsurface from azimuths perpendicular to the streamer direction (another important reason is the improved signal to noise ratio of this configuration).
Fig. 5: An ideal acquisition geometry (right) with a dense coverage of sources and
detectors compared with a state-of-the-art marine seismic survey (right) where a seismic
vessel is towing twelve streamers of 8 km length, 100 m apart. Sail line interval is 600 m.
Note that both geometries deliver a proper lateral resolution. However, only the ideal
geometry provides the desired range of azimuths. The marine survey illuminates-and-senses
the subsurface using a limited range of azimuths only.

In general, the focal beam method enables a separate assessment of the contribution of the
source configuration to the image quality and the contribution of the detector configuration.
This is because the images quality is obtained via the product of the two beams: the focal
detector beam and the focal source beam. This separate assessment makes it possible to
improve a particular acquisition geometry design efficiently: it becomes clear whether more
sources or detectors are required, or whether more source azimuths are required or detector
azimuths, etc., etc.

6. CONCLUSION

The focal beam analysis method provides a direct link between the acquisition parameters at
the surface (number of sources, source locations, number of detectors, detector locations,
which sources are sensed by which detectors) and the image quality at a target location in the
subsurface. The key parameters here are resolution and angle range, both in illumination and
in sensing. The image quality is obtained via the multiplication of the focal beams: the focal
source beam and the focal detector beam. This means that the contribution of the sources and
the contribution of the detectors to the image quality can be assessed separately. This
information can be used to design and/or update a particular acquisition geometry.

7. ACKNOWLEDGEMENTS
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REFERENCES

SURFACE-RELATED MULTIPLE REMOVAL IN SEISMIC DATA BY
A DATA-DRIVEN METHODOLOGY

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Abstract: For seismic exploration acoustic sources and receivers are relatively densely
positioned at the earth's surface in order to measure the reflection response from the
subsurface inhomogeneities. However, this surface acts as a perfect mirror, sending back all
upgoing wavefields. Thus, each primary reflection will generate a train of surface-related
multiple reflections behind it. These multiples can obscure primary reflections from deeper
targets. For the exploration of oil and gas reservoirs, multiples can be one of the main
problems in applying the seismic method, especially in a marine environment. The inherent
relationship between primaries and the multiples - via reflection at the surface of the earth -
can be exploited to predict multiples from the measured primaries and subsequently subtract
them from the measured data. A method based on these principles has been developed over
the last two decades and is currently being applied to many seismic datasets with large
success. Its main characteristic is that only the seismic data is involved in the prediction of
the multiples – based on cross-convolutions of the various measurements - and no subsurface
information is required. An overview of the methodology and examples of its application will
be given.

Keywords: Multiple reflections, Scattering series, Source signature, Inversion

1. INTRODUCTION

Any acoustic reflection measurement suffers from the occurrence of multiple reflections.
In depends on the measurement situation whether these multiple reflections interfere with the
desired primary reflections from the earth. In seismic exploration carried out in marine environments, multiple reflections related to the earth’s surface complicate the interpretation of the earth’s subsurface in many occasions. Especially in situations with water bottom depths up to 1 or 2 km multiple reflections may interfere with the desired reflections of geologic formations that contain prospects for hydrocarbon exploration. Therefore, multiple suppression is often one of the key pre-processing steps in the seismic data processing workflow. In general, multiple removal methods can be categorized into multi-channel filtering methods, based on the different spatial properties of multiples and primaries, and methods that are based on the inherent relationships between primaries and multiples and that can be considered as prediction and subtraction methods (see [1] for an overview). The first mentioned category is based on the fact that multiples have travelled in different subsurface layers compared to primary reflections that arrive at the same travel time, and therefore multiples and primaries exhibit a different move-out behaviour along the different receiver stations. The second category has a closer relation with the wave equation and will be considered in this paper.

![Fig.1: Each first order surface multiple consists of two primaries that are combined at the surface. By making all possible combinations of primary connections at the surface, and summing the results, a surface-related multiple is constructed.](image)

**2. THEORY OF SURFACE-RELATED MULTIPLE REMOVAL**

When a surface-related multiple is considered (as shown in Figure 1a) it can be observed that it consists of separate paths that are connected at the surface. Each reflection path is in itself a primary reflection. If we could make the proper combination of two primaries, a surface-related multiple can be constructed from our seismic data. However, the reflection point of a surface multiple is not known in advance. Therefore, the final result can be achieved by combining all possible reflection points at the surface. This is quantified by selecting a common source gather (one source, many receivers) and the measurements from a common receiver gather (one receiver, many sources) and combining them by a convolution process (see Figure 1b). When adding all these convolution products, wave theory tells us that the final multiple reflection will survive this stack and other contributions will cancel. This stacking process is called a Fresnel stack, in which only contributions from within the Fresnel zone will survive this process. In this case, the Fresnel zone is situated around the surface reflection point.

This process can be quantified mathematically. The basis of this multiple prediction method was put by [2] who noted that by autoconvolution of seismic measurements multiples could be generated. In [3] the relation between primaries and multiples in a two-dimensional situation was described. In [4] the multiple suppression was formulated into a mathematically sound theory for plane waves in a horizontally layered earth. In [5] a true multi-dimensional
approach to multiple removal was defined, in which the seismic data itself is used as the multi-dimensional multiple prediction operator and the knowledge of the source signature was emphasized. Finally, [6] reformulated this theory into an adaptive procedure, in which the required source wave field is estimated together with the suppression of multiples. Also the first field dataset in a multi-channel process was published there.

![Image](image-url)

Fig.2: Zero offset section of a synthetic dataset with multiples, resembling a typical Gulf of Mexico salt body situation. The data has been provided by SMAART J.V.

The prediction of surface-related multiple elimination (SRME) has been formulated as an iterative procedure by [7] and demonstrated for some data examples by [8]. In this formulation, for measurements in a 2D seismic acquisition environment, the prediction of surface-related multiples is given by:

\[
M^{(n)}(\omega, x_r, x_s) = \sum_{x_i} P^{(n-i)}_{0}(\omega, x_r, x_i) P(\omega, x_i, x_s),
\]

in which \(P(\omega, x_r, x_s)\) is the seismic data measured at receiver \(x_r\) and a source at \(x_s\) in the frequency domain, \(P^{(n)}_{0}(\omega, x_r, x_s)\) is the estimate of the primary data after \(n\) iterations and \(M^{(n)}(\omega, x_r, x_s)\) are the predicted multiples for this iteration. In the first iteration, the primary estimate \(P^{(n=0)}_{0}(\omega, x_r, x_s)\) is taken to be the input data \(P(\omega, x_r, x_s)\). In each iteration the predicted multiples are matched with the true multiples using a matching filter \(A(\omega)\) (see [8]):

\[
P^{(n)}_{0}(\omega, x_r, x_s) = P(\omega, x_r, x_s) - A(\omega)M^{(n)}(\omega, x_r, x_s).
\]

Typically, one or sometimes two of these iterations, according to the two above equations, are sufficient. This formulation assumes that the measured data contain the upgoing pressure wave field due to a dipole pressure source. If this is not the case, correction factors need be included.

This surface-consistent convolution, as described in equation (1), is illustrated with a synthetic dataset. This data, from which a zero offset section is shown in Figure 2, is provided by the SMAART J.V. and is based on a typical Gulf of Mexico situation: a complex shaped, high velocity salt body embedded in lower velocity sediments. Note the depth of the water bottom, which results in the first order multiples to arrive at 5.0 seconds at the left hand side of the section.
Fig. 3: Prediction of multiples for one output trace as cross-convolution of traces from a common receiver gather (a) with the traces from a common source gather (b) at the corresponding surface locations, according to Figure 1b. Stacking of the convolution results in panel c) gives the predicted multiples for one output location, related to the receiver in panel a) and the source of panel b).

Fig. 4: Result of surface-related multiple suppression for a few selected shot records from the synthetic data related to Figure 2. Note that the measurements from many shot records are involved for the multiple removal for one input shot gather.

Figure 3 displays the result of the surface consistent convolution for one source (shot number 200) and one receiver location. The sum of the convolution results of Figure 3c will result in the multiples for the middle trace of shot record 200 in Figure 4a. Repeating this procedure for all source and receiver combinations and applying an adaptive subtraction process, yields the multiple suppressed shot records of Figure 4b.
3. FIELD DATA EXAMPLE

In Figure 5 an example of the application of the SRME method to field data from the Gulf of Mexico is shown. Figure 5a displays the stacked section with multiples. Note the offset in the time axis, because this is a deep water situation. The first order surface multiples arrive around 4.0 seconds in the middle part. Figure 5b shows the stacked section after multiple removal. Note the reduction of multiples. The removed multiples are displayed in Figure 5c. Also note the effect of small variations in the water bottom geometry and reflectivity, which are amplified in the multiples. They are visible as vertical zones of alternating high and low amplitude.

Fig. 5: Application of SRME to field data from the Gulf of Mexico. a) Input stack with multiples. b) Result of multiple removal. c) Difference section, i.e. the removed multiples only. All figures have been plotted at the same amplitude scale.
4. EXTENSION TO 3D

So far the SRME method was applied in a 2D sense to 2D seismic data. However, the earth is always 3D and applying a 2D method involves the assumption that the properties of the earth are invariant in the perpendicular (i.e. the cross-line) direction. This is called the 2,5D approximation (see [9]). However, the SRME methodology can also be applied to 3D measurements in a full 3D sense. This requires measurements to be available on a dense areal grid of sources and receivers. Equation (1) can then be rewritten as:

\[
M^{(n)}(\omega, x_r, y_r, x_s, y_s) = \sum_{y_i} \sum_{x_i} P_0^{(n-1)}(\omega, x_r, y_r, x_i, y_i) P(\omega, x_s, y_s, x_i, y_i),
\]

where the coordinates of the measurements are defined in the \(x\)- and \(y\)-direction at the surface. Note that in practice such a dense sampling in sources and receivers is never achieved. Therefore, some sort of data interpolation methods needs to be incorporated for getting sufficiently dense sampled data, such that the double summation in equation (3) can be carried out without introducing aliasing artefacts ([10],[11]).

![Field data example used to compare 2D SRME with 3D multiple prediction](image)

Figure 6: Field data example used to compare 2D SRME with 3D multiple prediction. This represents a zoom of a time section from deep water data offshore Brazil. a) Window of the input data which contains multiple reflections. b) Result of 2D multiple prediction. c) Result of 3D multiple prediction. The arrows point at several multiple events, from which it can be observed that the 3D prediction results have significantly better timing accuracy. (Results provided by PGS)

A field data example is given in Figure 6, which shows the multiple prediction results for a zoom of a time section from a deep water dataset acquired offshore Brazil. Due to the irregular water bottom shape, strong 3D effects are present in the surface-related multiples. The multiples predicted by 2D SRME (Figure 6b) and by the 3D method employing sparse inversion (Figure 6c) are displayed. When comparing both versions of predicted multiples with the multiples visible in the input data (Figure 6c) it can be observed that the 2D
predictions show quite a large time shift on the multiples, whereas the timing of the 3D prediction result is more accurate. The arrows point at locations where this effect is well visible. Besides the time shifts of the 2D predicted multiples, it can also be observed that the overall shape and lateral character of the multiples are better represented in the 3D prediction result.

5. CONCLUSIONS

For the situation of multi-channel measurements in a multi-coverage acoustic reflection experiment, surface-related multiples can be predicted and subtracted by using the data itself as the multi-channel prediction operator. The procedure involves a surface-consistent multi-dimensional convolution of the seismic data with itself, followed by an adaptive subtraction process. Good results are obtained for synthetic and field data, both for 2D and 3D implementations.

6. ACKNOWLEDGEMENTS

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REFERENCES

Structured Session 3

Technology and Experiments in Long Range Propagation

Organizer: A. B. Baggeroer
Internal wave effects on shallow water acoustic propagation: Relative influences of linear stochastic waves and nonlinear waves

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Abstract: Much attention has been paid to the effects of shallow water nonlinear internal waves on acoustic variability. Lesser attention, however, has been given to the influences of the stochastic linear internal wave field, whose effects on mode coupling have been shown to be rather weak given the short distances involved in shallow water propagation. In this talk it is shown that both internal wave fields can have a significant effect on shallow water acoustic variability, and a useful coupled mode model that includes effects from both internal wave fields can be used to predict important acoustic field second moments like mean intensity and coherence. A dedicated effort to quantify the spectrum of random shallow water internal waves, including anisotropy is needed.
A NUMERICAL STUDY ON 3D BROADBAND SOUND PROPAGATION AROUND A CONICAL SEAMOUNT

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Abstract: In this paper, three-dimensional (3D) broadband sound propagation around a conical seamount is investigated numerically using the 3D spectral coupled-mode model (W. Luo, PhD. Thesis, MIT, 2007). The broadband pulses are generated using the Fourier synthesis technique. Since the calculation of 3D broadband pulses with the spectral coupled mode theory requires extensive computation time, parallel programs were developed with cluster computing systems to obtain results in a reasonable time. The numerical results are given for the Kermit-Roosevelt seamount problem, which is modelled by a simple conical seamount, as well as for a benchmark problem with significant blockage effects. Due to the limited computational ability, a 15Hz broadband source with 10Hz bandwidth is explored. The computed pulse arrivals show clear shadow zones and the reappearance of the convergence zone behind the seamount with the complicated reflected wave patterns.

Keywords: Parallel computing, three-dimensional sound propagation, broadband pulse, seamount, shadow zone, convergence zone

1. INTRODUCTION

Although long-range ocean acoustics has advanced to predict the pulse arrival time within a millisecond at megameter ranges, the application of acoustic models to the long-range acoustic propagation has been limited to two-dimensional (2D) or Nx2D models due to the complexity of the problem as well as computational efficiency. Therefore, ocean waveguides with strong azimuthal coupling still remain highly challenging because a full 3D model is
required. The acoustic propagation around seamounts is a good example of the problem. In addition, uncertainties from oceanographic variability—e.g., sound speed variability due to the internal wave, and the geoaoustic property of sea bottom—increase the complexity of modelling acoustic propagation. Due to the complexity of the problem, acoustic propagation around seamounts is not well understood; however, physical experiments and 3D models have been explored [3, 4].

Chapman and Ebbeson [1] measured acoustic shadowing of 10~15dB behind the Dickins Seamount in a relatively short range experiment performed in 1975; the acoustic source was at shallow depth and the convergence zone was blocked by the seamount. In 2004, the Basin Acoustic Seamount Scattering Experiment (BASSEX) was conducted in the North Pacific as a part of long-range ocean acoustic propagation experiments of NPAL04 (North Pacific Acoustic Laboratory 2004) [3]. The BASSEX experiment was focused on the bathymetric effects on acoustic propagation, and in particular, direct blockage, refraction, diffraction, and scattering by seamounts. Moored and ship deployed acoustic sources transmitted m-sequence signals at about 192 dB re 1μPa including two SPICEX sources which transmitted eleven 12.3 second sequences every hours at 250Hz carrier frequency (83Hz bandwidth), and a LOAPEX source which transmitted forty-one 30 second sequences at 68.2Hz carrier frequency (35Hz bandwidth). Figure 1 shows the measured bathymetry around the Kermit-Roosevelt and Elvis seamounts with the source locations [3]. The distances between the SPICEX sources to the Kermit-Roosevelt seamount are 617 and 504km, respectively.

Fig. 1: Bathymetry around the Kermit-Roosevelt and Elvis seamounts with the source locations (S1 & S2 for the SPICEX sources, LOAPEX (T1000) for the LOAPEX source).

The Five Octave Research Array, which is a towed hydrophone array with 64 sensors cut for 250Hz (3m spacing), was used to measure the signals transmitted from the aforementioned broadband sources at many locations around the Kermit-Roosevelt and Elvis

Fig. 2: Measured peak sound levels received from SPICEX source S1 (left panel) and S2 (right panel) [3].
seamounts. Utilizing the measured broadband signals from the towed array, the size of the shadow zone was obtained and is shown in Fig. 2. Figure 2 shows the measured peak sound levels from the SPICEX sources; deep shadow zones as well as the formation of convergence zones are clearly visible behind the seamounts.

In addition, pulse arrivals from the BASSEX experiment database and 2D simulations were reconciled for the SPICEX sources using Parabolic Equation (PE) method and ray tracing method [3, 4]. In the reconciliation, acoustic ray arrival time and grazing angle prediction accuracy were examined. Refracted rays in the convergence zone were reconciled well with the 2D models, as expected; however, the complicated acoustic arrival patterns due to the reflected acoustic rays showed less agreement with simulated results.

In this paper, 3D broadband sound propagation around a conical seamount is investigated numerically using the 3D spectral coupled-mode model [2]. The broadband pulses are generated by the Fourier synthesis technique. Since the calculation of 3D broadband pulses with the spectral coupled mode theory requires extensive computation time, parallel programs were developed with cluster computing systems to obtain results in a reasonable time. The numerical results are given for the Kermit-Roosevelt seamount problem, which is modelled by a simple conical seamount, as well as for a benchmark problem with significant blockage effects. Due to the limited computational ability, a 15Hz broadband source with 10Hz bandwidth is explored.

2. THEORY AND NUMERICAL IMPLEMENTATION

A more stable and numerically effective 3D spectral coupled-mode model was proposed by Luo [2] using the superposition representation of the external fields of the seamount and the two-way marching approximation. However, this model still requires improvement in computational efficiency to realize the broadband pulse simulation. In this work, the computational efficiency is increased: first, by the varying number of the azimuth and/or normal modes along the sections of the seamount and, second, by parallel computing.

In the two-way marching approximation, the azimuth modal cut-off [4] takes place as well as the normal mode cut-off. In the upslope sound propagation, the trapped modes are lost to continuous modes as the water depth decreases while marching inward, called the normal mode cut-off. Similarly, the azimuth modal cut-off happens due to the decrease of distance from the peak of the seamount.

Two programs using the parallel computing were developed in Fortran 95 and MATLAB\textsuperscript{\textregistered}; the programs were developed with widely used MPI (Message Passing Interface) libraries, openMPI, mpich2 and pMatlab. The pMatlab is a MATLAB library developed by the MIT Lincoln Laboratory. The numerical results using these programs were obtained using the cluster computer at MIT and the LLGRID system at the MIT Lincoln Laboratory. Since solutions from the spectral coupled-mode model at azimuth modes are independent of each other, the parallel computing can be easily implemented in terms of azimuth modes.

A broadband signal can be obtained from a Fourier transform of the frequency-domain solution over the source bandwidth [5].

3. NUMERICAL RESULTS

Here we consider a benchmark problem in a deep sea waveguide similar to the configuration of the BASSEX experiment. The schematic of the problem is given in Fig. 3.
The radius of the conical seamount at the base is 20km with 3800m of height; the slope of the seamount is 10.76 degrees. A flat bottom is assumed at the depth of 5000m for the outside of the seamount. The source depth is 100m, and the center frequency of the source is 15Hz with 10Hz bandwidth (Fig. 4). Range-independent sound speed is assumed as shown in Fig. 3, and the geoacoustic properties of the bottom are a compressional sound speed of 2000 m/s, a density of 1.0 g/cm³ and an attenuation of 0.1 dB/λ. The false bottom was introduced below the sea bottom to suppress the artificial reflections from the boundary. The angle between the centerline and the outermost ring with respect to the source is 11.31 degrees. The seamount is located at 100km from the acoustic source so that the significant refracted rays are all blocked by the seamount along the centerline as shown in Fig. 5; this placement can maximize the blockage effect by the seamount.

![Image: Schematic of deep sea waveguide with a conical seamount.](image1)

![Image: Broadband acoustic source with the center frequency of 15Hz (left panel) and the number of normal modes for the source bandwidth (right panel).](image2)

**Fig. 3:** Schematic of deep sea waveguide with a conical seamount.

**Fig. 4:** Broadband acoustic source with the center frequency of 15Hz (left panel) and the number of normal modes for the source bandwidth (right panel).

Figure 6 shows the transmission loss (TL) for a 10Hz continuous wave (CW) source at the depth of 300m. The left panel compares the TL from the 2D coupled normal mode [6] and 3D spectral coupled-mode model along the centerline; the TL from 3D are greater than that from 2D by 5–10dB, except the convergence zone. This phenomenon can be explained by the fact that energy is dissipating outward from the centerline due to the horizontal refraction in the 3D sound propagation. The TL on a horizontal plane in the right panel show clear shadow zones behind the seamount with weak appearance of the convergence zones.
Pressure fields are computed in 3D for 165 frequencies within 10Hz bandwidth (10 to 20Hz), and the transfer function is synthesized with the source spectrum given in Fig. 4 using the inverse Fourier transform, which results in a 16.4 second period time series. For the solution at each frequency, the propagation modes in the outermost ring are used, which vary from 42 to 87 modes at frequencies of 10~20Hz (Fig. 4).

Figure 7 shows the pulse arrivals for several acoustic paths in reduced time with a reference time of 1520m/s. The angle of the acoustic path is defined with respect to the centerline. The refracted rays without bottom interactions are all blocked behind the seamount up to 10 degrees; however, the conversion of the reflected rays into the refracted rays after the seamount generates weak convergence zones. The disturbances caused by the bottom bouncing waves due to the seamount can be found at much higher angles up to 16 degrees, as shown in the TL for a CW source (Fig. 6).

Figure 8 shows the pulse arrivals at a depth of 200m on horizontal plane; the reduced time with a reference speed of 1520 m/s is shown in the corner of each panel. The first convergence zone appears before the seamount at 0.75 seconds, 60km from the source. The
second, third, and fourth convergence zones outside the perturbation zone develop at 1.25, 1.75 and 2.5 seconds, 120km, 190km, and 250km from the source, respectively, with clear shadow casting inside the perturbation zone. At around 2.25 seconds, between the third and fourth convergence zone appearances, a convergence zone is formed by the reflected-refracted rays within the perturbation zone. Reflections due to the seamount generate quite complicated patterns of pulse arrivals in and around the shadow zone, which includes waves going away from the seamount to higher angles.

An approximated conical seamount based on the Kermit-Roosevelt seamount is given in Fig. 9 with the averaged sound speed profile. The same broadband acoustic source was considered. The pulse arrivals at a depth of 200m on horizontal plane are given in Fig. 10 in terms of the reduced time with a reference speed of 1520 m/s. In the figure, the shadow casting due to the seamount is clear, but the width of the shadow zone is narrower than that from the previous problem because the distance from the source to the seamount is increased to 512km. The convergence zone at 10.2 seconds is formed by the refracted rays passing over the seamount and joining with those without disturbances outside the seamount.

4. CONCLUSIONS

The 3D broadband pulses were obtained successfully around a conical seamount based on the 3D spectral coupled-mode model using the Fourier synthesis technique. For the 3D broadband pulse realization, efficient programs were developed using the varying number of the azimuth and normal modes, as well as parallel computing. The programs were applied to the Kermit-Roosevelt seamount problem, which is modelled by a simple conical seamount, as well as to a benchmark problem with significant blockage effects. The computed pulse arrivals show clear shadow zones and the reappearance of the convergence zone behind the seamount with the complicated reflected wave patterns.

In this work, a 15Hz broadband source was used due to the limited computational ability; therefore, more efforts for higher computational efficiency are required to apply this model to higher frequency – ideally, up to the 68Hz, the carrier frequency of LOAPEX source - to reconcile the measured data from the BASSEX experiment and the computed pulse arrivals.

5. ACKNOWLEDGEMENTS

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Fig. 7: Computed pulse arrivals along the acoustic paths at the depth of 200m. The angles are defined with respect to the centerline.

Fig. 8: Computed pulse arrivals on the horizontal plane at a depth of 200m in terms of the reduced time.
Fig. 9: Schematic of a conical seamount for the modelled Kermit-Roosevelt seamount with the averaged sound speed profile.

Fig. 10: Computed pulse arrivals on the horizontal plane at a depth of 200m.


Deep Water Mode Processing

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Abstract: In deep water the low modes are associated with energetic arrivals at long range. The low mode signals are of interest in acoustic tomography applications because they provide information about the environment around the sound channel axis. Since 1995, experiments such as ATOC, NPAL98, and SPICE04/LOAPEX provided an opportunity to study mode propagation in the North Pacific at ranges from 50 km to 5 megameters. These experiments used equally-spaced vertical line arrays with apertures of up to 1400 m to sample the axial modes. Upcoming experiments in the Philippine Sea will use new array technology to measure the mode signals. The Distributed Vertical Line Array (DVLA) [Worcester, et al., UAM 2009] facilitates sampling of a 6000 m water column with arbitrary sensor spacing. With its long aperture, the DVLA can provide substantially better mode resolution than previous arrays, but it also presents some new design challenges. First, since the DVLA's architecture does not require equally-spaced hydrophones, it is important to investigate how sensor placement can be optimized to improve the processing gain for the mode signals. Second, mode processing involves coherent combination across an entire aperture. Since the DVLA can span the full water column, the mode processor will be sensitive to phase deviations (due to residual mooring motion or environmental mismatch) over a 6000 m span. Third, the large number of sensors in the DVLA implies that a large number of snapshots will be required to implement adaptive mode processing. Given that the Philippine Sea is a dynamic ocean environment, obtaining a large number of stationary snapshots may be challenging. This paper explores the problems of sensor placement, coherent processing, and data snapshots in order to design appropriate mode estimation techniques for the DVLA.
DISTRIBUTED VERTICAL LINE ARRAY (DVLA) ACOUSTIC RECEIVER

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Abstract: A Distributed Vertical Line Array (DVLA) receiver able to span the full water column in water up to 6000 m deep has been developed to allow both modal and ray-based analyses of acoustic propagation. The DVLA is made up of distributed, self-recording hydrophones with timing and scheduling provided by a small number of central controllers, called D-STARs. The enabling technologies for this approach are (i) the availability of flash memory modules that can store several gigabytes of data and be located in a small pressure case at each hydrophone, making it unnecessary to transfer data from the hydrophones to the central controllers for storage, and (ii) inductive modems that allow low-bandwidth communication for command, control, and time synchronization between the D-STAR controllers and the Hydrophone Modules over standard oceanographic mooring wire. The DVLA is modular, consisting of subarrays with a nominal length of 1000 m, each of which has one D-STAR controller and up to 99 Hydrophone Modules. A D-STAR controller is shackled in-line at the top of each subarray. The Hydrophone Modules are clamped to the mooring wire at the time of deployment, making the DVLA readily configurable for different experiments. The D-STAR clocks are synchronized acoustically, avoiding electrical interconnections. The DVLA is navigated using acoustic transponders positioned on the seafloor around the mooring. The Hydrophone Modules make precision temperature measurements throughout the time that the DVLA is deployed, in order to provide the sound-speed profiles needed for beamforming.

Keywords: Array technology, sound propagation in deep water
1. INTRODUCTION

Uncertainty due to ocean variability limits the ability to make accurate predictions of acoustic propagation. Scattering due to internal waves and other ocean processes limits the temporal and spatial coherence of the received signal. Recent experiments to study the basic physics of low-frequency, long-range acoustic propagation in deep water and the effects of environmental variability on signal stability and coherence have been constrained by the lack of vertical line array (VLA) receivers capable of spanning the full water column in deep water. Such arrays are required to enable the separation of acoustic modes using spatial filtering and to fully characterize the acoustic time fronts formed in deep water propagation.

The earliest low-frequency, long-range acoustic propagation experiments, starting with the discovery of the deep sound channel in 1944, used wideband explosive sources. These experiments formed the basis for understanding propagation through a deterministic ocean and have shed some light on the statistics of its variability. But it is the development over the last two decades of low-frequency, wideband acoustic sources driven by controlled waveforms and large vertical receiving arrays, including ones that can store a year or more of data, that have finally provided the means to measure the spatial and temporal statistics of the acoustic fluctuations [1]. This technology allows individual multipaths and normal modes to be resolved and long time-series to be collected. These statistics are needed to advance the understanding of the effects of ocean variability on acoustic propagation.

Fully autonomous VLA receivers originally developed in the early 1990s for the Acoustic Thermometry of Ocean Climate (ATOC) project have been the workhorses in deep-water, long-range ocean acoustic propagation experiments for over a decade [1]. These arrays, which were developed to test and improve the understanding of acoustic propagation out to ranges of 3–10 megameters, were first used in the ATOC Acoustic Engineering Test in 1994 [2]. The ATOC VLAs consisted of two 20-element, 700-m long subarrays, forming a 40-element, 1400-m long array with hydrophones spaced 35 m apart [3]. It is difficult to deploy substantially longer VLA receivers using this technology, however, because of the weight of the complex electromechanical array cables, even though full-water-column spanning VLA receivers are sorely needed to further the understanding of deep-water propagation.

A modular, distributed VLA receiver that is capable of spanning the full water column in water up to 6000 m deep has therefore been developed to allow both modal and ray-based analyses of acoustic propagation in deep water.

2. DISTRIBUTED VLA (DVLA) ARCHITECTURE

The DVLA is made up of distributed, self-recording Hydrophone Modules with timing and scheduling provided by a small number of central controllers, called D-STARs (DVLA - Simple Tomographic Acoustic Receivers). The enabling technologies for this approach are (i) the availability of flash memory modules that can store several gigabytes of data and be located in a small pressure case at each hydrophone, making it unnecessary to transfer data from the hydrophones to the central controllers for storage, and (ii) inductively coupled modems that allow low-bandwidth communication for command, control, and time synchronization between the central controllers and the Hydrophone Modules over standard oceanographic mooring wire. The DVLA is fully autonomous, recording during time periods specified a priori. The maximum duty cycle for a one-year deployment is approximately 8%, depending on the sample rate.
The DVLA consists of subarrays with a nominal length of 1000 m, each of which has one D-STAR controller and up to 99 Hydrophone Modules. The D-STAR is shackled in-line at the top of each subarray. Hydrophone Modules are clamped to the mooring wire at the time of deployment, making the DVLA readily configurable for different experiments. The D-STARS and Hydrophone Modules are rated for operation to a depth of 6000 m. The DVLA uses a long-baseline acoustic navigation system to measure the array position and shape as the mooring moves in response to ocean currents.

2.1. Inductive Modem

The data link connecting the Hydrophone Modules to the D-STAR controller employs commercially available inductive modem technology that uses standard 3x19 jacketed oceanographic wire rope, avoiding the need for a custom electromechanical array cable [4]. Inductively coupled modems use toroidal transformers to couple data to and from instruments clamped on the mooring cable. At each Hydrophone Module, the mooring cable passes through a ferrite toroid, forming a single-turn primary winding for data transferred from the D-STAR controller to the Hydrophone Module. The Hydrophone Module has a 20-turn secondary winding to receive the data. The toroids are split in halves, so that they can be clamped around the cable without the need to thread the mooring cable through the toroid. A seawater return is used to complete the electrical circuit. The swaged fitting on the end of the mooring cable distant from the D-STAR controller provides one seawater ground.

Although the D-STAR could also use a toroidal transformer to couple to the mooring cable, it is more convenient to use a small transformer internal to the pressure case that is connected to the end of the mooring cable through an underwater connector. In this case, the end of the mooring cable is isolated from seawater and a clevis fitting is used to provide both the mechanical termination and the seawater ground.

![Fig. 1: Histogram of the time differences (μsec) between two Hydrophone Modules with 100-turn windings on the inductive couplers.](image)

The DVLA uses Inductive Modem Modules (IMM) and other components supplied by Sea-Bird Electronics, Inc. [4]. Sea-Bird’s inductive modems use Differential Phase Shift Key
(DPSK) modulation on a 4800-Hz carrier, providing 1200-baud, half-duplex communications. The system is not optimized for time transfer, however. Sea-Bird therefore modified the IMM firmware to provide precision time transfer using the IM Flag line. Laboratory tests indicate that the modified firmware provides time transfer with a standard deviation of 31 μsec between Hydrophone Modules (Fig. 1). The absolute time delay from the D-STAR controller to the Hydrophone Modules with a loop resistance of 100 (1000) ohms is 9.34 (9.54) ms.

2.2. D-STAR Controller

The D-STAR controller is based on the Simple Tomographic Acoustic Receiver (STAR) data acquisition system and acoustic source controller previously developed at Scripps Institution of Oceanography for ocean acoustic tomography (Fig. 2). The STAR provides a precision time base capable of keeping time autonomously with an accuracy of about 3 ms per year. A low-power (80 mW) microcomputer compensated crystal oscillator (Q-Tech MCXO QT-2002) serves as the primary system oscillator. A much higher power (10 W @ 25°C operating) rubidium oscillator (Symmetricom X72) is turned on periodically (typically once per day) to serve as a reference to measure the frequency of the QT-2002. The difference frequency is logged and used to correct the STAR clock after recovery.

![Fig. 2: D-STAR controller and a Hydrophone Module being deployed during the 2009 North Pacific Acoustic Laboratory (NPAL) Philippine Sea Pilot Study/Engineering Test (PhilSea09). (Photograph by L. Green, Scripps Institution of Oceanography.)](image)

The STAR also includes a long-baseline acoustic navigation system to measure mooring motion with an absolute precision of approximately 1 m rms, using acoustic transponders deployed on the seafloor. The D-STAR uses a single-frequency interrogate (9.0 kHz), multiple-frequency reply (11.0, 11.5, 12.0, and 12.5 kHz) system. The D-STAR controllers typically transmit interrogation signals once per hour continuously throughout the experiment and record the transponder replies using a 15-s recording window.

Finally, the STARs provide a RS-232/RS-422 serial port for an external modem. In the D-STARs, an IMM and transformer are added to the STARs, the underwater modem connector
is used for the connection to the mooring cable, and the D-STAR firmware is modified to provide control and timing signals to the Hydrophone Modules over the inductive modem communication link.

There are no electrical connections between the D-STAR controllers. Their clocks are therefore synchronized acoustically using the long-baseline interrogation signals. Each D-STAR records the interrogation signals of the other D-STARs. This is equivalent to having the D-STARs transpond off one another (although with some delay between the interrogate and reply signals) and allows the D-STAR clocks to be compared.

2.3. **Hydrophone Modules**

The Hydrophone Modules are designed (i) to record low-frequency acoustic signals, sampling at 976.5625, 1953.1250, or 3906.2500 Hz, (ii) to record the high-frequency, long-baseline acoustic navigation signals, sampling at 39,062.5 Hz, and (iii) to make precision temperature measurements (± 0.005°C). Separate channels of a quad, 24-bit, delta-sigma analog-to-digital converter (Texas Instruments ADS1274) are used for each of the three measurements. The signal conditioning and anti-aliasing filters differ from channel to channel, as appropriate. Each Hydrophone Module writes the data to a 16 GByte Secure Digital (SD) card. The SD cards are removed from the Modules following the experiment in order to download the data.

The same hydrophone (High Tech, Inc. HTI-90-U), which is rated for operation from 2 Hz to 20 kHz, is used for both the low-frequency acoustic and navigation channels. It has an integral preamplifier. Its equivalent self-noise is below Sea State 0 (Table 1). The self-noise levels of the signal conditioning circuitry and ADC in the low-frequency acoustic channel are below the equivalent self-noise of the hydrophone.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>Equivalent Self-Noise (dB re 1 μPa / √Hz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>10</td>
<td>54</td>
</tr>
<tr>
<td>100</td>
<td>35</td>
</tr>
<tr>
<td>1000</td>
<td>26</td>
</tr>
</tbody>
</table>

*Table 1: Equivalent self-noise vs. frequency for the HTI-90-U hydrophone.*

High-pass filters are included in both the hydrophone preamplifier ($f_{High-pass} = 10$ Hz) and Hydrophone Module signal conditioning circuitry ($f_{High-pass} = 7.7$ Hz) to mitigate the effects of low-frequency mooring strum due to vortex induced vibrations (VIV).

The Hydrophone Modules make precision temperature measurements throughout the time that the DVLA is deployed, in order to provide the sound-speed profiles needed for beamforming.

The Hydrophone Modules are normally completely powered off, except for the Inductive Modem Modules, which are in a micropower state waiting for a wake-up signal from the D-STAR controller. After broadcasting a wake-up signal, the D-STAR waits for the oscillators (20-MHz Maxim DS4026 TCXO) in the Hydrophone Modules to warm up prior to setting the clocks in the Modules and sending them instructions.

Each Hydrophone Module includes a USB port for programming and testing. Each Module operates from a single 40-Amp-hour lithium battery (Electrochem, Inc. 3B6100, Size TSD).
3. ACKNOWLEDGMENTS

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Towed CTD Chain Data Collection and Acoustic Propagation Predictions for the South China Sea

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Abstract: A number of conductivity, temperature and pressure (depth) (CTD) Chain tows were made over a flat, shelf region in the South China Sea approximately 115 nm southeast of the Korean Peninsula. Water depth of the area of covered varied from 65 to 80 meters and the site was located approximately 9 nm from a mild shelf break. Simultaneous acoustic data was collected using multiple sources and two acoustic arrays. Relative to the acoustic propagation paths, the CTD data collection tracks are both perpendicular and parallel. The CTD data set will be described in detail, together with acoustic propagation predictions for later comparison with the recorded acoustic data.
Towed Array Signal and Noise Measurements in the Philippine Sea

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\textbf{Abstract:} During the coordinated Philippine Sea 2009 experiment, conducted by the North Pacific Acoustic Laboratory working group, a 200m towed array (the ONR Five Octave Research Array) was used to measure signals and noise in deep water in the Philippine Sea. Broadband transmissions (225–325 Hz, 135-s LFM) were recorded from a Webb Research Corporation (WRC) sweeper source moored at the sound channel axis (1050m) in 6059 m of water at 22.960°N, 126.563°E. Receptions were recorded from the near field of the source out to 200 km to the southwest. Transmissions from an axially suspended ship source at three other locations were also recorded. Shallow source transmissions were emitted from a ship suspended source. These transmissions contained a comb of narrowband lines as well as period LFM signals. Data are being analyzed to study the evolution of deep-water pulses with range, the stability and shape of the convergence zone, deep-water geo-acoustic inversion and the feasibility of performing tomography with a towed array. In addition to signal measurements, many hours of towed array ambient noise measurements were taken to examine the structure and temporal statistics of the ambient noise field.

\textbf{Keywords:} Towed array measurements, tomography, deep water acoustics, ambient noise
1. INTRODUCTION

The North Pacific Acoustic Laboratory team has been conducting deep-water, long-range acoustic measurements dating back to the Heard Island Feasibility Test\(^1,2\). Experiments have included ATOC and more recently the NPAL experiment\(^3,4\) in the eastern North Pacific. In April-May 2009, as a pilot test for an extended 2010 experiment, a moored source, ship-suspended sources, a Distributed Vertical Line Array (DVLA) receiver, and a towed array receiver were deployed in the Philippine Sea to study propagation at ranges up to 200 km in a deep-water, active-mesoscale environment. As part of this experiment, sound transmissions from a moored axial source and a ship-suspended shallow source were recorded on a 200m nested aperture towed line array. The experiment, unfortunately, was conducted between the deadline for the inclusion of papers in this conference proceedings (April 2009) and when the UAM conference was actually held (June 2009). In this paper, we therefore present the science goals and anticipated measurements of the experiment. At the Underwater Acoustic Measurements meeting, experimental results and initial conclusions will be presented.

Measurements from three sources (two at the axis and one shallow) and ambient noise will be examined in this paper. The locations of the moored source and the primary propagation path are shown in Fig. 1. The axial moored source, labelled T1 for inclusion in planning of the full 2010 experiment, which will include six tomographic transceivers, is located at 22.960°N, 126.563°E. A ship-suspended source was used within 100 km of the location marked DVLA in the figure. The DVLA is a pair of vertical line arrays used for tomography and mode filtering. This array is not the focus of this paper, though it marks the locus of measurements conducted with the ship-suspended source and the towed line array.

From the bathymetry in Fig. 1, we see that the bathymetry for the entire propagation path T1-DVLA is greater than 5000 km. The ridge at 126.5E, 22N will provide the opportunity, using the mobility of the towed array receiver, to measure scattered and truncated ray-paths.
The FORA array is a 200 m nested five-octave array with continuous recording capabilities from 2 Hz to 20 kHz. The focus of this experiment will be in the band 20 Hz to 500 Hz. The array was towed at nominally 3 kts (1.5 m/s) from the R/V Kilo Moana. For the deep source transmissions, the array depth was set to 300 m (deepest operating depth) in order to receive energy from the deepest possible rays. For the shallow source measurements, the array was set as close to the source depth as possible to achieve maximum focusing.

The science goals associated with the towed array portion of this experiment include the evolution of the tomographic signal with range (deep source), the feasibility of conducting towed array tomography measurements (deep source), the stability and shape of the convergence zone (shallow source), deep water geo-acoustic inversion and coherence length of bottom bounce paths (both sources), and ambient noise spatial structure and temporal variability.

2. AXIAL SOURCE PROPAGATION

Two axial sources will be used during this experiment. The first axial source is moored at the sound channel minimum (~ 1050 m) in roughly 6000 m of water. The WRC sweeper source transmits a 135-s LFM with a frequency range from 225 to 325 Hz. The duty cycle is one transmission every 3 hours for tomographic time periods and one transmission every 5 minutes for high-resolution measurements, conducted over two 24-hour periods and one 72-
hr period during the three-week test. The second sound source is an HX-554, which will be suspended at the axis at two locations (55 and 110 km from the DVLA site along the T1 radial). The ship-suspended deep source will transmit m-sequences centered at 81.76 Hz continuously for approximately 24 hours at each stop.

The narrowband TL for the axial moored source is shown in Fig. 2. This is easily interpreted using ray-theory, showing a separation of high-angle and low-angle rays (modes). From this figure, it is clear that non-bottom interacting energy will only be observed on the towed array in pairs at 55, 65, and 115, 130 and 175, 190 km. Bottom interacting energy will be observed at ranges less than 100 km, depending on the sediment reflection properties.

![300-Hz Axial Source TL](image)

**Figure 2.** 300-Hz Narrowband Transmission Loss, using the Parabolic Equation method. Note the shadow zones and convergence zones that will be encountered by an array towed at 300m. Also note the presence of bottom-interacting energy out to 100 km.

The TL field in Fig. 2 demonstrates several features of deep-water acoustic propagation that will be studied in this test. The first is the location and shape of convergence zones and the range/depth of turning rays. The structure of bottom bounce energy will be examined out to ranges of 100 km to examine bottom/sub-bottom roughness and deep-water geo-acoustic properties. From Fig. 1, we see that there is a seamount chain with minimum depth extent on the order of 4000 m just off the T1-DVLA path. Ray-path blockage and scattering from this bathymetric feature will be examined.

### 3. SHALLOW SOURCE PROPAGATION

Shallow source propagation in deep water is dominated by the convergence zone phenomenon. In this environment, the water depth is greater than the source reciprocal depth giving a depth-excess (of roughly 500 m) leading to strong convergence zone propagation. The 300-Hz TL for this environment is shown in Fig. 3. A shallow receiver only receives sound along two primary paths in this environment. Refracted paths come to the surface every convergence zone (CZ), approximately 62 km. The second path is the bottom bounce (BB) path, which can be seen in Fig. 3 at ranges from 5-100 km.
Figure 3. Shallow source Transmission Loss (300 Hz) showing convergence zone (CZ) and bottom bounce (BB) paths.

The science objectives for this portion of the test are to study the predictability, structure and stability of the CZ and BB paths. The CZ path structure will depend upon local oceanographic variability. The BB paths will depend upon geo-acoustics as well as bottom roughness and will be significantly less sensitive to the oceanography due to the result from Snell’s law that steep rays do not refract as much.

4. AMBIENT NOISE

The third sets of measurements taken during the PhilSea09 experiment are ambient noise measurements. Many hours of towed-array ambient noise data will be collected at various locations (T1, DVLA), as well as array orientations. The emphasis will be on the analysis of the directionality and stationarity of the deep-water ambient noise field. The experiment path lies along the Philippines-Okinawa shipping lane. Local shipping traffic is expected to dominate the noise field at low frequencies. Even though traffic on this shipping lane is significantly less than along the Taiwan-Japan path to the north, it will be significant and is expected to dominate the shipping noise. Ambient noise can be thought of as consisting of three primary components: wind-noise, distant shipping noise and nearby interferer traffic. Wind-noise is primarily from overhead and is sensitive to local weather. Distant shipping can impact the overall stable background structure of the noise field, especially at low frequencies. Local interferer traffic dominates the temporal and spatial dynamics of the noise field on 10-30 minute time scales. Radar measurements will be simultaneously recorded to help sort out the effect of local interferers.

In Fig. 4, the 120-Hz TL from a 100-m deployed array at the DVLA site to the surface as a function of source position is plotted for the Philippine Sea. This illustrates the effect bathymetry has on the expected spatial distribution of distant noise. Near the receiver, convergence zone propagation dominates. There are regions to the south where bathymetric blockage should limit the impact of distant shipping. To the north, the continental slope prevents signals from effectively propagating from the East-China Sea out into deep water.
5. ACKNOWLEDGMENTS

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Structured Session 4

Habitat Mapping and Underwater Acoustics

Organizer: Philippe Blondel
Use of broadband acoustic scattering techniques for high-resolution imaging, detection, and quantification of physical and biological scattering features in the ocean.

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Abstract: Broadband acoustic scattering techniques, and associated signal processing techniques, are emerging as a powerful tool for the remote sensing of physical and biological scattering features in the ocean. Broadband measurements are beneficial as they 1) allow the frequency spectrum of oceanic scatterers to be measured, thereby resulting in enhanced classification potential relative to more traditional single-frequency scattering techniques, and 2) result in increased spatial resolution, in turn leading to high-resolution images of oceanic scattering features, obtained through use of pulse compression signal processing techniques that exploit the broadband content of the signal. Using two heavily modified, off-the-shelf side scan sonar systems from Edgetech, measurements of broadband acoustic scattering from physical and biological scattering features in the ocean have been performed over the frequency band from 1.5 kHz to 600 kHz, with some gaps. In this presentation, limits of single-frequency acoustic scattering measurements are discussed, and arguments are presented for the benefits of using broadband acoustics. Examples of broadband field measurements supporting these arguments are presented from different recent field experiments, and include zooplankton versus turbulent oceanic microstructure discrimination and quantification and resonance fish classification.
ANALYSIS OF SINGLE BEAM, MULTIBEAM AND SIDESCAN SONAR DATA FOR BENTHIC HABITAT CLASSIFICATION IN THE SOUTHERN BALTIC SEA

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Abstract: Benthic habitat characterization is important for the study and conservation of the biodiversity of the Baltic ecosystem. The main objective of this paper is the development of complementary acoustic techniques for monitoring of Baltic benthic habitats. The study area was located in the southern Baltic Sea and characterised by a considerable diversity of geomorphologic forms and benthic assemblages. The simultaneous registration of the acoustical data was conducted with two single-beam echosounders working at different frequencies, a multibeam echosounder and a sidescan sonar. The high resolution multibeam data were used to estimate seabed corrugation, a crucial feature for bottom surface characterization. To identify morphological forms and benthic habitats, a parametric approach was applied to the multibeam data. Firstly, spectral, wavelet, and fractal parameters were computed in windows sliding along the separated bathymetric transects. The vectors of computed parameters were then used as an input into Principal Component Analysis and subsequently to fuzzy C-means clustering classification system. Moreover, angular dependency of the backscattering intensity was analysed. Also the information from single beam echosounders and sidescan sonar was utilised. The classification algorithms were validated with video records and biological sampling.

Keywords: benthic habitats, multibeam echosounder, classification algorithms
1. INTRODUCTION

The acoustical maps, containing the information on the shape and geological nature of the seabed itself and the benthic marine organisms present, represent an essential tool for the conservation and management of the seafloor of the Polish Exclusive Economic Zone within the Baltic Sea and allow to predict accurately the impact of anthropogenic activities on the habitats. Taking it into account, the sets of acoustical backscattering data was collected by complementarily used different acoustical tools as multibeam echosounder, sidescan sonar, sub-bottom profiling system and single beam echosounders. Special attention was focused on the narrow euphotic zone of the depth between 4-20m elongated parallel to the Polish cost and containing different forms of benthic habitats. The total length of the surveyed area was about 220 km and of a width slightly above 1 km. For the habitat mapping purpose a special test polygon was chosen. The surveyed test site featured a diversified seafloor geomorphologic forms and associated habitats and was located 1.2 km NE of the Rowy harbour (see Fig.1.).

![Fig.1: The study area - measurement polygon located 1.2 km NE of the Rowy harbour](f=54°40'02″N, l=17°03'10″E).

In the Polish marine areas, bottom covered with boulders and pebbles is rare, and therefore the area of boulder field located near the Rowy harbor stands out against the practically bare of benthic fauna and vegetations Polish inshore zone. The high biological values of this area are undoubtedly a great impact on the varied morphology of abrasion platform with boulders and pebbles scattered over the surface of the bottom, allowing attachment of organisms [1]. Moreover, the small depth of the area provides favorable light conditions for plants and indispensable conditions for photosynthesis.

The acoustical measurements were accompanied by biological and geological sampling and video inspection. The classification methodology were concentrated on the multibeam echosounder data, which have been developed in two directions. The first “classical” method utilised the shape parameters of the angular dependency of the backscattering intensity, when the second method used the shape parameters of the bathymetric transects computed in a sliding window. The acoustical data were integrated with the collected biological and geological data in order to verify the developed classification algorithms.
1.1. MORPHOLOGY OF SEAFLOOR TEST AREA

The polygon is located in the inshore area within the zone of bottom relief having polygenetic origin with relics of periglacial forms together with contemporary forms of marine origin. The polygon depth varies from ~4m up to ~14m. The right map in Fig.1. shows the investigated area with depicted MBES imagery of measurement polygon.

The bottom surface is rough and varied with clearly formed embankment of ~300m in width stretching in the NW-SE direction. The highest part of the embankment at depths of about 4-7m, is an abrasive platform, with many young inselbergs and stony gravely abrasive pavement on the bottom surface. The embankment slopes are furrowed with numerous, relatively broad erosion gorges. The south slope is relatively short (up to 50m) and adjoins to the shore slope, which arose as a result of sand accumulation. The north slope 100m long at a depth of approximately 15m becomes a nearly flat surface of accumulated marine sands and fine-grained sands and muddy sands. This is an area of relict moraine embankment occurrence, made of till covered in stony gravely abrasive pavement and numerous relict and contemporary erosion gorges with surfaces covered with contemporary accumulative marine sands.

2. EXPERIMENT METODOLOGY

The measurements were conducted from the r/v “IMOR” equipped with precise navigational instrumentation, multibeam echosounder Reson 8125 (455kHz, range: 0.5m - 120m, no. of beams: 248, scan width: 120º , beam width: 0.5º), chirp dual frequency sidescan sonar EdgeTech 4200SP (300/600 kHz), single beam echosounders: Simrad EK-500 (120 kHz) and BioSonics DT-X (420 kHz). The USBL underwater positioning system was used to calibrate measurements, where the acoustical signals were backscatter from biologically recognized areas. Divers collected biological samples from the eight areas limited by frames (1x1m) and made video recordings of benthic habitats. Figure 2 shows the metallic frame surrounding area before pick up of biological samples (left photo) and after sampling (right picture). Based on information from divers, video and photographic

*Fig.2: The metallic frame surrounding biological measurement area before samples picking (left photo) and after samples collection (right photo). Transducer of the USBL positioning system is visible attached to the frame.*
documentation, laboratory analysis of benthic material and information from the literature [1], the characteristics of the individual stations were extracted.

It should be noted, that for the purpose of resolution enhancement, the number of acoustical transects exceeds the number which assures the needs of polygon area coverage. For that reason the spatial resolution of the bathymetric map obtained from multibeam echosounder measurements reaches 0.05m.

3. DATA PROCESSING

The MBES data processing delivers segmented maps of different geomorphologic and associated habitable areas. For this target’s realisation were made two classification algorithms based on different ideas – the parameterisation of the shape of angular dependency of backscattering intensity and the second method – the parameterisation of the high resolution bathymetric transects.

3.1. PARAMETRICAL ANALYSIS OF BOTTOM BACKSCATTER INTENSITY

The idea of MBSE seafloor classification based on angular dependency of the backscattered intensity features is known in several classification systems [e.g. 2,3,4,5]. In contrast to these systems, the method presented here utilises only the shape parameters of the backscattering intensity computed for the separated two sides of returning MBSE signals. For each backscattering intensity function were computed 26 spectral, fractal, and wavelet transformation parameters.

The normalized power spectrum of backscattering intensity angular dependency in logarithmic form is defined as [6]:

$$ C_j = \log_{10} \left( \frac{A \cdot S(\omega)}{S_{\text{max}}(\omega)} + 1 \right) / \log_{10} (A + 1), \quad (1) $$

where $A=10^5$ – const., $S(\omega)$ - power spectral density function and its maximum value $S_{\text{max}}(\omega)$. The classification parameters were defined as the relationships between parts of spectral density functions:

$$ S_{\text{r}} = \frac{1}{f_{\text{Ny}}} \int_{0}^{f_{\text{Ny}}} C_j \, df, \quad S_{\text{m}} = \frac{1}{S_{\text{r}}} \int_{0}^{\frac{1}{m} f_{\text{Ny}}} C_j \, df, \quad (2) $$

where $m=2, 4, 8, 16$ and $f_{\text{Ny}}$ is the Nyquist frequency.

The spectral moments of the $r$-th order are very sensitive for signal shape variation are defined as:

$$ m_r = \int_{0}^{\infty} \omega^r S(\omega) \, d\omega, \quad (3) $$

where $S(\omega)$ is the Fourier power spectral density with moment order of $r=0, 1, \ldots, 7$. The spectral widths $\sigma^2$, $\nu^2$ and spectral skewness $\gamma$ are defined as [7]:

\[ 134 \]
The other parameter based on power spectral density function is the fractal dimension computed from the spectrum slope $\beta$ and defined by Mandelbrot \[8\] as $D_{FFT} = (5 - \beta)/2$.

The next group of very useful parameters in data segmentation process are wavelet transformation coefficients and related wavelet energies. For the backscattering intensity signal, the wavelet energy content was computed using the 7-channel dyadic decomposition (scale $a = 2^j$, $j = 1, \ldots, 7$) with a 3rd-order Coiflet wavelet:

$$E_{j,\text{Coif} 3} = \int_{h_{\text{min}}}^{h_{\text{max}}} C^2(a, b) \, db,$$

where $C(a, b)$ is the wavelet transformation coefficient, $h_{\text{min}}$ and $h_{\text{max}}$ are boundary values of scale $b$ (time). The energy distribution indicator - entropy $h_{\text{Coif3}}$ is defined as \[9\]:

$$h_{\text{Coif3}} = \sum_{j=1}^{7} E_{j,\text{Coif} 3} \cdot \ln E_{j,\text{Coif} 3},$$

The technique of Hurst exponent determination via the averaged wavelet coefficient method \[10, 7\] was used in this calculation. The Hurst exponent $H$ and subsequently the Hausdorff dimension is equal $D_{H, \text{Daub7}} = 2 - H$. The above defined parameters formed 26-element vectors for either side of backscattering intensity. For elimination of the excessive fluctuation of parameters values the moving average procedure was used. The strong correlation between some of the parameters required the elimination of redundant information. A Principal Component Analysis (PCA) has been applied to the data for the removal of this redundancy. The number of chosen Principal Components is determined by their summed variations. In successive calculations we used seven firsts PCs, which ensured above 96% (96.40%) of cumulative variation and resulted in a loss of less than 4% of information (Fig.3.b).

![Fig.3: PCA plot containing four clusters indicating separate features of seafloor sound reflectivity a), summed variations of first seven PCAs b).](image-url)
In the next step, the Calinski-Harabasz index [11] was calculated to determine the number of clusters centers needed for the classification procedures. The Principal Components were the input to fuzzy c-means (FCM) data clustering algorithm [12]. The example of the result of this algorithm product is presented below (Fig.4).

![Fig.4: Example of result of segmentation using shape parameters of the angular dependency of backscattering intensity.](image)

### 3.1. PARAMETRICAL ANALYSIS OF THE SHAPE OF BATHYMETRIC TRANSECTS

The second classification method utilises the information included in the shape of bottom surface. From the high resolution bathymetric 3D map of tested polygon (the white rectangle area in Fig 1.b) 150 vertical and 150 horizontal parallel bathymetry cross-sections were extracted. An example of one bathymetric vertical cross-section taken in the middle part of investigated area is presented in Fig.5. Such cross-section was the object of high-pass filtration procedure necessary for elimination of the depth level dependency on the parameters values.

![Fig.5: Example of one bathymetric vertical cross-section taken in the middle part of investigated area (white rectangle in Fig.1.a).](image)

For each consecutive cross-section the shape parameters in sliding window were computed. There were 26 spectral, fractal and wavelet transformation parameters defined in section 3.1. The spatial resolution of such a parameterised bathymetric map were depended on sliding window width (256, 512 or 1024 samples) and the distance between consecutive cross-sections. The segmentation procedure was almost the same as method presented in section 3.1. The set of parameters were object of the PCA process. After the choice of first 6
Principal Components and the computation of the Calinski-Harabasz index [11], Principal Components were input to the FCM [12] classification algorithm. Fig. 6 presents a comparative set of a bathymetric map and segmented bottom imageries.

![Fig. 6: Comparative set of a bathymetric map (a) and segmented bottom images for 3 clusters (b) four clusters (c) and five clusters (d).](image)

4. CONCLUSIONS

The both MBES bottom imagery segmentation schemes presented in this study have many promising features which allow them to be applied for extracting morphological forms of seabed and habitats. The first method based on angular dependency of the backscattered intensity delivers information about the reflectivity of the measured area, when the second method is strongly associated with the morphology of the investigated polygon. Both techniques precisely indicated areas of relicts of periglacial forms as well as contemporary forms of marine origin. The results of sidescan sonar bottom imageries, echograms made using single beam echosounders, sedimentological and biological sampling, and video frames analyses, confirm precision and effectiveness of both supplementary segmentation systems. The benthic flora and fauna settled in bottom geomorphologic forms create separated habitats detectable by both systems. The correctness of the method was verified by the results of underwater video recordings, single beam echosounder registrations and biological samples taken in situ.
4. ACKNOWLEDGEMENTS

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USE OF A HIGH-RESOLUTION PROFILING SONAR FOR SEAGRASS DETECTION

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Abstract: Seagrasses are flowering plants that develop extensive underwater meadows and play a key role in the coastal ecosystem. In the last few years, several techniques have been developed to map and monitor seagrass beds in order to protect them. We present here the results of a survey using a profiling sonar - the Sediment Imager Sonar (SIS) - and a towed video sledge to study a Zostera marina bed off Calshot, southern UK. The survey aimed to test the instruments for seagrass detection and describe the area for the first time. Data processing from the SIS and video data will be presented along with correlation of the 2 instruments.

From the SIS data, the bed elevation was recorded as the strongest backscatter along a beam. The presence of seagrass was indicated as a high backscatter intensity above the seabed. For each sweep, four parameters were calculated from the SIS data: depth of the seafloor, Seagrass Index (used to assess seagrass presence), canopy height and vegetation patchiness. From the video, Zostera density was estimated together with macroalgae abundance and bottom type. The results of an algorithm developed to detect seagrass from the SIS data were tested against video data. Patchiness calculated from the SIS data was strongly correlated to seagrass density evaluated from the video, indicating that the parameter could be used for seagrass detection. Canopy height evaluated from the acoustic data was consistent with field observations. The SIS therefore proved to be a useful instrument for surveying seagrass beds although it needs to be tested over other seagrass types.

Keywords: seagrass detection, acoustics, Zostera marina, canopy height
1. INTRODUCTION

Seagrasses are flowering plants that have adapted to the submerged marine environment. They develop extensive underwater meadows forming complex, highly productive ecosystems. Seagrass beds play a key role in coastal ecosystems by attenuating currents and waves [1, 2], promoting sedimentation and reducing erosion [3] and stabilising bottom sediments through their roots [4]. They also provide shelter and refuge for adult animals, serve as nurseries for juvenile fish [5] and supply food for herbivores which graze on live seagrass leaves and consume epiphytic algae that grow on seagrass leaves [6].

Preservation, restoration or creation of seagrass beds are increasingly recognised as being essential for the sustainable management of the coastal environment [7, 8]. Therefore it has become a growing concern to accurately map and monitor seagrass beds in order to assess their state and protect them [9]. Several techniques have been used to detect seagrass beds and estimate seagrass density, among them, underwater video camera systems and sonars. Underwater video camera systems have proved to be a good technique for surveying seagrass beds as they provide direct observation of the seabed [10, 11]. They can give a record of submerged vegetation species composition and abundance as well as a description of the non-vegetated seabed. However, the data quality is limited by water clarity and boat speed and detailed interpretation of the video is time-consuming and subject to individual observer bias [8].

Acoustic systems such as side-scan sonar, echosounder or mutibeam sonar can be used to measure the amount of acoustic energy scattered by plants as a proxy for seagrass abundance. The use of those techniques is growing as they facilitate more quantitative and spatially-referenced studies of submerged vegetation abundance [12, 13]. A density contrast between an object and seawater usually produces a strong backscatter response in the water column. This can be created by several phenomena, such as bubbles, suspended sediment, fish, the seabed or submerged vegetation [14]. As seagrasses have air-filled tissues [15], the density contrast between the plants and seawater creates a strong acoustic echo which can be used to assess seagrass canopy presence [13].

Zostera marina L. (eelgrass) is one of the numerous seagrass species of the genus Zostera. It is widely distributed in both the northern Pacific and the northern Atlantic, and is the dominant seagrass species in the latter [8]. Z. marina is a flowering plant with dark green, flat leaves shooting from a rhizome that binds the sediment [16]. It forms extensive submarine meadows on muddy or sandy substrates developing in subtidal (0-10 m), sheltered water [4].

The survey presented here was carried out to test the use of a profiling sonar, the Sediment Imager Sonar (SIS), for seagrass detection. The SIS was used together with a video camera sledge to map the extent and structure of a Zostera marina bed that had not been previously mapped and the SIS capacity to detect seagrass was tested against the video data.

2. MATERIAL AND METHODS

2.1. SURVEY SET UP AND INSTRUMENTS DESCRIPTION

A survey was undertaken on 12th September 2007 onboard the R.V. Bill Conway around high tide, using the SIS and a video camera system, off Calshot beach (West Solent) on the south coast of England. Calshot beach is composed of gravel (flint) with sandy patches and patchy Zostera marina meadows. The Sediment Imager Sonar, produced by Marine Electronics Ltd. (Guernsey), is a high-frequency (1.1 MHz), single-beam sonar (beam width angle 1.8°) with a rotating head. High-resolution acoustic images of the seabed and the water
column are recorded for each sweep and converted to ASCII image intensity with a custom software (Sediment Imager Converter 1.0). Data available after conversion are beam angle (degrees), distance from transducer (m) and backscatter (0 - 255). The sonar beam is swept at right angles to the sonar body and during the survey, the instrument was facing the bow so that sweeps (46.8° centered downward, beams every 0.9 °) were perpendicular to the boat’s direction of travel. The pulse duration was fixed to 40 µs and the measurement range was limited to 6 m yielding an along-beam resolution of 1.5 cm. The video system used during the survey is composed of a downward facing video camera (Divecam-550C from Bowtech Products Ltd, Aberdeen, UK) and a light source mounted on a specially designed aluminium sledge. The sledge was towed behind the vessel with a Kevlar cored multi-conductor cable. With the camera held at approximately 50 cm above the bottom, the field of view averaged an area of approximately 50 x 70 cm. Boat location was recorded with the onboard differential GPS and overlaid on the video signal.

During the survey, the SIS was fixed 94 cm below the waterline to a downrigger mounted on the starboard side of the boat while the camera system was towed around 20 m behind the boat. A monitor and a computer in the cabin gave real-time displays of the video and SIS images. Four lines parallel to the shore were surveyed in 3.5 hrs at a speed of 1 to 2 knots to allow good quality video images. Lines were about 1.5 km-long and were spaced 50 to 100 m apart. Salinity and temperature profiles were measured on site prior to survey using a YSI 30 probe to input the correct sound velocity in the SIS software.

2.2. DATA PROCESSING

The video was replayed at half speed to allow adequate time for observations to be made. Every 2 to 3 seconds, the video was paused and relative Zostera density (ZosD from 0 for no seagrass to 4 for dense canopy), macroalgae abundance (AlgI from 0 for no algae to 2 for numerous algae), bottom type (BoTyp from 1 for sand to 3 for gravel), time and position were logged.

The SIS swept over the bed perpendicularly to the direction of travel. Hence, it was recording data over a line going 4 to 10 m in the direction of boat travel and 1 to 3 m towards one side of the instruments (negative beam angles to the port side, positive to the starboard side) before changing direction and recording another sweep. The resulting SIS track was therefore in zigzags. Most of the processing was done on 5 averaged beams in the middle of the sweep (beams at ±1.8° from the vertical) in order to produce one data point per sweep. First the depth of the bed was computed as the depth of the maximum backscatter (Fig. 1). A sharp contrast in acoustical impedance at the sediment/water interface defines the bed as the strongest acoustic scatterer of a given beam [13]. Hence the depth of the strongest backscatter usually denotes the depth of the seabed. This depth was corrected for the height of the tide during the survey and the depth of the transducer under the boat in order to provide bathymetric data along the survey lines. Secondly, the value of the maximum backscatter, i.e. the backscatter on the seabed, was recorded. Thirdly, a ‘Seagrass Index’ (SI) was calculated by averaging the intensity of the backscatter 10 to 15 cm above the seabed (Fig. 1). When present, submerged vegetation usually produces a strong backscatter immediately above the bottom to a height that depends on canopy height [17]. The Seagrass Index was used to discriminate seagrass canopies from bare seabed, i.e. low SI corresponding to bare seabed (Fig. 1a) and high SI indicating dense vegetation (Fig. 1b). Forth, the height of the canopy (CH) was computed as the height above the bed where the backscatter was greater than a threshold value (average backscatter in the water column + 60). Finally, ‘patchiness’ (P) was calculated for each sweep. As the SIS swept the seabed perpendicularly to the boat track, it
provided data on the lateral extent of the canopy. An attempt to measure the seagrass bed patchiness, defined here as the percentage of the seabed fully occupied by seagrass plants during a sweep, was computed as the percentage of beams in a sweep where the SI was superior to a threshold value (backscatter of 120). A P of 0 indicates that no seagrass was recorded on the whole sweep and a P of 1 indicates that a dense canopy was recorded over the whole sweep. As all the beams in a sweep were used to calculate P, full advantage was taken of the lateral extent of the seabed surveyed using the SIS.

Fig. 1: Backscatter intensity along the 5 averaged beams as a function of depth from transducer head, for a sweep over (a) bare seabed and (b) a seagrass canopy. The depth of the bed is coincident with the depth of maximum backscatter (black dot on the line bed). The Seagrass Index (SI) is the average backscatter 10 to 15 cm above the bed (black rectangle) and the height of the canopy (CH) is the first point where the backscatter is above a threshold value. In Fig. 1a, the SI was 93 yielding a SI of 0.1 i.e. no or little seagrass; no point satisfied the threshold condition for the canopy height which therefore was 0. In Fig. 1b, the SI was 177 yielding a SI of 3.9 i.e. dense seagrass canopy; the first point satisfying the threshold condition for canopy height is situated 17 cm above the bed, so CH = 17 cm.

Data points collected with the two instruments were not coincident; video observations were made every 2 to 4 m whereas SIS data points were approximately every 4 to 10 m depending on boat speed. Furthermore, due to the survey setup, the SIS was positioned around 22 m in front and 2 m on the starboard side of the camera. The two datasets had to be merged to enable comparison of results. First, co-ordinates were corrected from the layback of the video (around 25 m depending on the depth) and the SIS (2 m) to the GPS antenna. Thereafter, the values of the parameters calculated from the video were interpolated every metre (in order to achieve a regular grid and therefore the same number of video points around each SIS point) and averaged 5 m around each SIS data point. Averaging video parameters around SIS points reduces uncertainty due to variations in the height of the camera above the bed and the lateral difference between the location where the camera and SIS were recording the seabed, which could not be corrected.
3. RESULTS

A total of 1007 sweeps, each composed of 75 beams, was recorded during the 3 hours of the camera survey. In each sweep, the seabed was clearly recognisable by its strong backscatter (Fig. 2a). Where seagrass was present, a strong backscatter was observed above the bottom, which was weaker than the bottom return but stronger than ambient noise. Seagrass could be detected over the whole length of the sweep (Fig. 2b) or only parts of it where the seagrass bed was patchy (Fig. 2c).

Fig. 2: Examples of SIS sweeps over (a) bare seabed, (b) continuous seagrass canopy and (c) patchy seagrass. Distance along the x-axis refers to the horizontal distance along the sweep. The strongest backscatter denotes the seabed and a backscatter higher than ambient noise above the bed indicates the presence of seagrass. The seabed is seen to be undulating due to boat movements with waves.

The correspondence was good between Zostera Density (ZosD) calculated from the video data and patchiness (P) computed from the SIS data ($R^2 = 0.61$, $n = 1007$, $p < 0.001$). The correlation was not as strong between Seagrass Index and ZosD ($R^2 = 0.44$, $n = 1007$, $p < 0.001$). In order to compare the 2 datasets, SI and P were scaled to the video data values as followed:

$$SI_s = 4 \frac{(SI - 85)}{(180 - 85)}$$
$$P_s = 4 \frac{(P - 0.05)}{(0.85 - 0.05)}$$

This enables direct comparison of the 2 datasets as $SI_s$ and $P_s$ vary between 0 (no seagrass) and 4 (dense canopy) as the video parameter does. Along the survey lines, there was general agreement between the video and SIS evaluations of seagrass density; $P_s$, $SI_s$ and ZosD showed the same trends and intensity ranges (Fig. 3a and 3b). Only at some locations, e.g. around 4500 m along the line, $P_s$ predicted a canopy less dense than ZosD. $SI_s$ showed similar trends to ZosD and $P_s$ but had more along-line scatter. Discrepancy encountered between video and SIS estimations of seagrass density may be caused by an erroneous estimation of seagrass density from the video due poor visibility (high turbidity in the water column) or variations in the height of the sledge revealing more or less of the seagrass bed and therefore affecting observer classification. Phenomenon affecting the seagrass acoustic properties, such as variations in gas production or epiphyte loading, may also account for some of the differences and should be further tested.

Seagrass canopy height ($CH$) was found to be between 0.1 and 0.2 m (Fig. 3c). The video system could not produce quantitative data for canopy height and so assessment of accuracy was not possible for this parameter. However, canopy height, which is dependent on current speed and seagrass density [1], was observed during dives and found to be between 0.15 and
0.3 m. It thus appears that CH calculated from the SIS data was a good approximation of the measured height of the canopy.

The algae abundance parameter (AlgI, Fig. 3d) did not correlate with seagrass density evaluated from the SIS data, Ps or SI (R² = 0.03 and p < 0.001 for both parameters). Furthermore, when AlgI was greater than 1.5 and ZosD was 0, i.e. significant quantities of algae seen on the video but no seagrass, both the scaled patchiness (Ps) and Seagrass Index (SI) had values below 0.5, i.e. no seagrass or only few shoots were present. This suggested that SI and Ps were not influenced by algae presence.

Fig. 3: Comparison of the different parameters computed from video and SIS data. The shaded areas show zones with little or no seagrass. (a) seagrass density calculated from SIS data, Ps (grey) and from video data, ZosD (black); (b) seagrass density SI, calculated using SIS data; (c) canopy height CH estimated from SIS data; (d) algae abundance (AlgI) described from video data; (e) bottom type (BoTyp) rated from video data; (f) backscatter intensity on the seabed (bed back) calculated from SIS data.
Seabed backscatter (Fig. 3f) did not show the same along-line variations as bottom type assessed from the video, BoTyp (Fig. 3e). Furthermore, backscatter on the seabed exhibited strong point to point variations. Acoustic attenuation is affected, amongst others, by sound travel distance, turbidity in the water column and presence of submerged vegetation. Results indicated that backscatter on the seabed was weakly correlated with depth during the survey \((R^2 = 0.3)\) and that it was not correlated with average backscatter in the water column \((R^2 = 0.06)\) or Zostera Density \((R^2 = 0.01)\). A relationship was sought between bottom type and bed backscatter in areas with no or little seagrass \((ZosD < 1)\) by averaging the bed backscatter for each bottom type. The average backscatter on the seabed was found to be the lowest on sandy bottom (179), intermediate on mixed sand and gravel (190) and highest on areas of gravel (198). However, the standard deviations of each distribution (20, 21 and 23 respectively), were higher than the difference in average values therefore reflecting the high variability of the seabed backscatter. It was thus estimated that this parameter could not be used to determine bottom type, even in areas with little or no seagrass.

4. DISCUSSION

The results show that it is possible to use the SIS to assess *Zostera marina* abundance and canopy height through \(P, S_i, \) and CH and that these parameters were not affected by the presence of algae. The SIS has therefore proved a reliable tool to detect seagrass, giving estimates of canopy height and density as well as measures of bathymetry.

Agreement between the 2 instruments was very good; seagrass density evaluated from the video was better correlated with the parameter initially calculated to measure patchiness \(P\) than to the one specifically used for seagrass detection \(S_i\) \((R^2 of 0.61 and 0.44 respectively)\). The Seagrass Index \((SI)\) was calculated from 5 beams over the 52 contained in the sweep and hence covers 0.3 to 0.8 m of the seabed. The patchiness \(P\), on the other hand, was calculated from the data of an entire sweep i.e. a line of 4 to 10 m in boat direction and 0.5 to 1.5 m on each side of the instrument depending on water depth. Therefore, SI specified the presence and abundance of seagrass at a precise location whereas \(P\) showed an average of seagrass cover between two points. As the video data were averaged 5 metres around the SIS data points, they were also assessing seagrass cover between two points. This certainly explains the fact that \(P\) showed a better correlation with video data than SI did.

The processing methodology used to calculate SI, \(P\) and CH was found empirically by trial and error. The threshold values and heights of averaging worked for the survey presented here, however, they will probably need to be changed for other surveys, depending on environmental conditions and seagrass species surveyed. For example, the height where the backscatter is averaged to calculate SI might have to be different in the case of a smaller seagrass species. Moreover, threshold values might need to be adjusted to water turbidity.

5. CONCLUSION

The use of both a video camera sledge and a high-resolution profiling sonar during the survey allowed the evaluation of the SIS for seagrass detection and a preliminary mapping of the seagrass bed in Calshot. The advantages of the camera system were direct observations of the seabed, the possibility of flora identification and bottom type classification. However, data processing was time-consuming and subject to observer error. The SIS has proved to be a useful tool for seagrass surveying. An algorithm to analyse the data was developed and tested. The Seagrass Index \((SI)\) was calculated as the average backscatter 10 to 15 cm above the bottom of 5 beams averaged in the middle of a sweep and the patchiness \((P)\) as the
percentage of beams in a sweep where the average backscatter 10 to 15 cm above the bottom was higher than a threshold value. It was possible to estimate seagrass abundance using SI and P. Furthermore, the SIS was used to estimate canopy height and accurately measure depth. On the other hand, the backscatter on the bottom showed too much scatter to be used for bottom type determination. The system needs to be tested with different seagrass species in order to assess the influence of seagrass morphology on the SIS backscatter.

REFERENCES

REPRODUCIBILITY OF SINGLE-BEAM ACOUSTIC SEABED CLASSIFICATION FOR HABITAT MAPPING

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Abstract: Single-beam acoustic seabed classification continues to be a popular method for mapping seabeds and their sediments, which are important components of benthic habitat. Modern methods can generate maps of acoustic classes that are useful and reasonably accurate. Research toward improved methods continues. A continuing impediment to this research is ranking the accuracy of maps produced by new methods. Non-acoustic data, or ground truth, is usually sparse compared to the detail of the acoustic survey, which can mean that ranking maps for accuracy can be inconclusive. Here we present a new tool for ranking classification maps, namely the reproducibility of acoustic classes from repeated surveys of the same area on different days. Methods that have high reproducibility achieve that by capturing echo characteristics that are strongly influenced by seabed type while suppressing details that are driven by sea state or the water column. Six surveys, done with a 50 kHz sounder over a pair of transects near Miami, FL, USA, between 1 May and 13 August 2007, were used to evaluate two questions. First, how reproducible were classifications of this dataset using QTC IMPACT\textsuperscript{TM} (Quester Tangent Corporation)? Second, can classification be improved with adjustments to the standard IMPACT processing? Reproducibility was quantified with the overall accuracy and the Kappa statistics, which are both derived from the confusion matrix whose rows and columns are numbers of sites with particular class assignments under distinct circumstances.

Keywords: acoustic classification, seabed classification, echo analysis, confusion matrix, reproducibility
1. INTRODUCTION

A major challenge in developing methods for acoustic seabed classification is comparison of results from competing methods. Since the primary goal is accuracy, comparing acoustic classes with ground truth is the ultimate test. This may not be helpful in ranking the accuracy of class maps made by competing methods, though, because results from acoustic processes are often one hundred or more times more dense than the locations of ground truth. Acoustic class maps can be far more detailed than ground truth (indeed this is one motivation for acoustic seabed classification), so another method is needed to rank classification algorithms. One method is to compare borders between acoustic classes with detailed bathymetry. While this can be accurate for borders of rocky regions, it is not useful in discriminating clastic sediments. Other methods that can be of some value are examining consistency in overlap regions and homogeneity in regions expected to be homogeneous, but they are somewhat subjective.

This paper presents a new method for assessing and ranking maps of acoustic seabed classes, based on reproducibility among a series of surveys of the same area. Methods that have high reproducibility achieve that by capturing echo characteristics that are strongly influenced by seabed type while minimizing the effects of sea state and the water column.

2. SURVEY AND ACOUSTIC CLASSIFICATION

The survey site [1], Fowey Rocks, has geomorphology typical of the major Florida Keys reefs. Two parallel transects were selected, each about 2 km long with depths from 5 to 60 m. A small boat fitted with a Suzuki 2025 echo sounder was used to survey both transects three times on each of six dates: May 1, 2, 9, and 28, and August 3 and 13, 2007. Echo time series were acquired by a QTC5™ data acquisition system, which sampled the amplified echoes at 5 MHz followed by decimation and digital filtering. The sounder operated at 50 kHz and 500 W transmit power, with a rectangular beamwidth 42º x 16º and a pulse length of 0.3 ms.

![Fig.1: Vertical sections across the north (left) and south (right) transects showing the main bathymetric features, namely a series of linear reefs spaced by belts of relatively flat sediment in shallow water and sloping sediment in deeper water. Colours are acoustic classes from the 1May data set.](image)
Unsupervised acoustic classification is the segmentation of an area into regions that are acoustically similar [2]. While the basic process is well established, research into improved methods and algorithms continues. In this work, the initial sets of classes were produced in QTC IMPACT™. The steps in IMPACT's process are bottom picking, depth compensation, stacking, feature generation, dimension reduction with principal components analysis, and objective clustering with ACE. Further sets came from a developmental version of IMPACT that both replicates the commercial version and has the flexibility to use a wide variety of other techniques, particularly different feature algorithms.

Averaging a number of consecutive echoes, which is called stacking, is a method of increasing the signal-to-noise ratio. The noise is ping-to-ping variability and the signal is shape and spectral character of echoes. Tables 1-4 contain results based on stacking five echoes, a common practice. Table 5 compares classes with stacks of 1, 5, and 15 echoes.

ACE returns assignments of records into any number of classes, and recommends an optimal number of classes [2]. These data sets, from the six survey days and processed in various ways, almost always had an optimal class number from 4 to 6. Results that are to be compared for reproducibility need to have the same number of classes, so in each case the five-class solution was selected. This means that every classified point was assigned to a class numbered from 1 to 5. Figure 1 shows bathymetric cross-sections of the survey lines overlaid with classes from QTC IMPACT using data from 1 May. Figure 2 shows all 18 replicates (three on each of the six days) of the northern of the two transects coloured by the classes derived for each day.

Features are values calculated from echo time series that capture information that discriminates among echoes from different seabeds. Normalized cumulative integrals discriminate well by capturing details of echo shape. These features are values of the cumulative integral at fixed fractions of the number of samples being processed. Another set of shape features, evaluated recently [3], is the echo centre of gravity, energy spread, and skewness. These three are abbreviated to cg_es_skew in Table 4.

Fig. 2: Acoustic classes on the northern survey line at Fowey Rocks. The two lines were surveyed three times on each of the six days. Left plot shows the 18 northern transects plotted together; right with 50 m artificial spacing between them. Class colours (black, grey, blue, red, and yellow) correspond to positions of class centres in feature space.
The commercial version of QTC IMPACT uses only features derived from echo shape and spectral character, not from amplitude. Amplitude features had been avoided because, if not corrected carefully for depth, they can produce depth-dependent artifacts. This is particularly true at high sonar frequencies where sound absorption can be important but water temperature and salinity are often unknown. This risk was minimized in the developmental version of IMPACT by compiling tables of amplitude against depth in a first pass through the echo data, and calculating from that a fitted trim-TVG correction curve to use in preparing each echo for feature generation. This is similar to Quester Tangent’s method for range and angle compensation of multibeam images [4].

3. MEASURES OF REPRODUCIBILITY

Classification processes are reproducible if they consistently assign nearby locations to the same geological class. Acoustic classes from QTC IMPACT and most other methods, though, are maps of acoustic diversity, showing areas that are acoustically homogeneous and distinct from other areas. Non-acoustic data are needed to attach geological names. In this work we wish to measure reproducibility based on acoustic classes alone.

Each day’s data were classified independently of the others, with its five classes randomly numbered. Tables are therefore needed showing the correspondence of class numbers on any particular day to the numbers on any other day. Matching classes could be based on proximity in feature space or geographically. Matching in feature space (Q space) is the more pure test: match in feature space then measure reproducibility of geographical neighbours. Figure 3 illustrates matching in feature space. Class centres are shown in a plot of the first two dimensions (Q space is three-dimensional) with colours and shapes denoting dates and class numbers respectively. Elliptical borders separate the obvious groups of six. Each of the

![Fig.3: Groups of class centres in feature space (Q space). Each colour represents one of the six days and each symbol one of the five classes. Grouping is needed because clustering assigns numbers to classes randomly. In these data, which are from the first row in Table 1, the single acoustic class at high Q1 and low Q2 was assigned numbers 2, 1, 3, 4, 2, and 5 in the independent classification processes of each of the six days.](image-url)
five groups has one class from each of the six days, and these five groups are the correspondences we seek. To form these groups in less obvious cases, one could calculate the net intra-group distances for all possible arrangements and choose its minimum. However with six days and five classes there are almost 25 billion possible groups. To identify groups in a reasonable time, two groups of six were identified by eye, leaving distances to be calculated for only 7776 candidate groups.

A complication in making these matches is that principal components analysis within QTC IMPACT gives random signs to the component axes. Because all the situations studied in this work were at least reasonably reproducible, it was usually obvious which axes required their sign to be reversed to coincide with the others. The indicated sign reversals were done before matching.

Geographical interpolation was the next step in calculating reproducibility. While the repeated surveys followed the same transects accurately, the locations in these surveys to which classes were assigned did not coincide precisely. Taking one survey as the base set, all the classified points in another set that lie within a search radius, 10 m, were sought and a modal class identified using an occurrence histogram. This is the method of categorical interpolation in QTC CLAMS\textsuperscript{TM} [2]. Let us take, as an example, a point in Class 1 in the first day’s survey. Nearby points of the second day’s survey might be predominantly in class 3, giving one correspondence between Day 1 Class 1 and Day 2 Class 3. Repeat for 500 points of the base set, and tabulate the correspondences in a 5x5 matrix with the base day’s classes as columns and the other day’s classes as rows. Table 1 is an example. Finally, rearrange the rows according to the feature-space matches found as described above. If reproducibility were perfect, all the off-diagonal elements of this matrix would be zero. A perfect situation is very rare, which is why these matrices are called confusion matrices.

The statistic called overall accuracy [5] is the trace of a confusion matrix divided by its sum, thus the fraction of assignments that agree. Some correct matches occur randomly, and the statistic Kappa, $\kappa$, corrects for this [5].

$$\kappa = \frac{OA - P_r}{1 - P_r}$$

Here OA is the overall accuracy and $P_r$ is the random probability of a correct match in the confusion matrix. One way to calculate $P_r$ is to assume all classes are equally probable, in which case it equals the reciprocal of the number of classes. Other methods include class populations.

In the literature, OA stands for Overall Accuracy and is a measure of a set of class assignments against a set of validation data that is known to be correct, or, at least, trustworthy. As used here, however, it is a measure of reproducibility among class maps, with no assumptions about which map has the highest fidelity. In this context, OA should perhaps stand for Overall Agreement and all references to accuracy should read agreement, even though accuracy is often used in this looser sense by other authors.

4. RESULTS AND DISCUSSION

The thesis of this work is that reproducibility among repeated surveys of the same area can indicate the relative merits of current and developmental methods for acoustic seabed classification. Methods that have high reproducibility achieve that by capturing echo
characteristics that are strongly influenced by seabed type while suppressing details that are driven by sea state or the water column.

Seasonal variations often occur in coastal waters and seabeds, due to physical and biological processes in those habitats. At Fowey Rocks there were seasonal changes between the four surveys in May and the two in August, as shown in Table 3. The overall accuracy of reproducibility was less between May-August pairs of days than among the four days in May. While these changes were real, their nature is not known. This seasonal variation does not interfere with assessing the merits of classification methods because each of the columns in Table 3 has the same pattern and would serve for this purpose.

For the transects at Fowey Rocks, surveyed three times on each of six days, reproducibility of classes among the days can be improved by using different families of

<table>
<thead>
<tr>
<th>Test set, May 2</th>
<th>Reference set, May 1</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Class 1</td>
</tr>
<tr>
<td>Class 3</td>
<td>85</td>
</tr>
<tr>
<td>Class 4</td>
<td>0</td>
</tr>
<tr>
<td>Class 5</td>
<td>37</td>
</tr>
<tr>
<td>Class 1</td>
<td>13</td>
</tr>
<tr>
<td>Class 2</td>
<td>2</td>
</tr>
</tbody>
</table>

Table 1: A confusion matrix for 409 locations in the 2 May data set that were close enough to 500 random locations in the 1 May reference set to have a reference class assigned. For this matrix, the overall accuracy and Kappa are 0.65 and 0.51. The values in Table 2 were averages from three of these matrices for each pair of dates.

<table>
<thead>
<tr>
<th></th>
<th>May 1</th>
<th>May 2</th>
<th>May 9</th>
<th>May 28</th>
<th>Aug 3</th>
<th>Aug 13</th>
</tr>
</thead>
<tbody>
<tr>
<td>May 1</td>
<td>100%</td>
<td>54%</td>
<td>51%</td>
<td>58%</td>
<td>40%</td>
<td>40%</td>
</tr>
<tr>
<td>May 2</td>
<td>62%</td>
<td>100%</td>
<td>53%</td>
<td>58%</td>
<td>41%</td>
<td>39%</td>
</tr>
<tr>
<td>May 9</td>
<td>54%</td>
<td>50%</td>
<td>100%</td>
<td>48%</td>
<td>49%</td>
<td>56%</td>
</tr>
<tr>
<td>May 28</td>
<td>55%</td>
<td>58%</td>
<td>54%</td>
<td>100%</td>
<td>40%</td>
<td>40%</td>
</tr>
<tr>
<td>Aug 3</td>
<td>48%</td>
<td>46%</td>
<td>52%</td>
<td>38%</td>
<td>100%</td>
<td>56%</td>
</tr>
<tr>
<td>Aug 13</td>
<td>42%</td>
<td>46%</td>
<td>58%</td>
<td>44%</td>
<td>53%</td>
<td>100%</td>
</tr>
</tbody>
</table>

Table 2: The Kappa measure of reproducibility for unsupervised classification with QTC IMPACT, with each of the six days treated as both reference and test data set.

<table>
<thead>
<tr>
<th>Feature Families</th>
<th>Within May</th>
<th>May to August</th>
<th>All pairs of days</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>67%</td>
<td>57%</td>
<td>61%</td>
</tr>
<tr>
<td>2</td>
<td>59%</td>
<td>61%</td>
<td>61%</td>
</tr>
<tr>
<td>3</td>
<td>55%</td>
<td>46%</td>
<td>51%</td>
</tr>
<tr>
<td>4</td>
<td>55%</td>
<td>67%</td>
<td>63%</td>
</tr>
<tr>
<td>5</td>
<td>80%</td>
<td>70%</td>
<td>74%</td>
</tr>
<tr>
<td>6</td>
<td>80%</td>
<td>70%</td>
<td>73%</td>
</tr>
</tbody>
</table>

Table 3: Averaged Overall Accuracy with various feature sets considered seasonally. Six day-to-day comparisons were made among the four days in May, and eight between May and August dates. Overall accuracies for the one pair of dates within August were omitted from this table because they are erratic.
features, particularly features that explicitly use echo amplitudes. Table 4 has values of overall accuracy, both raw and corrected for random assignments (Kappa), for the set 1, the current feature set of QTC IMPACT, and several developmental feature sets, 2-6. These new features will soon be commercially available in a real-time version of QTC IMPACT and in a new release of the post-processing software suite.

The first three families of features in set 1 of Table 4 (cumulative integral, quantile, and histogram) capture echo shape, while the FFT and wavelet features were designed to capture spectral content. Figure 2 shows that they produce useful and realistic maps of acoustic class even though the statistical values are less than impressive. Switching to an alternate method of depth compensation, in which the echo time series is not resampled to a fixed number of samples but rather the feature algorithms are modified to compensate as needed, does not affect reproducibility (wavelet features were dropped because they do not fit the alternate method). Replacing some of these shape features with others does not improve reproducibility, nor does calculating some fractal-based features in place of Fourier transforms. However, these results do show that a smaller number of features can be as effective as the original 166 features. Adding features based on echo amplitudes are, for this data set, a significant improvement. Kappa values as high as 80% were obtained with feature sets that include amplitude and with averaging over the best-matching five of the six days. Overall, this indicates that smaller feature sets with well-chosen shape and spectral features and with amplitude features significantly improves reproducibility and thus the quality of class maps.

Another process that improves reproducibility is to stack a larger number of echoes. Usually five echoes are stacked, that is, their time series are averaged to reduce ping-to-ping variability, which is a form of noise reduction. Table 5 shows that reproducibility is much lower without stacking but is usefully improved by stacking three times as many echoes. In

<table>
<thead>
<tr>
<th>Feature Families</th>
<th>n</th>
<th>OA</th>
<th>K</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Cumulative integral, quantile, histogram, wavelet, FFT</td>
<td>166</td>
<td>61%</td>
<td>49%</td>
</tr>
<tr>
<td>2. Cumulative integral, quantile, histogram, FFT</td>
<td>104</td>
<td>61%</td>
<td>48%</td>
</tr>
<tr>
<td>3. Cumulative integral and histogram, quantile, histogram, FFT, cg_es_skew, fractal</td>
<td>117</td>
<td>51%</td>
<td>38%</td>
</tr>
<tr>
<td>4. Cumulative integral, cg_es_skew, cumulative histogram, fractal</td>
<td>30</td>
<td>63%</td>
<td>53%</td>
</tr>
<tr>
<td>5. Cumulative integral, cg_es_skew, cumulative histogram, fractal, amplitude</td>
<td>35</td>
<td>74%</td>
<td>63%</td>
</tr>
<tr>
<td>6. Cumulative integral, cg_es_skew, fractal, amplitude</td>
<td>25</td>
<td>73%</td>
<td>63%</td>
</tr>
</tbody>
</table>

Table 4: Averaged Overall Accuracy (OA) and Kappa (K) with various feature sets. The number of features is n. Row 1 is the QTC IMPACT process with its usual depth compensation. A simpler compensation method was used for the other rows. Averages are from tables such as Table 2, omitting the 100% values on the diagonal. Values of Kappa over 0.80 were observed with amplitude features included and excluding the one day that matched least well.

<table>
<thead>
<tr>
<th>Number of consecutive echoes stacked</th>
<th>OA</th>
<th>K</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 (no averaging)</td>
<td>0.36</td>
<td>0.19</td>
</tr>
<tr>
<td>5</td>
<td>0.61</td>
<td>0.49</td>
</tr>
<tr>
<td>15</td>
<td>0.69</td>
<td>0.60</td>
</tr>
</tbody>
</table>

Table 5: Effect of stacking various numbers of consecutive echoes. Overall Accuracy (OA) and Kappa (K) were averaged as for Table 4. Classification was done with IMPACT’s present features and usual depth compensation.
planning a survey, choosing the number to stack is a compromise between classification accuracy and spatial resolution.

A third process that improves reproducibility is to ignore echoes that were recorded with the boat pitched or rolled beyond some angle, or were from a steeply sloped seabed. Table 6 shows how reproducibility is improved by angle filtering, in an evaluation done without stacking. One filter was simply vessel pitch of more than 2°. Filtering by angle of incidence is more complicated because seabed slope and vessel attitude both have to be considered. The fraction of echoes that had angles of incidence above 5°, and were then set aside by the filter, ranged from 33% for the data from 3 Aug to 70% for 1 May. Both filters produce substantial improvements in reproducibility, as would be expected with such small angular tolerances and substantial fractions of the echoes being ignored. Further work is needed to define maximum allowable angles, which are expected to vary with beamwidth.

For each of these three processes, reproducibility has been shown to improve as the classification system is modified to emphasize seabed effects on the echoes. Developmental feature sets, especially those with features that are based on echo amplitudes, improve reproducibility, as does stacking a larger number of echoes, and as does using only echoes with angles of incidence near zero. This supports the thesis that improved reproducibility among repeated surveys of the same area is a reliable indication that the quality and accuracy of acoustic seabed classification has also been improved.

REFERENCES

A COMPARISON OF SINGLE BEAM AND MULTIBEAM SONAR SYSTEMS IN SEAFLOOR HABITAT MAPPING

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Geoscience Australia GPO Box 378, Canberra, Australia

Abstract: Multibeam and single beam sonar (MBS and SBS) systems were used to map the seabed habitats within the Mandu Mandu region of the Ningaloo Marine Park, Western Australia. Backscatter and bathymetry data were collected with the Reson SeaBat 8101 MBS operating at 240 kHz and the Simrad EQ60 SBS operating at 38 and 200 kHz. Backscatter data were classified into 3 classes using supervised classification. The accuracy of the results was assessed with data from video transects taken over the area. This study found that the habitat map produced from MBS backscatter was 90% accurate and those from SBS were approximately 80% and 70% accurate at 200 kHz and 38 kHz respectively. Misclassification in the map produced by MBS was highest across heterogeneous or boundary areas. Misclassification in the habitat maps produced from the SBS backscatter was primarily due to errors of interpolation. Although closer line spacing would have likely improved the accuracy of the SBS results, the distributions of backscatter data for the different classes shows MBS data to have better discrimination than data from SBS.

Keywords: Multibeam sonar, Single beam sonar, backscatter, habitat mapping
1. INTRODUCTION

Multibeam sonar (MBS) and single beam sonar (SBS) systems have been shown to be useful tools in benthic habitat mapping [1]. In general, SBS is considered to be a less expensive system than MBS to map a survey area, but provides much lower spatial resolution. This study compares seafloor classification results obtained from backscatter data collected with MBS and SBS from the Mandu Mandu region of the Ningaloo Marine Park in Western Australia, which is aimed to assess the accuracy of the different systems. The RoxAnn technique was used for processing single beam backscatter data [2]. Data collected from the MBS were processed using algorithms developed by the Centre for Marine Science and Technology (CMST) [3, 4]. Backscatter data were classified into 3 classes using supervised classification. The accuracy of the results was assessed with data from video transects under taken in the area.

2. METHODS

2.1. Data collection

From late April to mid May 2006, the Australian Institute of Marine Science’s vessel Cape Ferguson collected SBS data using a Simrad EQ60 sonar operating at 38 and 200 kHz. Multibeam data were collected by Fugro Survey PTY LTD from the same vessel using a Reson Seabat 8101 sonar system operating at 240 kHz. The sonar data sets were collected in conjunction with underwater video transects, which were used for the classification accuracy assessment.

2.2. Single beam sonar processing

The RoxAnn method of processing SBS data uses echo-integration approach to derive values for a tail part of the first echo return (E1) and the whole of the second echo return (E2). While E2 is primarily a function of the gross reflectivity of the sediment and therefore depends primarily on its acoustic impedance (or hardness), E1 is influenced primarily by backscatter from the small to meso-scale roughness of the seafloor. Therefore, E1 and E2 can be related in principle to the acoustic roughness and hardness of the seafloor respectively, although each of these parameters depends in general on both roughness and hardness.

Echoview software developed by Myriax has been used for quality control and processing of the SBS data. Following procedures described in [5], the two RoxAnn parameters E1 and E2 were derived using the Echoview software. Interpolation using Kriging [6] was used to produce gridded maps of E1 and E2 over the area from the along track single beam data. The grid cell size used was 50m.
2.3. Multibeam sonar processing

The MBS data were processed using the CMST MBS toolbox and the angle cube algorithm developed by the CMST [3, 4]. The CMST MBS toolbox uses the beam time series data (sometimes referred to as snippets) to calculate the backscatter strength based on the energy of backscatter pulses for each beam. This is done by integrating the squared amplitude of the snippet time series. The received backscatter energy is normalised by the energy of the transmitted pulse and corrected for receive gain, which makes the backscatter estimates independent of the system settings. In the normal operation mode, the MBS system applies time variable gain (TVG) to the received signals to compensate for spreading and absorption losses along beams. The CMST MBS toolbox removes the system TVG correction, as it is not always adequate to the actual conditions of acoustic propagation, and corrects the backscatter energy estimates for the actual spreading and absorption losses along each beam. Finally, the backscatter energy is corrected for the footprint size of each beam to obtain estimates of the surface backscattering coefficient (referred to here as backscatter strength as used in the logarithmic scale).

Backscatter strength produced by the CMST MBS toolbox can be corrected for angular dependence with an algorithm that corrects data for a section-average angular response [3]. A more advanced method was developed in [4] and referred to as the 'angle cube' algorithm. The angle cube method represents backscatter data from the survey area as a function of 3 dimensions: spatial coordinates X and Y, and the incidence angle, which produces a 3-dimensional sparse array of data. Then data in each angle layer are interpolated into each node of the X-Y spatial grid, producing a 3-dimensional matrix, or an angle cube. Of the commonly used interpolation techniques, Kriging was found to give satisfactory results, as the interpolated predictions did not reveal any unrealistic values. Finally, the angle average backscatter strength is calculated, for this dataset the backscatter was averaged over 5-60°, backscatter collected at incidence angles greater than 60° were noise. As the MBS used was not calibrated, the backscatter values are relative.

2.4. Classification

Three classes were observed in the video transects, namely 'rhodolith', 'sand' and 'mixed' in Mandu Mandu. Class descriptions are presented in Table 1. A cluster analysis was adopted here to classify backscatter characteristics which were the RoxAnn E1 and E2 parameters for the SBS and backscatter strength for the MB S. A supervised clustering technique with Bayesian distance was used. A training set was set up comprising distinct seabed habitats identified from video footages and results of video analysis conducted by AIMS. The mean of E1 and E2 for the single beam and the mean of backscatter strength for the multibeam, and their covariance matrices were estimated from the training set. The seed centroids were derived from the training set. Using these seeds, the supervised clustering technique was eventually performed on the rest of the data.
### Table 1: Habitat class descriptions

<table>
<thead>
<tr>
<th>Class</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>rhodolith</td>
<td>All hard seabeds, such as rhodolith, rubble, coralline, hard rock/reef</td>
</tr>
<tr>
<td>sand</td>
<td>Relatively flat sand</td>
</tr>
<tr>
<td>mixed</td>
<td>Sand (dominant) mixed with sparse “rhodolith”. Sand waves/dunes or sand with big ripples.</td>
</tr>
</tbody>
</table>

### 3. RESULTS

Fig. 1 shows the bathymetry and backscatter data obtained during the MBS survey of Mandu Mandu in the Ningaloo Marine Park in Western Australia. The track lines from the SBS survey used for this study are also shown in Fig. 1, as well as the video transects used for the classification accuracy assessment. The seafloor backscatter strength processed by the CMST MBS toolbox and the angle cube algorithm demonstrates the extent of the different seafloor substrates without artefacts of the system settings and angular dependence of backscatter.

The MBS backscatter data interpolated into the locations of the SBS measurements were compared with the RoxAnn parameters E1 and E2 at 38 and 200 kHz (Fig. 2). There was a noticeable correlation between the MBS backscatter values and E1, particularly for the 200-kHz SBS data. This correlation is reasonable as both MBS backscatter and E1 values are derived from the first bottom returns and are sensitive to changes in the seafloor roughness. The correlation is greater at higher frequency of SBS because of a closer operating frequency between the MBS and SBS (240 and 200 kHz). On the other hand, there was no significant correlation between the E2 parameter and the MBS backscatter. This is also reasonable as E2 is more sensitive to changes in the bottom reflectivity at vertical incidence rather than the seafloor roughness.
Fig. 1: (a) Bathymetry [m] and (b) backscatter [dB] from the multibeam sonar survey of the Mandu Mandu region of the Ningaloo Marine Park. Black lines are the track lines of the single beam survey and magenta lines are video transects.

Fig. 2: A comparison of the backscatter from the multibeam sonar data (240 kHz) and E1 and E2 calculated from single beam sonar data at 38 and 200 kHz.

The distributions of E1 and E2 derived from the 38 and 200 kHz SBS data and 240 kHz MBS backscatter from areas of sand and rhodolith are compared in Fig. 3. The plots show the separation in the sand and rhodolith classes to be largest in the MBS backscatter data,
followed by the E1 values and then the E2 values. This larger separation of the two different classes in the MBS backscatter data is due to the inclusion of backscatter from oblique incidence angles in contrast to the SBS data.

![Graphs showing probability density distributions for E1 and E2 backscatter at different frequencies for sand and rhodolith.](image)

**Fig. 3:** Probability density distributions of all data sand, rhodolith of E1 for (a) 200 kHz and (b) 38 kHz; and E2 of (c) 200 kHz and (d) 38 kHz and (e) 240 kHz multibeam backscatter.

Classification maps produced from the cluster analysis are shown in Fig. 4 and results of the classification accuracy assessment are summarised in Table 2. It was found that the MBS backscatter data were the most accurate, followed by the results of the SBS 200 kHz and then the SBS 38 kHz. Based on the video data collected results from MBS were highly accurate in recognising sand and rhodolith respectively. Misclassification in the map produced by MBS was highest across heterogeneous and boundary areas, such as the mixed areas. Misclassification in the habitat map produced from the SBS backscatter was primarily due to
errors of interpolation. However, the larger separation between classes seen in MBS backscatter indicates that MBS backscatter is better at discriminating the different classes found in this survey area. The ground-truth data was not comprehensive enough to consider the classification accuracy assessment to be absolute, but does provide an indicative accuracy of the different sonar systems, especially relative to each other.

![Classification maps of the Mandu Mandu region](image)

**Fig. 4:** Classification maps of the Mandu Mandu region produced from cluster analysis of (a) MBS backscatter, (b) SBS 200 kHz and (c) SBS 38 kHz. Classes are: rhodolith (brown), sand (blue) and mixed (green).

**Table 2:** Summary of classification accuracy derived from confusion matrices.

<table>
<thead>
<tr>
<th>System</th>
<th>Classification accuracy (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Overall</td>
</tr>
<tr>
<td>SBS 38kHz</td>
<td>69</td>
</tr>
<tr>
<td>SBS 200kHz</td>
<td>84</td>
</tr>
<tr>
<td>MBS 240kHz</td>
<td>91</td>
</tr>
</tbody>
</table>
4. CONCLUSIONS

It has been demonstrated that both SBS and MBS systems are reasonably accurate in seafloor habitat mapping. The MBS system, however, offers better resolution and coverage and in turn better classification accuracy than the single beam system can provide. Although closer track line spacing would have likely improved the accuracy of the SBS results, the distributions of backscatter characteristics for the different classes shows MBS to have considerable better discrimination than the data from SBS. This is most likely to be due to the inclusion of backscatter characteristics from oblique incidence angles in the angle cube algorithm, which are more sensitive to changes in the seabed roughness than those at vertical incidence. While MBSs are inherently more expensive system than SBSs, the results show that the track line spacing for a SBS survey needs to be considerably smaller than that of a MBS survey to have comparable results. As the costs associated with surveys are related predominantly to vessel time, the cost-benefit of using a MBS system with respect to ship time needed is considered to be an important factor when choosing a sonar system for seabed habitat mapping.

5. ACKNOWLEDGEMENTS

The authors would like to thank the Australian Institute of Marine Science and the Western Australian Marine Science Institution for effort associated with acoustic data acquisition, funding and providing video transect data.

REFERENCES

THE POTENTIAL OF INVERTING GEO-TECHNICAL AND GEO-AcouSTIC SEDIMENT PARAMETERS FROM SINGLE-BEAM ECHO SOUNDER RETURNS

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Abstract: Seafloor characterization is important in many fields including hydrography, marine geology, coastal engineering and habitat mapping. The advantage of non-invasive acoustic methods for sediment characterization over conventional bottom grabbing is the nearly continuous versus sparse sensing and the enormous reduction in survey time and costs. Among the various acoustic systems for seafloor characterization, the single-beam echo sounder is of particular interest due to its simplicity and versatility. Seafloor characterization algorithms can be roughly divided into two categories: model-based and empirical, where the latter simply relies on the observation that certain echo features, such as amplitude, duration and skewness of the echo, are correlated with sediment type. Here we apply the model-based approach where we compare the measured echo signal with theoretically modeled echo envelopes in the time domain. For modeling the received echo sounder signals use is made of a physical backscatter model that fully accounts for water-sediment interface roughness and sediment volume scattering. We use differential evolution, a fast variant of a genetic algorithm, as the global optimization method to invert the model input parameters mean grain size, spectral strength of the interface roughness and volume scattering cross section. In the model grain size determines geo-acoustic parameters like sediment sound speed, density and attenuation. The analysis is applied to simulated data.

Keywords: Single-beam echosounder, seafloor classification, optimization
1. INTRODUCTION

Up-to-date information regarding sea- or river floor composition is of high importance for a large number of applications. These include e.g. cable and pipeline route planning, geology and off-shore construction projects. Currently the most common method for obtaining the required information is to take sediment samples. These samples are then analysed in a laboratory, a time-consuming and costly process. An appealing approach therefore consists of using acoustic remote sensing techniques for classification of the sediments, employing measurement equipment such as single-beam and multibeam echosounders, which are already widely used.

Different approaches towards acoustic remote sensing for classification can be found in the literature. In general, these approaches can be divided into two groups, viz., a phenomenological and a physical approach. In the phenomenological approaches, features such as energy or time spread are determined for the received echo signals. These features are known to be indicative for the sediment type. However, independent measurements, such as cores, are needed to link the sediment classes obtained from the features to real sediment properties or sediment type. In contrast, the physical approaches make use of model calculations and determine the seafloor type by maximizing the match between modeled and measured signals, where seafloor type, or parameters indicative for seafloor type, are input into the model. The advantage of this approach is that in principle, no independent measurements such as sediment samples are needed.

In section 2 the model which is used in the remainder of this article for predicting the signals as received by a single beam echo sounder (SBES) system is described. Section 3 presents a short description of the global optimization method employed for maximizing the match between measured echo signals and model outputs. Section 4 presents a comparison between modeled and measured echo shapes. Hereto, use is made of simulated data. Conclusions and way ahead are presented in section 5.

2. MODELLING THE SINGLE-BEAM ECHOSOUNDER SIGNALS

For the shape of the echo intensity as received by the SBES we can write

\[ y(t) = \int \sigma_b(\theta) B(\theta) \frac{e^{-\alpha r}}{r^4} dA \]  

(1)

with \( \theta \) the angle of incidence, \( \sigma_b(\theta) \) the backscattering cross section, \( A(\theta) \) the instantaneous ensonified area that contributes to the sound received at time \( t \) and \( B(\theta) \) the transmit/receive directivity pattern of the transducer. \( \alpha \) is the water attenuation coefficient and \( r \) is the slant range, i.e., \( r = \sqrt{x^2 + H^2} \) with \( x \) the horizontal distance towards the receiver and \( H \) the water depth.
This can be further worked out as

\[
y(t) = \int_{x_1(t)}^{x_2(t)} \sigma_0 \left( \tan^{-1}\left( \frac{X}{H} \right) \right) B \left( \tan^{-1}\left( \frac{X}{H} \right) \right) \frac{e^{-4\pi x}}{r^4} - 2\pi x \, dx
\]  \hspace{1cm} (2)

Here, \( x_1(t) \) and \( x_2(t) \) denote the two \( x \)-values that bound \( A(\theta) \). For \( x_2(t) \) we have

\[
x_2(t) = \sqrt{\frac{c^2 t^2}{4} - H^2}
\]

with \( c \) the speed of sound in the water, which is assumed to be constant.

\( x_1(t) \) is dependent on \( t \) according to

- For \( t \leq t_0 + T \): \( x_1(t) = 0 \)
- For \( t > t_0 + T \): \( x_1(t) = \sqrt{\left( \frac{ct - cT}{2} \right)^2 - H^2} \)

with \( t_0 = \frac{2H}{c} \) and \( T \) the pulse duration.

\( B(\theta) \) is known given the transducer configuration. In literature several expressions for the backscattering cross section \( \sigma_b(\theta) \) are described. For the work described here, we have considered the backscattering cross section as presented in [1]:

\[
\sigma_b(\theta) = \sigma_r(\theta) + \sigma_v(\theta)
\]  \hspace{1cm} (3)

Thereby, both the backscatter cross section due to interface roughness scattering \( \sigma_r \) and the one due to volume scattering \( \sigma_v \) are accounted for. Both are calculated per unit area and per unit solid angle.

\( \sigma_r \) is obtained by appropriate interpolation between the three following approximations.

- The Kirchhoff approximation, valid for smooth to moderately rough sediments and grazing incidence angles near 90°;
- The composite roughness approximation, valid for smooth to moderately rough sediments and grazing incidence angles away from 90°;
- The large-roughness scattering with a scattering cross section determined from an empirical expression which is derived for rough sediments like gravel and rock.

Parameters that are input into the model are listed in Table 1. While the interface roughness scattering mainly varies with the spectral strength \( w_2 \), the volume parameter \( \sigma_2 \) is the crucial parameter for the volume scattering.
Table 1: Seafloor parameters, their symbols and lower and upper bounds from [1]

<table>
<thead>
<tr>
<th>Seafloor parameter</th>
<th>Symbol</th>
<th>Lower bound</th>
<th>Upper bound</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mean grain size</td>
<td>$M_z$</td>
<td>-1</td>
<td>9</td>
</tr>
<tr>
<td>Sediment – water ratio of mass density</td>
<td>$\rho$</td>
<td>1.145</td>
<td>2.5</td>
</tr>
<tr>
<td>Sediment – water ratio of sound speed</td>
<td>$\nu$</td>
<td>0.98</td>
<td>2.5</td>
</tr>
<tr>
<td>Imaginary to real wave number ratio</td>
<td>$\delta$</td>
<td>0.00148</td>
<td>0.01374</td>
</tr>
<tr>
<td>Sediment volume scattering cross section to attenuation coefficient ratio</td>
<td>$\sigma_2$</td>
<td>0.0002</td>
<td>0.005</td>
</tr>
<tr>
<td>Exponent of the bottom relief spectrum</td>
<td>$\gamma$</td>
<td>n.a.</td>
<td>n.a</td>
</tr>
<tr>
<td>Strength of the bottom relief spectrum [cm$^4$]</td>
<td>$w_2$</td>
<td>5e-5</td>
<td>3e-2</td>
</tr>
</tbody>
</table>

For $\rho$, $\nu$, and $\delta$ empirical expressions exist that couple these parameters to a single parameter, i.e. the mean grain size $M_z[\phi] = -\log_2 d[\text{mm}]$. In general, measurements of these parameters show a small spread around the empirical expressions. In contrast, the spectral strength $w_2$, and the volume parameter $\sigma_2$ are known to deviate more significantly from default values obtained from expressions relating them to $M_z$. A value of 3.25 for $\gamma$ is known to work well for many types of sediment.

Simulated signals are shown in Fig. 1 for three different sediment types. As model inputs use is made of default values for the seafloor parameters [1].

![Simulated signals](image)

**Fig. 1:** Simulated signals for $M_z = -1 \phi$ (solid blue line), $M_z = 3 \phi$ (dashed green line), and $M_z = 9 \phi$ (dash-dotted, red line).

3. **MAXIMISING THE MODEL-DATA AGREEMENT**
In the model based approach, parameters of the seafloor are derived by maximising the match between the measured signal and the modelled signal. Those parameters corresponding to the maximum match should reflect the seafloor composition. As a measure for the agreement between model echo signal and measured echo signal, the following energy function has been selected:

\[ E = \frac{1}{\sum_k [y_{\text{meas}}(t_k) + y_{\text{mod}}(t_k)]} \sum_k [y_{\text{meas}}(t_k) - y_{\text{mod}}(t_k)]^2 \]  

(4)

Here \( y_{\text{meas}} \) and \( y_{\text{mod}} \) denote the measured and modeled echo shape, respectively.

Table 1 lists all input parameters. For the current approach, however, these will not all be inverted for. We limit ourselves to three parameters, i.e., \( w_2 \), \( \sigma_2 \), and \( M_z \), employing the empirical expressions for deriving the values for \( \rho \), \( \nu \), and \( \delta \) from \( M_z \) [1]. This results in three unknown parameters to be inverted for.

For the optimization (minimization of \( E \)) use is made of the global optimization method differential evolution (DE). DE, just like the generic algorithm (GA), starts with an initial population of randomly chosen parameter value combinations [2]. These \( m_{k,i} \) are improved during \( N_G \) successive generations of constant size \( q \), i.e., \( k = 1, ..., N_G \) and \( i = 1, ..., q \).

A partner population is constructed from the initial population \((k = 1)\) according to

\[ p_{k,i} = m_{k,i} + F(m_{k,i_2} - m_{k,i_3}), \quad i = 1, ..., q \]  

(5)

Here, \( p_{k,i} \) is the partner for \( m_{k,i} \), and \( m_{k,i}, m_{k,i_1}, m_{k,i_2}, m_{k,i_3} \) are three different parameter value combinations chosen at random from the population. \( F \) is a scalar multiplication factor between 0 and 1. Higher values for \( F \) result in an increasing difference between the original parameter values and those contained in the partner population. Small values for \( F \) result in parameters in succeeding generations that differ only a little from those in previous generations. This setting actually corresponds to that of a local search, i.e., the exploration of the search space is limited.

Descendants \( d_{k,i} \) result from applying crossover to \( m_{k,i} \) and \( p_{k,i} \) with crossover probability \( p_c \). With DE, crossover leads to parameter values of \( m_{k,i} \) being replaced by parameter values of \( p_{k,i} \). The number of parameter values of \( p_{k,i} \) copied into the new parameter value combination is dependent on the value of \( p_c \). For higher value of \( p_c \) more (on the average) values contained in \( p_{k,i} \) are copied into \( d_{k,i} \). For low \( p_c \)-values generations will differ only slightly from the previous generations. This is the case even if a high value for \( F \) is selected.

Values for the energy function are determined for all descendants. Descendant \( d_{k,i} \) replaces \( m_{k,i} \), becoming its successor, only if its energy is lower. This process is repeated for \( N_G \) generations.

Optimal settings of the DE parameters were derived in [3] and are:

- Population size \( q \): 16
- Multiplication factor $F$: 0.6
- Crossover probability $p_c$: 0.55

Based on preliminary inversion results the number of generations $N_G$ was set to 200.

4. RESULTS

For the current contribution we have limited ourselves to the use of simulated data to assess the performance of the approach. The simulated data have been created for a series of sediment types. These are listed in Table 2.

The SBES considered has the following characteristics

- Transducer diameter: 0.24 m;
- Pulse length: 0.25 ms;
- Frequency: 38 kHz.

For each setting 10 independent optimization runs have been carried out. Figure 2 presents the results of the optimization for setting 3 and 2 as given in Table 2. The plots indicate the typical convergence behaviour of the three parameters. It can be seen that all optimizations converge to the true parameter value, i.e., true values are found precisely. The volume scattering parameter $\sigma_2$ requires a somewhat larger number of generations for convergence than the other two.

For all other settings similar conclusions regarding parameter sensitivity can be drawn. However, the exact parameter sensitivities differ as can be seen in the differences in convergence behaviour of $\sigma_2$ for settings 2 and 3 (lower plots of Fig. 2). The lower $M_z$ of setting 2 limits the contribution of the volume scattering to the total backscatter strength. Consequently, the echo signal is less affected by the volume scattering parameter, resulting in a lower sensitivity of the energy function to this parameter.

<table>
<thead>
<tr>
<th>Setting</th>
<th>$M_z [\phi]$</th>
<th>Grain size [mm]</th>
<th>$w_2 [\text{cm}^4]$</th>
<th>$\sigma_2$</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>8</td>
<td>0.0039</td>
<td>0.0005</td>
<td>0.0005</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
<td>0.2500</td>
<td>0.0035</td>
<td>0.0020</td>
</tr>
<tr>
<td>3</td>
<td>5</td>
<td>0.0313</td>
<td>0.0005</td>
<td>0.0020</td>
</tr>
<tr>
<td>4</td>
<td>5</td>
<td>0.0313</td>
<td>0.0050</td>
<td>0.0020</td>
</tr>
</tbody>
</table>

Table 2: True and inverted seafloor sediment parameters for the 4 sets of inversions.
Fig. 2: Inversion results. The plots indicate the parameters corresponding to the lowest energy as a function of generation for two settings as given in Table 2 (left: setting 3, right: setting 2).
5. CONCLUSIONS AND OUTLOOK

It can be concluded, based on the simulations presented in this paper, that seafloor classification using a model-based approach on single-beam echosounder data is feasible. All unknown parameters can be retrieved correctly. As a next step the method will be applied to real SBES data. These data have been acquired in the Cleaver bank area (North Sea). This area is an attractive area for testing sediment classification methods as it contains a large range of sediment types with mean grain sizes ranging from 5 φ to -1 φ [4].

REFERENCES

Structured Session 5

The historical Developments in Underwater Acoustic Propagation Modelling

Organizer: Mike Buckingham
Underwater Acoustic Wave Propagation Prediction: Models and Results

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April 3, 2009

ABSTRACT

Besides experimental observations and measurements, the result of under acoustic wave propagation can be obtained by available prediction models. Most available models solve their representative wave equations. Using any two different models to predict the result of the same underwater acoustic propagation problem under the same environment, one expects that the results, produced by these two different models, turn out to be the same. In some cases, these two results do not agree. This raises users' concern about the accuracy of the result. This paper analyzes the disagreement and discusses what can be considered to be an acceptable satisfactory result. To make the result satisfactory, this paper recommends using a new numerical procedure to ascertain the objective. An available prediction model is selected to illustrate the effectiveness and desirability of this procedure.

1 Introduction

Underwater acoustic wave propagation result can be obtained through experimental observations and measurements that are realistic because both experiments and measurements are taken under the actual physical environment. Besides experimental observations and measurements, the result can also be obtained by available underwater acoustic wave propagation prediction models. Historically, a number of models have been developed for predicting underwater acoustic wave propagation. Most prediction models solve representative wave equations. Using any two different models to predict the result of the same underwater acoustic wave propagation problem under the same environment, one, naturally, expects that the results produced by these two different models turn out to be the same. In some cases, there is a disagreement between the two results. The disagreement can occur if these two prediction models are solving a not well-posed problem [2]. This causes the users' concern about the accuracy of the result. This paper analyzes the disagreement and discusses the accuracy consideration, comments on the models' prediction results to what extent a result can be considered acceptable. This discussion leads to the recommendation to apply a newly introduced desirable numerical procedure to obtain satisfactory results [1]. What the recommended procedure can do will be illustrated.

This paper begins with a discussion on modeling, models, models' capabilities, and acceptable satisfactory results. A new procedure described in reference [1], is recommended here to obtain the required satisfactory result. An available 3-dimensional underwater acoustic wave propagation prediction model is selected to illustrate what the procedure can do and to show how an acceptable
2 Modeling and Models

Generally speaking, modelers develop a model aiming at producing satisfactory results for the problem under consideration. There is no exception to develop a model to predict underwater acoustic wave propagation. Available underwater acoustic wave propagation models are mostly mathematical. One solves the propagation problem by solving a representative wave equation. Developing such a model under general ocean environment is very desirable but very complicated. To model the entire ocean physics into the model is very complicated and difficult to do; to include the ocean physics as complete as possible is what we want to do. So far there does not exist a model which is general enough to handle all underwater acoustics wave propagation problems.

In reality, prediction models have been specifically developed for special applications. There exist a number of such prediction models, which can be used for the same special applications. Each model has its features, its advantages, its capabilities, and its limitations. Among these available models some of them are better than the others in the sense that they are more useful or accurate. Generally speaking, the success of a model depends upon how easily it can be used; more importantly how accurate are its predictions. Model users must note that every model has its range of validity; one should not use the model outside its range of validity.

3 Models’ Capabilities and Predicted Results

It is very desirable that a model has useful capabilities as many as possible. In fact it is very difficult to achieve this objective. However, models can be developed to suit special purposes. Many underwater wave propagation prediction models are available in this category.

By all means a user uses a model for his/her prediction, the predicted result is expected to be satisfactory. In fact, whatever the result turns out to be, the user has no questions but accepts the result based on the assumption that the user is comfortable with the model the user is using.

When a model is developed, before the model to be released for public use, the model developer must have checked the model’s capability as well as its accuracy. Knowing the capability and the accuracy of a selected model, a user can feel comfortable to use this model for his/her application.

Among the available underwater wave propagation prediction models, if one selects two different prediction models possessing the same capability to solve the same problem under the same environment, the user can expect that the results, produced by these two different models, turned out to be the same. What shall we say if the results are not the same? Offhand, the user may think that at most one is correct, the other is not; it may be possible that both results are not correct. It is reasonable to assume that these available prediction models are solving well-posed problems [2]. Depending upon the method of solution, the results may differ in accuracy. Using the word CORRECT is correct; using the words NOT CORRECT may not be true. After we give an analysis, the NOT CORRECT result may still be acceptable. This explanation may become clear if the reader reads the analysis and the discussion below.

Let us introduce a couple keywords to describe the result disagreement. We use the word ACCURATE to replace the word CORRECT. We use the word "LESS-ACCURATE" to replace the
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words "not correct". Thus, judging from the disagreement, we can say one result may be accurate; the other is less-accurate. This presents a possibility to suggest that less-accurate results may be considered acceptable depending on the accuracy requirement.

What is considered to be an acceptable satisfactory result?

To answer this question we start with discussing how accurate a predicted result can be accepted? In the field of applied mathematics, if a model is developed for solving differential equations, the model has to pass a set of severe test problems to meet the accuracy requirement. In some fields of science extreme accuracy is required; that is why packages of severe test problems are available. In the acoustics field, a model can produce an extreme accuracy that may not be needed. Knowledgeable acousticians recommend checking a newly developed model result with the reliable measured data. In fact, to obtain measured data depends upon a number of factors; the measuring equipments, the human inaccuracy, and the environmental conditions, ... etc.; all those factors contribute some inaccuracies. However, to obtain measured data is done in realistic physical environments; the data can be accepted as a reference solution. If a set of measured data is considered a reference solution, then the predicted result can be compared with the reference solution to determine whether or not the prediction result is satisfactory?

What result can be considered acceptable?

There is no rule to follow what is an acceptable figure for the result. A suggested acceptable figure can be "3 dB" or less. Now, we have "3 dB" to go by.

Our aim is to discuss the underwater acoustic wave propagation prediction "models and results"; the following questions come up:

(1) If a model is chosen to predict the underwater acoustic wave propagation, how comfortable does the user feel about the predicted result?

(2) There is no recognized reference solution available that can be used for comparison. How can the user tell that the result is within the reference "3 dB" figure?

(3) In the event the user found the result not satisfactory, what can the user do?

All of the above questions can be answered after the new procedure to improve the accuracy is introduced.

4 A Numerical Accuracy-improving Procedure

This section describes a new procedure [1] to correct the result until the satisfactory result is obtained. To illustrate this desirable procedure, an existing prediction 3-dimensional model, FOR3D, is selected for demonstration purpose.

This procedure is to employ the Predictor-Corrector Method[9] to reach the goal. An explicit method (predictor) is used to predict the result. Then, the explicit-associated implicit method (Corrector) is applied to examine the predicted result. If the result is satisfactory within accuracy requirement, the result is considered satisfactory. Otherwise, the corrector continues to correct the answer until it meets the accuracy requirement. For accurate predicted result, the result needs to
be corrected only once. As introduced in reference [1], the procedure is easy to adopt and works very well. The illustration of this procedure clearly demonstrates its advantage.

4.1 A Selected Prediction Model

A historical development was the introduction of the Parabolic Equation Approximation Method [3] to transform an elliptic wave equation into a set of parabolic equations which appear to be computer memory saving and easy to handle long-range wave propagation under range-dependent environment. This method is known as PE. Since the 1970's, there have been a number of PE models developed. References [4, 5, 6] give comprehensive information of these models up to 2004. The PE model selected for demonstration purpose in this paper belongs to this category.

Prior to the description of the selected 3-dimensional prediction model, definitions of a couple of operators, named $X$ and $Y$ are defined below.

$$X = (n^2(u, \theta, z) - 1) + \frac{1}{r^2 k_0^2} \frac{\partial^2}{\partial z^2}$$

$$Y = \frac{1}{r^2 k_0^2} \frac{\partial^2}{\partial \theta^2}$$

(4.1)

where $k_0$ is the reference wavenumber, $n(r, \theta, z)$ is the index of refraction, $r$ is the range variable, $z$ is the depth variable, and $\theta$ is the azimuth variable.

Let $u = u(r, \theta, z)$ be the 3-dimensional wave field. A selected PE model to represent the underwater acoustic wave propagation has the expression:

$$\frac{\partial}{\partial r} u = -ik_0(\sqrt{X + Y})u.$$  (4.2)

Applying the rational function approximation to treat the square-root operator, we obtain

$$\sqrt{X + Y} \approx (1 + \frac{1}{2} X - \frac{1}{8} X^2 + \frac{1}{2} Y)$$  (4.3)

Then, Eq. (4.2) can be written as

$$\frac{\partial u}{\partial r} = -ik_0(1 + \frac{1}{2} X - \frac{1}{8} X^2)g + \frac{1}{2} g$$  (4.4)

where $g = g(u, r, \theta, z) = -\frac{1}{2} ik_0 Y u$.

4.1.1 A Numerical Solution

Let $\Delta r$ be the range step size and $\delta = -ik_0 \Delta r$. Reference [1] applied the first-order GAB (Generalized Adam-Bashforth method), an EXPLICIT method, to solve Eq. (4.4) which gives the following solution:

$$u^{n+1} = e^{\delta} e^{(1+\frac{1}{8}X - \frac{1}{8}X^2)} u^n + \Delta r g^n.$$  (4.5)

A computer code was developed to implement the solution of Eq. (4.5). This entire code uses the combination of Finite difference discretization for partial derivatives, Ordinary-differential-equation
method, and Rational function approximations for the square-root operator and exponential operator to solve a 3-Dimensional underwater acoustic wave propagation representative equation: thus, the synopsis FOR3D is used.

The FOR3D computer code uses only the explicit method, the Predictor, to solve Eq. (4.4). At that time, the Corrector method did not come to our mind and was not needed for the reason that the computer code produces satisfactorily acceptable results. This can be seen after the Correct procedure is introduced.

In order to carry out the application of the Predictor-Corrector procedure, an implicit method, called the CORRECTOR, has to be introduced. Following the reference [1], we use the first-order Implicit Generalized Adam-Moulton (GAM) method as the corrector. Therefore, the application of the GAM to solve Eq. (4.4) gives

\[ u^{n+1} = e^{t} \left[ \sinh(1+\frac{x}{2}) - \frac{1}{2} \Delta r (g^{n} + g^{n+1}) \right]. \]  

(4.6)

Note that in both equations (4.5) and (4.6) the \( g^{n} \)-function has the following expression:

\[ g^{n} = \frac{1}{2} \frac{i k_{0}}{(r_{n} k_{0})^{2}} \frac{\partial^{2} u^{n}}{\partial \theta^{2}}. \]  

(4.7)

where \( r_{n} \) is the range value at the present range.

Also note that the \( g^{n+1} \)-function in Eq. (4.6) has the following expression:

\[ g^{n+1} = \frac{1}{2} \frac{i k_{0}}{(r_{n+1} k_{0})^{2}} \frac{\partial^{2} u^{n+1}}{\partial \theta^{2}}. \]  

(4.8)

The FOR3D model formulates Eq. (4.5) into a marching scheme after the square-root and exponential operators are approximated by rational function approximations. Then, the marching scheme is solved by a system of equations. Over there both sides have range-dependent functions. Computationally all range-dependent functions are evaluated at the mid-range point. Therefore, the \( r_{n} \) in the \( g^{n} \) and \( r_{n+1} \) in the \( g^{n+1} \) are evaluated at \( (r_{n} + \frac{1}{2} \Delta r) \). Therefore, \( r_{n} = r_{n+1} = r_{n+1/2} \).

Prior to the discussion and illustration of the Predictor-Corrector Method, we want to distinguish the wave fields, produced by both explicit (predictor) and implicit (corrector) methods. Let the wave field \( u^{n+1} \) be the wave field produced by the predictor formula (4.5). Let \( u^{c} \) be the wave field produced by the corrector formula (4.6). The purpose is to examine the difference between \( u^{n+1} \) and \( u^{c} \). How to estimate the difference between \( u^{n+1} \) and \( u^{c} \) is going to be addressed later.

There is enough information for us to discuss the Predictor-Corrector Method and the Predictor-Corrector Procedure.

**What is the Predictor-Corrector Method?**

The predictor-corrector method is using two methods to solve the same Eq. (4.4). An explicit method is used to solve the equation as a predictor. Then, the predictor solution is used as an input in the implicit formula to solve the same equation. As an illustration, Eq. (4.5) is used to solve Eq. (4.4) (the wave field solution is indicated by \( u^{n+1} \)). Then, the \( u^{n+1} \) is used in the \( g^{n+1} \) on the right hand side of Eq. (4.6) to solve the same Eq. (4.4).
How the Predictor-Corrector procedure works?

As an illustration, the predictor Eq. (4.5) is applied to obtain the $u^{n+1}$ for Eq. (4.4). Then, the corrector Eq. (4.6) is used to obtain the $u^n$ making use of the predicted $u^{n+1}$. Once the $u^{n+1}$ and the $u^n$ are obtained, compare these two results for satisfactory accuracy. If a predictor predicts satisfactory result, it needs to be corrected only once. If not satisfactory, continue to correct until the satisfactory result is obtained. Continuing to correct may be needed many times, the user can put a cap on the number of corrections. A variable-step size procedure [1] may be applied to minimize the number of corrections and to obtain the acceptable result.

How to determine whether or not the result is acceptable?

The criterion is to compare $u^n$ against $u^{n+1}$. If $|u^n - u^{n+1}|$ is less than the allowable tolerance, then, the $u^{n+1}$ is accepted as a satisfactory solution. A natural question arises:

How to relate the "allowable tolerance" to the "dB figure"?

The answer to this question can be found in reference [10], where a table is constructed to list the "number of significant digits" in computation vs. "dB figures". From the table, if a "3 dB" figure is asked for, the computational result has to keep 2 significant digits. There is no way to tell the acceptable number of significant digits if only predictor is used. It needs a corrector to adjust the acceptable number of significant digits to meet the requirement.

Next, we discuss the allowable tolerance and the related predictor-corrector predicted result. The allowable tolerance is pre-determined by the user. The predictor-corrector procedure is to produce the result and to improve it, if necessary, until the allowable tolerance is met. A mathematical analysis will serve the purpose to demonstrate how to reach the objective.

Solving Eq. (4.4) by the Predictor-Corrector method is to use Eq. (4.5) as a predictor and use Eq. (4.6) as a corrector. The predictor produces the result, indicated by $u^{n+1}$. The predictor result is used as a part of input for the corrector. The result produced by the corrector is indicated by $u^n$. To estimate the $|u^n - u^{n+1}|$, we derive a mathematical expression for the difference. Subtracting Eq. (4.5) from Eq. (4.6) gives:

\[ u^n - u^{n+1} = \frac{1}{2} \Delta r (g^{n+1} - g^n). \]  

(4.9)

Using the expression in (4.7) for $g^n$ and the expression (4.8) for $g^{n+1}$, Eq. (4.9) can be written as

\[ u^n - u^{n+1} = \frac{1}{2} \Delta r (ik_0 \frac{1}{(r^{n+1}k_0)^2} \frac{\partial^2 u^{n+1}}{\partial \theta^2} - ik_0 \frac{1}{(r^n k_0)^2} \frac{\partial^2 u^n}{\partial \theta^2}). \]  

(4.10)

In view of handling the range-dependent function by evaluating the function at the mid-range point, i.e., $r_n = r_{n+\frac{1}{2}} = r_{n+1}$, then, Eq. (4.10) can be written as

\[ u^n - u^{n+1} = \frac{1}{2} \Delta r (ik_0 \frac{1}{(r_{n+\frac{1}{2}} k_0)^2} \frac{\partial^2 (u^{n+1} - u^n)}{\partial \theta^2}). \]  

(4.11)
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Once the allowable tolerance is determined, the control of the influential factors on the right hand side of Eq. (4.11) can be made small enough in order to meet the tolerance requirement where the \( \Delta r \) is an important factor.

First, take a look at the expression inside \([\ldots]\) in Eq. (4.11), indicated by \( \alpha \) such that \( \alpha = \frac{1}{(r_{n+\frac{1}{2}})^{2}} \). We want to estimate the \( \alpha \). Rewrite \( \alpha \) in the following expression:

\[
\alpha = \frac{1}{r_{n+\frac{1}{2}}^{2}} \frac{1}{r_{n+\frac{1}{2}}^{2}}.
\]

(4.12)

In Eq. (4.12), the term \( \frac{1}{r_{n+\frac{1}{2}}^{2}} \) is \(< 1\) because the range function, \( r_{n+\frac{1}{2}} \), is \( > 1 \). As the range advances gradually, the \( \frac{1}{r_{n+\frac{1}{2}}^{2}} \) becomes smaller. This analysis justifies that \( \frac{1}{r_{n+\frac{1}{2}}^{2}} < 1 \).

Furthermore, we take a look at another expression, \( \frac{1}{r_{n+\frac{1}{2}}^{2}} \) in Eq. (4.12). Equation (4.2) has been derived by the parabolic equation approximation, the far-field condition, \( |\rho_{0}| > 1 \) must be obeyed. Thus, the expression, \( \frac{1}{r_{n+\frac{1}{2}}^{2}} \) must be \(< 1 \). As the range advances this expression is becoming much, much smaller \(< 1 \).

From the above analysis, \( \alpha \) is \(< 1 \) and much \(< 1 \). It remains by looking at the remaining expressions, \( \frac{\Delta r}{\alpha} \frac{2}{\alpha_{n+1}} (a_{n+1} - a_{n}) \). It is seen that the range step size \( \Delta r \) can be made small enough to assure, not only this term but also the right hand side of \( \alpha \), to be small. Reference [1] introduced a computational technique to find the smallest acceptable \( \Delta r \). Some details of this new computational procedure is not discussed in this paper; this paper merely demonstrates that it is possible to select the desirable range-step size to obtain the satisfactory result.

5 End Comments

The advantage of using the Predictor-Corrector Method is to obtain required satisfactory result that has been shown in the demonstrative example. Not every mathematical model is readily available to apply the Predictor-Corrector procedure. Some effort is needed to transform the representative equation into an equation such that it is recognizable by the Predictor-Corrector Method.

As shown in the predictor-corrector procedure, as long as a reasonable size of the \( \Delta r \) is selected, the solution shall turn out to be satisfactory. In this case, the solution needs to correct only once.

It is assumed that each prediction model is solving a well-posed problem; the selected model for illustration is in this category. It must also be assumed that before the model is released to public, the modeler has checked thoroughly the model’s capability and accuracy; therefore there should be no question about the prediction result; the question is whether or not the user considers it acceptable?

To determine whether or not the predicted result is acceptable, the application of the Predictor-Corrector Method is recommended which can help the user to make a decision. If the user needs the predicted result to meet his/her accuracy requirement, the Predictor-Corrector Method can further, obtain the user required satisfactory result.
CONCLUSION

Any existing underwater acoustic wave propagation prediction model can be used to obtain propagation result. In the event if the user wants the result to meet his/her accuracy requirement, a computationally stable Predictor-Corrector Method is recommended.

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Applications of an exact integral transform between solutions of parabolic and elliptic wave equations

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THREE-DIMENSIONAL SOUND PROPAGATION AND SCATTERING AROUND A CONICAL SEAMOUNT

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Abstract: A three-dimensional propagation and scattering model is developed for an ocean with axisymmetric bathymetry and an offset acoustic source. Based on the same theoretical foundation as the formulation presented by Taroudakis [M. I. Taroudakis, J. Comput. Acoust. 4, 101-121 (1996)], the present approach combines a spectral decomposition in azimuth with a coupled-mode theory for two-way, range-dependent propagation. However, the earlier formulations were severely limited in terms of frequency, size and geometry of the seamount, the seabed composition, and the distance between the source and the seamount, and were therefore severely limited in regard to realistic seamount problems. Without changing the fundamental theoretical foundation, this approach applies a number of modifications to the numerical formulation, leading to orders of magnitude in numerical efficiency for realistic problems. Further, by using a standard normal mode model for determining the fundamental modal solutions and coupling matrices, and by applying a simple superposition principle, the computational requirements are made independent of the distance between the seamount and the source and receivers, and dependent only on the geometry of the seamount and the source frequency. Therefore, realistic propagation and scattering scenarios can be modeled, including effects of seamount roughness and realistic sedimentary structure. Numerical examples show that strong mode coupling may occur at the boundary of a seamount. In addition, coherent artificial backscatter can be diffused in the model by the use of stair steps of random horizontal step sizes, which results in a faster convergence rate in terms of the number of ring-shaped sectors. [Work supported by the Office of Naval Research.]

Keywords: three-dimensional propagation and scattering, seamount, mode coupling
1. INTRODUCTION

Over the past several decades, a large number of numerical models have been developed for acoustic propagation in the ocean. Most of these models provide solutions for two-dimensional (2D) (range and depth) problems, and they provide satisfactory solutions for the majority of propagation problems where the environmental dependence on azimuth is insignificant.

However, there are several classes of propagation problems where three-dimensional (3D) effects cannot be ignored. These include, for example, propagation over and around a seamount, or through a strong eddy. For such problems, the 2D models, in which energy does not couple between planes of constant azimuth with respect to the source, often fail to provide accurate solutions, whereas a 3D model is able to provide accurate solutions for the field in range, depth, and azimuth. An important 3D effect, as shown in [1] and this paper, is that the azimuthal coupling increases the width of shadows behind islands and seamounts, which can be explained physically by horizontal refraction [2][3].

The $N \times 2D$ approach introduced first by Perkins and Baer [4] assumes out-of-plane scattering to be insignificant, and is therefore applicable only to problems with weak transverse environmental variability.

In 1996, a coupled-mode formulation for the solution of the Helmholtz equation in the presence of a conical seamount was developed by Taroudakis [5]. In his work, the conical seamount was divided into a number of rings, in each of which a series expansion of the acoustic pressure in terms of normal modes and an azimuthal Fourier series was applied. The expansion coefficients were obtained by solving linear systems of equations resulting from the continuity conditions at the vertical interfaces separating the ring-shaped environmental sectors. The formulation introduced was theoretically exact, but the numerical implementation was inefficient, and associated with numerical stability issues, which severely limited the applicability to realistic seamount problems. Firstly, the applied expansion involved Hankel functions of high orders, for which the numerical evaluation is inherently unstable for high Fourier orders and small arguments. Secondly, when the source is far from the seamount, the number of azimuthal modes required for convergence is too large to make this formulation applicable. Finally, the choice of radial Hankel functions in Taroudakis' formulation yields unstable solutions of the mode coupling equations at high azimuthal orders.

To alleviate some of these issues, Eskenazi modified Taroudakis' model by applying the direct global matrix (DGM) approach [6] to obtain stable coupled-mode systems [7]. However, Eskenazi's model was still inefficient for distant source problems, and the use of the global coupled-mode equations still limited the applicability to relatively simple, canonical seamount problems.

Here a 3D spectral coupled-mode model is developed, based on the same theoretical foundation as used by previous investigators, but applying a number of numerical tools which extend the numerical feasibility to problems characterized by large Fourier orders, such as those involving distant sources and realistic seamounts of significant extent. The seamount propagation problem is reformulated into a scattering problem. As demonstrated in this paper, this reformulation has significant impact in terms of numerical efficiency. The modifications are as follows.

(i) Functions $J_m(\cdot)$ and $H^{(1)}_m(\cdot)$ are used as the two linearly independent range solutions, instead of $H^{(1)}_m(\cdot)$ and $H^{(2)}_m(\cdot)$ in Taroudakis' approach, due to the fact that $J_m(\cdot)$ and
\( H_m^{(1)}(\cdot) \) remain linearly independent numerically for both large and small arguments [8], whereas the two Hankel functions become numerically indistinguishable at high order \( m \).

(ii) Normalized Bessel and Hankel functions are used to avoid overflow and underflow problems; in addition, the asymptotic forms of the normalized Bessel and Hankel functions for small and large arguments are used.

(iii) The efficiency is improved dramatically by introducing the superposition representation of the external field with respect to the seamount [9][10][11].

(iv) The single-scatter approximation used in this model vastly improves numerical efficiency, and as has been shown in the past, is highly accurate for most ocean acoustic propagation problems.

(v) A noteworthy feature of the present approach, which is also true in previous models, is that the coupling matrices are independent of azimuthal orders, so they need to be calculated only once.

(vi) This model is perfectly scalable, and therefore it can be easily parallelized to run on computer clusters, making it applicable to large-scale 3D problems.

2. THEORY

The geometry considered involves a point source, offset horizontally relative to a conical seamount, as illustrated in Fig. 1. This is a 3D problem in which the acoustic field depends not only on range and depth, but also on azimuth.

![Fig. 1: An ocean waveguide with a conical seamount, which is approximated by a number of range-independent ring-shaped sectors.](image)

In the coupled-mode approach, a number of range-independent ring-shaped sectors are used to approximate such a conical seamount. The notation \( r^j \) is used to denote the range at the interface between ring \( j \) and ring \( j+1 \), for \( j=1,2,...,K-1 \), with \( K \) denoting the total number of ring-shaped sectors.

A cylindrical coordinate system is introduced, centered at the axis of the seamount. The location of the point source is denoted by \( r_s = (r_s, z_s, \phi_s) \), and the location of a field point is
denoted by \( \mathbf{r} = (r, z, \varphi) \). The 3D Helmholtz equation in the outer medium containing the point source is \([5][10][11]\)

\[
\frac{1}{r} \frac{\partial}{\partial r} \left( r \frac{\partial p}{\partial r} \right) + \frac{1}{r^2} \frac{\partial^2 p}{\partial \varphi^2} + \rho(z) \frac{\partial}{\partial z} \left( \frac{1}{\rho(z)} \frac{\partial p}{\partial z} \right) + \frac{\omega^2}{c^2(z)} p = -\frac{\delta(r-r_s)}{r} \delta(z-z_s) \delta(\varphi-\varphi_s),
\]

where \( p = p(r, z, \varphi) \) is the acoustic pressure (factoring out the harmonic time dependence \( e^{-i\omega \tau} \), \( \omega \) being the angular frequency), and \( \rho(z) \) and \( c(z) \) are density and sound speed profiles in the outer source ring, respectively.

In ring \( j \), which is a range-independent environment, the 3D Helmholtz equation is the homogeneous version of Eq. (1), with \( \rho(z) \), \( c(z) \), and \( p(r, z, \varphi) \) being replaced by \( \rho^j(z) \), \( c^j(z) \), and \( p^j(r, z, \varphi) \), respectively.

The environmental representations used by Taroudakis [5] and Eskenazi [7] are very similar. Both divided the seamount environment into three regions, one being the innermost, cylindrical sector containing the summit of the seamount, one containing all the ring-shaped sectors between the central cylinder and the source range, and the third being a semi-infinite ring outside the source range. The difference between the two lies in the numerical implementation of the spectral coupled-mode representation of the field. Taroudakis [5] used unnormalized Hankel functions of the first and second kind in all sectors for representing the range-dependence of the field. However, as demonstrated by Ricks et al. [8], this choice is numerically unstable at high Fourier orders. Recognizing this, Eskenazi [7] used the stable, normalized Bessel function of the first kind and Hankel function of the first kind.

The present representation of the field differs from Taroudakis' [5] and Eskenazi's [7] in that it eliminates the virtual sector boundary at the source range, and instead extends the outermost sector to the base of the seamount. Therefore, in this representation, region I is the innermost cylindrical sector, and region II contains all the ring-shaped sectors between the central cylinder and the base of the seamount, and region III is a semi-infinite ring outside the seamount. The acoustic field in region III is then represented as a superposition of the unperturbed field produced by the source in the absence of the seamount, and a scattered field produced by the seamount. This basically reformulates the seamount propagation problem into a scattering problem. Although rather trivial, this reformulation has significant impact in terms of numerical efficiency, as demonstrated in the following.

By using \( K \) to denote the total number of sectors and \( r_j \) to denote the range of the base of a seamount, the pressure field is expressed as follows.

(I) Inner cylinder \( r \leq r^1 \),

\[
p^1(r, z, \varphi) = \sum_{m=0}^{\infty} \sum_{n=1}^{\infty} b^1_{ms} j^1_m(r) \Psi^1_n(z) \Phi_m(\varphi).
\]

(II) Intermediate sectors \( r^1 < r \leq r_j \), where \( r_j \) is the radius of the base of the seamount. In ring \( j \), i.e., \( r^{j-1} < r \leq r^j \),
\[ p'(r, z, \varphi) = \sum_{m=0}^{\infty} \sum_{n=1}^{\infty} \left[ a_{mn}^j H_{mn}^j(r) + b_{mn}^j J_{mn}^j(r) \right] \Psi_n^j(z) \Phi_m(\varphi). \]  

(4) 

(III) Outer region containing source \( r > r_j \), 

\[ p(r, z, \varphi) = p_i(r', z) + \sum_{m=0}^{\infty} \sum_{n=1}^{\infty} a_{mn}^K H_{mn}^K(r) \Psi_n^K(z) \Phi_m(\varphi), \]  

(5) 

where \( p_i(r', z) \) is the 2D normal mode solution in the absence of the seamount [6], and \( r' \) is the range of a field point from the source. 

Here, \( a_{mn}^j \) and \( b_{mn}^j \) are coupling coefficients which are determined by applying source conditions and boundary conditions at each vertical interface. Functions \( \Psi_n^j(z) \) are local, depth-dependent eigenfunctions and \( \Phi_m(\varphi) \) are azimuthal eigenfunctions [10][11].

3. NUMERICAL IMPLEMENTATION

Compared to the earlier formulations, there are three principal reasons for the improved numerical efficiency of the present formulation. The first is the use of the superposition principle for handling the source field. The second is the use of asymptotic representations of the normalized range functions. The third is the use of a marching solution of the coupled-mode equations, based on the single-scatter approximation.

In Eskenazi’s and Taroudakis’ formulations, convergence for a field point outside the seamount, i.e. \( r > r_j \), requires the number of azimuthal orders of at least \( k_0 r_j \). In contrast, the use of the superposition principle reduces the minimum number of azimuthal orders to \( k_0 r_j \), where \( r_j \) is the radius of the base of the conical seamount. Therefore, for a fixed source frequency, the number of azimuthal orders required depends only on the size of the seamount, not the source distance. For realistic seamount problems this can lead to orders of magnitude in numerical efficiency.

Both of Eskenazi’s and Taroudakis’ formulations suffered from potential overflow and underflow problems in the numerical evaluation of the radial Bessel and Hankel functions. Although providing numerical stability of the solution for the modal coefficients, the use of the normalized Bessel and Hankel functions does not in itself solve this numerical problem for small and large arguments with respect to Fourier order. However, using the asymptotic representations for the Bessel and Hankel functions leads to closed form asymptotics for the normalized functions as well, therefore eliminating the overflow and underflow problems [10][11].

4. NUMERICAL RESULTS

The schematic of the problem is shown in Fig. 2. The waveguide consists of a 5000 m inhomogeneous water column limited above by a pressure-release flat sea surface and below by a homogeneous fluid half space with a compressional speed of 2000 m/s, a density of 1
$g/cm^3$, and an attenuation of $0.1 \text{ dB/} \lambda$. A false bottom at depth 7000 m is introduced to take into account the contribution from the continuous spectrum. The seamount is 100 km from the source, with a radius at the base of 20 km, and the same acoustic properties as the bottom. Two different heights of the seamount are considered, $H = 1000 \text{ m}$ and $H = 3800 \text{ m}$, respectively. The source depth is 100 m and the source frequency is 10 Hz.

![Fig. 2: Schematic of a deep water waveguide with a conical seamount and a penetrable bottom.](image)

In this problem, the modal starting field is limited to contain only the waterborne modes, which are the first 12 modes; otherwise a maximum of 74 modes are used. In addition, for the purpose of azimuthal convergence, the number of azimuthal modes is taken to be 898.

To compare the results of the $N \times 2D$ model and those of the 3D model introduced in this paper, the height of the seamount is set to vary from 1000 m to 3800 m. Results of transmission loss (TL) in the horizontal plane at depth 300 m are shown in Fig. 3.

From Fig. 3 (a) and Fig. 3 (b) it can be seen that when the height of the seamount is 1000 m, which is relatively small, the 3D effects are insignificant and the $N \times 2D$ model is a good approximation of the 3D model. However, as the height of the seamount rises to 3800 m, from the results shown in Fig. 3 (c) and Fig. 3 (d), it can be seen that the approximation of the $N \times 2D$ model to the 3D model deteriorates, and the span of the shadow zone in the 3D result is more pronounced than in the $N \times 2D$ result. This example indicates that the azimuthal coupling can be important for seamount problems, and the broadening of shadows behind seamounts by horizontal refraction can be captured by implementing a true 3D model.
5. CONCLUSIONS

The coupled normal mode method is a very efficient, simple and accurate method to solve range-dependent problems. First, since the normal mode method solves the full-wave Helmholtz equation, it is applicable to cases where the backscattering is not negligible. Second, it is free of angular limitations for propagation angles less than critical, and therefore provides accurate solutions at long ranges. This is achieved by including evanescent modes by introducing a false bottom or by calculating the branch-line integral. Third, as an important feature of the spectral coupled-mode approach, the coupling matrices are independent of azimuthal orders, which makes this approach applicable to problems of scattering from azimuthal symmetric features.

Here a spectral coupled-mode model has been developed and applied to analyze the propagation and scattering around conical seamounts. Although based on a theoretical foundation identical to that used in earlier related work, this approach provides more efficient and stable numerical solutions for realistic seamount problems. The keys to its superior performance are the fundamental superposition principle applied to introduce the source, the choice of normalized range solutions which can be evaluated accurately and remain numerically stable even at high azimuthal orders, and the use of a marching, single-scatter approximation for the solution of the coupled-mode equations.
The numerical examples show that azimuthal coupling can be important for seamount problems. When the azimuthal variation is weak, the out-of-plane scattering is negligible and therefore, as shown by the results, the $N \times 2D$ approach is a good approximation of the true 3D approach. However, when the azimuthal variation is strong, azimuthal coupling cannot be ignored. In this case, the shadows behind seamounts are broadened by horizontal refraction. A true 3D approach, instead of the $N \times 2D$ approach, can capture this feature. The dependence of the 3D effects on the environment is also illustrated through the numerical example, which shows that for seamounts with steep slopes and reaching far into the sound channel in terms of wavelengths, the 3D effects are significant. In a situation such as this, a true 3D model is required, rather than an $N \times 2D$ model.

6. ACKNOWLEDGEMENTS

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WAVEFRONT MODELLING AND LOW FREQUENCY RAY THEORY

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Abstract: For many years ray theory in underwater acoustics was assumed to be inapplicable at low frequencies because the conventional derivations of the equations of ray tracing involve a high frequency approximation. The inclusion of beam displacement in shallow water modelling showed that ray theory could be valid at low frequencies. Wavefront modelling was developed from a solution of the wave equation which does not assume high frequencies. The acoustic field is obtained as a sum of terms, each of which is a phase integral corresponding to a ray path with a given sequence of turning points. A conventional ray trace is used to find an approximation for each phase integral which is then evaluated analytically to give the amplitude, phase and arrival time for each ray contribution. The method handles multiple arrivals and gives accurate results for caustics, foci and shadow zones all of which are effectively low frequency situations because they involve path differences of less than a wavelength. Recent applications describe the detailed interference arising from reflection of short pulses from surface waves and give good agreement with experimental waveforms.

Keywords: Rays, low frequency, wavefronts
1. INTRODUCTION

Ray tracing has provided a physically intuitive picture of underwater sound propagation for many years but has always been regarded as a high frequency approximation for two main reasons. First, the conventional derivation of the ray trace equations assumes a high frequency approximation. Second, geometric ray tracing gives rise to shadow zones with no sound penetration and this can only be true in the high frequency limit. However, the main difficulty with simple ray theory is that it predicts infinite intensity at caustics. Several authors described methods of modifying ray theory to calculate the field at caustics and in shadows using complex rays and analytic continuation but none were easy to include in routine calculations. These approximations and difficulties with ray theory saw other methods such as normal mode theory and PE preferred for accurate calculations.

An early indication that ray theory could be applied at low frequencies was found when it was shown that the attenuation rate of modes and their equivalent rays in a Pekeris model were identical if beam displacement was included in the ray path. Ray tracing with beam displacement was then used to show that receiver waveforms in a shallow water wave guide could be calculated equally accurately using rays or normal modes.

Another physically intuitive but much less familiar picture of propagation is provided by the time evolution of wavefronts. Sound energy propagates away from a source as a series of wavefronts which turn up and down either by reflection or refraction. The folded wavefront evolves in time and shows how each section of the wavefront passes a receiver. This is particularly clear in deep water where the early arrivals on a vertical array are interpreted as wavefronts sweeping upwards or downwards past the array. Given the clear physical significance of wavefronts it is surprising that wavefront modelling has been developed only recently.

The first case considered in wavefront modelling was long range SOFAR propagation in which the relevant equivalent rays had turning points well away from the surface or bottom. Receiver waveforms were accurately modelled even near caustics and in shadow zones. The second case considered was shallow water propagation with surface waves present. Reflection of sound beneath a wave crest leads to focussing and the formation of caustics and shadow zones. Wavefront modelling gave good agreement with a reference solution in this situation of rapid range dependence. A tank experiment was then performed and wavefront modelling was able to reproduce the detailed waveforms resulting from the interference of energy reflected from different parts of the same crest.

2. SOLUTION OF THE WAVE EQUATION

Wavefront modelling is developed directly from a solution of the wave equation. The main steps will be outlined here. Full details are given in Ref.6.

The appropriate Hankel transform solution of the time independent wave equation expresses the acoustic field as an integral over horizontal wave number with depth dependence contained in solutions of the depth separated wave equation. The next step is to replace the depth functions with their WKB approximations and transform the corresponding Wronskian into a power series as described originally in Ref. 8. The acoustic field can be then be written as a sum over terms of the following form

$$p_n(r,z) = Q(2\pi r)^{-1/2} e^{-i\pi/4} Error! Error! Error! exp(i\phi_n) dk$$ (1)

This term gives the acoustic pressure $p_n(r,z)$ as a function of range $r$ and depth $z$ corresponding to a ray path which has $n$ complete cycles up and down. The subscript $j$ takes 4
values according to whether the ray is going up or down at source and receiver. The parameters are identified as follows: Q is the source strength, k is the horizontal wave number, γs and γr are vertical wave numbers at source and receiver, Ra and Rb are reflection coefficients above and below the source, n_j' and n_j" are the number of corresponding reflections.

The phase \( \phi_{nj} \) is given by

\[
\phi_{nj} = \text{Error!} + n_j'\psi_a + n_j''\psi_b + \kappa r
\]  

(2)

where \( z_s \) and \( z_r \) are source and receiver depths, \( \psi_a \) and \( \psi_b \) are the phases of \( R_a \) and \( R_b \) and the notation \( z_s~ \) means the integral follows the ray path up and down. 'Reflection' also includes rays which turn over by refraction.

The expressions above give the field in a horizontally stratified situation. Range dependence can be incorporated by allowing variation of the horizontal wave number k and replacing \( \kappa r \) with an integral over range. It is then convenient to replace wave numbers with the corresponding functions of angle and the result becomes

\[
p_{nj}(r,z) = Q(2\pi r)^{-1/2} e^{i\pi/4} \text{Error!Error!Error!Error!} \exp(i\phi_{nj} + \delta_0) (d\theta_r/d\theta_s)^{1/2} d\theta_s
\]  

(3)

where \( \omega \) is the angular frequency, \( c_s \) is the sound speed at the source, \( \theta_s \) and \( \theta_r \) are ray angles at source and receiver respectively and the products of reflection coefficients allow for the reflection coefficient to be different at each reflection. The parameter \( \delta_0 \) arises from some square root terms and is given by

\[
\delta_0 = (\pi/4)[1 - \text{signum}(\theta_s, \theta_r)].
\]  

(4)

The phase is now given by

\[
\phi_{nj} = \text{Error!} + \text{Error!} + \text{Error!} + \text{Error!}
\]  

(5)

It is shown in Ref. B that the phase \( \phi_{nj} \) in Eq. (5) is the total phase of a ray which leaves the source at angle \( \theta_s \) and passes the vicinity of the receiver. In general it is not an eigenray and will not pass through the receiver. The phase is that of a plane wavefront through the receiver perpendicular to the ray and is a function of launch angle.

It is readily shown that the travel time \( T_{nj} \) along the ray path to the wave front is given by

\[
T_{nj} = \text{Error!} + \text{Error!}
\]  

(6)

Combining Eqs. (5) and (6) gives

\[
\phi_{nj} = \omega T_{nj} + \text{Error!} + \text{Error!}
\]  

(7)

which shows that the total phase is the sum of the phase due to the travel time plus the phase changes on reflection.

3. PHASE INTEGRALS

Equation (3) has the form of a phase integral. The main contribution to the integral comes from the vicinity of points of stationary phase. It is shown in Ref. 6 that the launch angle for the point of stationary phase is also the launch angle of the eigenray which passes exactly through the receiver. An analytic approximation to the field can be found by replacing the phase by its simplest polynomial approximation and analytically evaluating the resulting canonical integral. For a single point of stationary phase the phase \( \phi_{nj}(\theta_s) \) is approximated by a quadratic and the result can be written

\[
p_{nj} = Qr^{-1/2} \text{Error!} \left[ (\omega/c_s)\cos\theta_s \right]^{1/2} \left[ (\omega/c_r) \cos\theta_r \right]^{-1/2} |dz*/d\theta_s|^{-1/2} \exp[i(\phi_{nj} + \bar{\delta})]
\]  

(8)

where \( \theta_s \) and \( \theta_r \) are parameters for the eigenray and \( z^* \) is the depth of the ray as it passes the
range of the receiver. The extra phase term $\delta$ is given by

$$\delta = \left(\frac{\pi}{4}\right) \left[1 - \text{signum}(\theta_0, d\theta_0, d\theta_0)\right]. \tag{9}$$

The result in Eq. (8) is the same as is obtained from ray geometry and energy conservation.

For cases where there are two nearby points of stationary phase the second derivative of the phase $\phi_{nj}(\theta_0)$ has a zero at $\theta_0$. This corresponds to the field near a caustic where two rays with the same sequence of turning points arrive close together with a phase difference less than $\pi/2$. The phase is approximated by a cubic expansion about $\theta_0$ and the canonical integral leads to an Airy function. The result can be written

$$p_{nj} = Q(2\pi r)^{1/2} e^{i\pi/4} \text{Error!}[(\omega/c_s)\cos \theta_0]^{1/2}$$

$$\times \left|\frac{d\theta_0}{d\theta_s}\right|^{1/2} \beta \exp[i(\phi_0 + \delta_0)] \text{Ai}(-|\phi_0|) \tag{10}$$

where

$$\beta = |\phi_0'''|^{-1/3}. \tag{11}$$

The values $\phi_0$, $\phi_0'$ and $\phi_0''$ are the phase and its derivatives at $\theta_0$.

The field given by Eq. (10) also gives the field both near the caustic and in the shadow zone. The result remains finite and smooth as the receiver passes through the caustic and into the shadow zone. On the caustic the two points of stationary phase coincide and the two eigenrays coincide. On the shadow side of a caustic there are no points of stationary phase and no eigenrays but the integral can still be evaluated analytically by expansion around $\theta_0$ where the second derivative vanishes.

As the receiver moves away from the caustic the two eigenrays can be treated as separated when the phase difference exceeds $\pi/2$. For numerical work a uniform asymptotic expression given in Ref. 5 is used when the phase difference lies between $\pi/50$ and $\pi/2$. The uniform asymptotic expression matches smoothly on to the fields given by Eqs. (8) and (10).

Near a focus there are three nearby points of stationary phase and the third derivative of the phase has a zero. The phase is expanded as a quadratic and the canonical integral leads to a Pearcey function. The result can be written

$$p_{nj} = Q(2\pi r)^{1/2} e^{i\pi/4} \text{Error!}[(\omega/c_s)\cos \theta_0]^{1/2}$$

$$\times \left|\frac{d\theta_0}{d\theta_s}\right|^{1/2} \alpha \exp[i(\phi_0 + \delta_0)] \text{Pc}(\alpha \phi_0', \alpha^2 \phi_0''/2) \tag{12}$$

where

$$\alpha = (\phi_0'''/4)^{-1/4} \tag{13}$$

and the phase derivatives are evaluated where the third derivative vanishes.

The field given by Eq. (12) is used when the three eigenrays arrive with phase differences less than $\pi/2$ and matches smoothly on to the field given by Eqs. (8) and (10) as the eigenrays arrive further apart.

4. DEEP WATER PROPAGATION

Wavefront modelling begins with a ray trace to the range of the receivers. For the present
example we consider a Munk sound speed profile and a range of 80 km from a source at the sound speed minimum at 800 m depth. The ray trace gives travel time $T$ and depth $z^*$ for the range of the receiver as a function of launch angle of the ray. The ray depth $z^*$ at the receiver range is shown in Fig. 1. The launch angle range is $\pm 14^\circ$. Rays at larger angles meet the surface and do not contribute at long ranges.

As rays propagate in deep water they cycle up and down by refraction and the depth $z^*$ at the range of the receiver moves up and down. For the case shown in Fig. 1 $z^*$ has two maxima and two minima and each corresponds to the ray touching a caustic.

![Fig. 1: Depth $z^*$ of a ray at the receiver range as a function of launch angle.](image)

The depth vs time diagram for the above example is shown in Fig. 2a. In deep water rays turn by refraction and the wavefront becomes folded back on itself as illustrated in Fig. 2a. The fold at about 271 m depth corresponds to the caustic at $8.2^\circ$ in Fig. 1. The section of wavefront to the left of the caustic in Fig. 2a arrives at greater depths first and corresponds to upgoing rays. The section to the right of the caustic corresponds to downgoing rays that have turned and touched the caustic. The other two folds in the wavefront correspond to caustics at depths of 417 m and 1293 m and angles of $-5.5^\circ$ and $4.8^\circ$ respectively in Fig. 1.
Fig. 2: (a) Ray depth as a function of arrival time at the receiver range. (b) Waveforms as a function of depth.

The sequence of ray arrivals at any depth is found from the depth vs time diagram of Fig. 2a. The waveform for a receiver at any given depth is calculated by placing pulses of the appropriate amplitude and phase at the appropriate time. Examples of such constructed waveforms are shown in Fig. 2b. Comparison of Figs. 2a and 2b shows how the folded wave front gives rise to a series of pulses at various depths.

The source pulse for Fig. 2 is a two cycle cosine pulse at 75 Hz smoothed by band pass filtering between 37.5 and 112.5 Hz. The first pulse at depths from 600 to 1400 m is an inverted copy of the source pulse because the ray has touched two caustics and has accumulated a phase change of $-\pi$.

5. CAUSTIC AND SHADOW ZONE

Comparison of Figs. 2a and 2b shows that the third pulses in Fig. 2b at 1300 m and 1400 m depth are in the shadow zone of the caustic at 1293 m. For these pulses there is no eigenray but the phase function has a vanishing second derivative near the caustic. As described above the phase function is approximated numerically by a cubic polynomial and the approximate integral is evaluated analytically to give the Airy function field of Eqs. (10).
Fig. 3: Phase variation as a function of launch angle for receivers at the depths shown. The zero of phase is arbitrary and the graphs are offset by 5 radians for clarity. The solid curves are the exact phase. The dashed curves are the cubic approximation.

It is interesting to see how the phase function varies in the vicinity of the caustic and the results are shown in Fig. 3 for receiver depths of 1100, 1200, 1300 and 1400 m and the section of wavefront near 54.1 s in Figs. 2a and 2b. At 1100 m depth the phase function has turning points for source angles of 2.84° and 6.19°. The phase difference is 4.47 radians and so the two stationary points are treated as isolated. The corresponding waveform at 1100 m and 54.1 s in Fig. 2b has two pulses which interfere and largely cancel where they overlap.

At 1200 m the eigenray angles for the two arrivals near 54.1 s are 3.55° and 5.78°. The phase difference is 1.52 rad so the two arrivals are treated as a pair using the Airy function formula of Eq. (10) and the uniform asymptotic expansion of Ref. 5.

At 1300 m and 1400 m the phase function has no point of inflexion and so there are no eigenrays and a receiver at both 1300 m and 1400 m depths would be in the shadow zone. The dashed curves show the cubic approximation to the phase evaluated at the point of inflexion. The Airy function expression of Eq. (10) then gives the phase and amplitude of the field in the shadow zone and leads to the waveforms at 54.1 s in Fig. 2b.

The dashed curves in Fig. 3 show how the approximation to the phase function changes as the receiver depth changes. The dashed curves diverge from the exact phase function quite quickly and are a good approximation only over a short range of source angle values. Nevertheless the analytic approximation to the phase integral gives the amplitude and phase of the field to good accuracy and produces the correct pulses at 54.1 s in the waveforms of Fig. 3.

6. DISCUSSION AND CONCLUSIONS
Asymptotic evaluation of phase integrals assumes that the phase varies rapidly away from the points of stationary phase so that the resulting rapid oscillation of the real and imaginary parts of the integrand give negligible contribution to the integral.

The phase information in Fig. 3 allows us to assess whether the assumptions of asymptotic evaluation are justified and what is meant by low frequency. The depth time diagram of Fig. 2a is obtained purely from a ray trace and is independent of frequency. The phase information in Fig. 3 is obtained by multiplying the ray travel time by the angular frequency and adding any phase changes on reflection as in Eq. (7). Therefore the phase shown in Fig. 3 is proportional to the frequency. At higher frequencies the phases would vary rapidly and the assumptions required for asymptotic evaluation of the phase integrals would be satisfied. The values given in Fig. 3 are for a frequency of 75 Hz and at 1400 m the phase across the angular range shown is only about $3\pi$. The phase change is even smaller for the other depths.

It is clear that the assumption of rapid phase variation is not justified at 75 Hz. However, it was shown in Ref. 5 that waveforms like those of Fig. 2b are accurate and have the correct amplitude and phase. A likely explanation is that the phase integral is actually a contour integral in the complex plane so the asymptotic evaluation should use the method of steepest descents rather than stationary phase.

The small variation of phase across the integrand shows that this is a very low frequency situation. Nevertheless, the calculated waveforms are accurate and show that the ray theory based wave front approach described here is accurate at low frequencies.

Wavefront modelling is still under development. Future work will include the beam displacement which occurs when the reflection coefficient is a function of angle.

REFERENCES

Structured Session 6

Geoacoustic Inversions

Organizers: Ross Chapman & Peter Gerstoft
3D GEOACOUSTIC CHARACTERIZATION OF SEABED SEDIMENT FROM THE MODAL INVERSION EXPERIMENTS

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Abstract: In the summer of 2006, the US Office of Naval Research sponsored the “Shallow Water ’06” (SW06) experiment. This was a multi-discipline, multi-institutional experiment combining studies on the coastal oceanography and ocean acoustics of continental shelf and slope environments. This paper describes the objectives, approach, and results of experiments designed to extract geoaoustic properties of the sediment from acoustic data. The objective was the estimation of sediment properties for an approximately 90 sq. km area of the shelf surrounding a moored array of hydrophones. Both narrowband (NB) and broadband (BB) data, over the frequency band 50-300 Hz, were collected on the central 16 channel vertical line array, as well as on several other moored receivers in the area. Range-dependent wave number estimates were obtained from data measured along several tracks for a towed source broadcasting tones at 50, 75, 125, and 175 Hz. These data were inverted using qualitative regularization in a perturbative inversion scheme to obtain sediment sound speed estimates at discrete ranges along the tracks. The approach then combined these results from perturbation inversion with chirp seismic survey data interpretations to derive a 3D map of sound speed in the sediment. Based on the resulting 3D geoaoustic model, acoustic fields were predicted and compared to measurements made in the same region having different source/receiver geometries. Separately, modal travel time estimates were obtained from BB data and used as input data in an inversion scheme based on travel time perturbations. Both the NB and BB inversion results indicated the existence of a low-speed layer 5 – 10 m deep in the sediment. Depending on frequency, this layer acts as a duct, trapping energy in the sediment layer, the implications of which will be discussed for inversion scheme performance and interpretation of results.

Keywords: inversion, shallow water acoustics
1. INTRODUCTION

In the summer of 2006, the Office of Naval Research sponsored the “Shallow Water '06” (SW06) experiment [1] on the New Jersey shelf area of the North Atlantic. This environment is characterized by both dynamic oceanographic conditions and spatially dependent geomorphology. A goal of SW06 was to gain a better understanding of acoustic propagation in these complex environments. In particular, in shallow water, the ocean bottom plays a significant role in acoustic propagation. Based on this, a multitude of approaches exist for extracting geoacoustic parameters of the bottom from acoustic data [2]. However, often, these approaches are applied to data sets taken at different locations and under different conditions, making it difficult to compare results amongst methods.

In this work, two approaches based on modal input data are applied to data collected on the same vertical line array (VLA) of hydrophones. In the first approach, an acoustic source was towed out and back along three ~5 km tracks oriented along different radials with respect to the VLA. Horizontal wave numbers were obtained for a discrete set of ranges along each track and used as input data to a perturbative inversion algorithm. The resulting ensemble of inversion results for all three tracks was then combined with chirp seismic survey data to obtain a 3D map of sediment sound speed for an area around the VLA [3].

In the second approach, a broadband linear frequency modulated (LFM) signal was broadcast from the J-15-3 source while station keeping at a standoff of ~15 km from the array. By applying time-frequency analysis techniques to the received data on the array, modal travel time differences could be estimated. Estimates of modal travel times obtained from a variety of source/receiver geometries were then used in another form of a perturbative inversion algorithm to estimate sediment sound speeds [4].

In this paper, we present geoacoustic inversion results obtained from the two data types and inversion algorithms. Although the data sets do not cover the exact same geographic region, they do intersect. It is of interest to look at any qualitative agreement or disagreement in the inversion results for these regions. This is a first step in synthesizing results obtained from different inversion approaches obtained for the same area.

2. DISCRIPCION OF EXPERIMENT

The 16-channel VLA spanned the water column between 13.5 and 78 meters depth. For the first approach, a J-15-3 low frequency source, broadcasting continuous tones at 50, 75, 125 and 175 Hz, was towed at constant depth from the R/V Endeavor. Repeated runs along three ~5 km tracks were made at speeds of 2 – 10 knots over 30 hours. The locations of the ship tracks and VLA are shown with bathymetry in Fig. 1. In the second approach the J-15-3 was used to broadcast a 0.5 sec LMF sweep from 40 – 290 Hz. The signal was broadcast from several source locations and acquired on the VLA and on a set of five single hydrophone receiver units (SHRU). The location of two SHRU receivers and source locations for the LFM shots are also shown in Fig. 1. The signaling scheme at each source location consisted of repeated transmission of the LFM signal. The signals were transmitted every 3 seconds with approximately 120 pings transmitted at each station. In addition to the acoustic measurements, detailed properties of the water column were measured via a towed CTD chain at the source and temperature sensors on the VLA.
Fig. 1: Segmented bathymetry map of the New Jersey shelf. Open symbols indicate VLA and SHRU locations, solid lines CW source tow tracks, and dots LFM source positions.

3. MODAL INVERSE APPROACHES

Modal inversion using wave number input data is based on the relationship between a perturbation in the sound speed profile for a given waveguide model and the resulting perturbation in modal wave numbers. Representing parameters and functions of the background model by the subscript \( b \), the perturbation relationship between \( \Delta c(z) \) and \( \Delta k_n(\omega) \) is given by [5]

\[
\Delta k_n(\omega) = \int_0^\infty -\frac{1}{k_n c_b^2(z)} \left| \phi_n(z) \right|^2 dz.
\]

In determining \( \Delta c(z) \) from data, we take as input data \( \Delta k_n(\omega) = k_n - k_{bn} \), the difference between wave numbers observed from measurement and those of the background waveguide environment, respectively. The integral equation (1) is then inverted to find \( \Delta c(z) \).

Inversion based on modal travel time differences can be formulated based on the phase of the far-field acoustic pressure field expressed as a sum of normal modes. In this case, the phase is given by \( \theta_n(r) = k_n r \) which, taking the difference on both sides becomes

\[
\Delta \theta_n(\omega) = \Delta k_n r.
\]

The travel time for mode \( n \) to travel a distance \( r \) is determined by the group velocity given as \( v_{gn} = \frac{\partial \omega}{\partial k_n} \). Using the group velocity and based on the relationship given in Eq. (2), a
relation between a perturbation to the background sound speed profile and the resulting perturbation in modal travel times can be derived [4]

\[
\frac{\partial \psi_n}{r} = \frac{1}{v_n(\omega)} - \frac{1}{\nu_\omega(\omega)} = \frac{\partial}{\partial \omega} \int_0^\infty -\omega^2 \Delta c(z) k_{nn} c_n(z) \rho_n(z) \phi_n(z, \omega)^2 \, dz.
\]

(3)

In this case, the measurement data are observed differences in modal travel times between measurement data and modelled data for a known source/receiver separation distance. Eqs. (1) and (3) have the same kernel and have the form of a Fredholm Integral equation of the first kind. Detailed solutions to these types of equations, and in particular for estimating geoacoustic properties can be found in the literature [6].

Inversion based on Eqs. (1) and (3) can also be applied to the range-dependent waveguide case by a suitable division on the environment into range-independent segments. For each range-independent segment, the input data are estimated and the integral equation inverted. In the case of inversion based on range-dependent wave number estimates, the total solution is then formed as an ensemble of the independent solutions. In this way, a 3D sediment map of an area can be formed.

4. INPUT DATA FOR INVERSIONS

Wave number estimation was accomplished using a sliding window auto regressive estimator [6] applied to the cw data with an aperture of 2000 meters weighted by a Hann window. Wave numbers estimated at each step in range were used in a discretized form of Eq. (1) to yield an ensemble of sediment sound speed inversion results along all three tracks shown in Fig. 1. For the broadband data, modal travel time data were estimated by time-frequency analysis of the data. In this case, because absolute time of the source pings was not available, travel time differences between individual modes was used as input data and compared to the background model [4]. Figure 2a shows a range-wave number plot from which input data for inversion based on modal eigenvalues can be extracted. Figure 2b illustrates the modal arrival time analysis which provides input data for inversion based on travel time differences.

![Fig. 2 (a) Wave number vs. range for one channel cw data on VLA (125 Hz)](image1)

![Fig 2(b) Modal dispersion curve from which modal travel times are estimated](image2)
5. RESULTS

Fig. 3. Compressional wave speed for sediment along R/V Knorr track (right) estimated by combining chirp survey data (left) with cw inversion results from other tracks.

A 3D map of the compressional sound speed in the sediment was determined by combining analysis of the data along the three cw tracks indicated in Fig. 1, with chirp seismic data shown in the left of Fig. 3. For each of the three cw tracks, an ensemble of sound speed profiles was estimated along the length of the tracks. From the cw inversion result, along with interpretation of the seismic data, four distinct geoaoustic provinces were determined. These provinces were defined by a terminating half-space below the 'R – reflector' with a sound speed of 1725 m/s; a slow speed layer that pinches out within the region with a sound speed of 1585 m/s; a layer with a sound speed of 1670 m/s; and a sand layer at the far edge of the region away from the VLA with a sound speed of 1740 m/s. The right hand panel of Fig. 3 shows a depth-range plot of sediment sound speed extracted from the full 3D characterization along the R/V Knorr track indicated in the figure. This track was run in conjunction with another set of acoustic measurements in the same area and represents a towed source path where the signal was measured on the MPL VLA. Using this sediment model, the signal on the MPL VLA could be predicted based on the cw inversion results. For frequencies of 53 and 103 Hz, greater than 90% correlation was obtained with the data recorded at 53 and 103 Hz.

Fig. 4: Compressional wave speed vs depth in the sediment estimated by inversion using modal travel times for the six regions indicated in figure 1
Sound speed profiles in the sediment were also determined using modal travel times as input data in a discretized form of Eq. (3). Results were obtained for the six regions indicated by 41-62 in Fig. 1 by estimating modal travel times for multiple combinations of source/receiver locations shown in the figure. The ranges required in the inversion algorithm were fixed by the source receiver/geometries given the a priori division of the region as indicated. The number of regions into which an environment can be divided is dependent on the number of different source/receiver combinations available for analysis [4]. The resulting sound speed profiles for the six regions shown in Fig. 4 indicate the region to be generally characterized by a region of slower sound speed, or duct, in the sediment. The ducting region varies in layer thickness and relative slowness throughout the region. The general trend is that the duct is thinner and slower near the region of the VLA – region 42. To the East, the duct gets broader and the sound speed near the surface increases – regions 51 and 61. This is consistent with the cw inversion results where a high speed sand layer was indicated at the surface to the East. Relative to the cw inversion results, the broadband results are highly smoothed over the regions of interest. In addition, the depth profiles are smoothed by the use of Tikhonov regularization. The steps observed in the profiles are an artifact of plotting a constant sound speed for each layer. As a consequence, it is not possible to resolve the pinching layer as in the previous case.

6. CONCLUSION

Narrowband and broadband data were collected for use in modal inversion in a region on the NJ shelf. Combining modal wave number inversion with chirp seismic data yielded a high-resolution 3D map of the sediment sound speed for a region near the VLA. Using this model, acoustic data from another experiment was predicted with a high degree of correlation. Inversion results based on modal travel time estimates yielded a lower resolution picture of the sediment sound speed character out to 15 km from the VLA. The results from this analysis lacked the high resolution of the 3D map, but had consistent features, in particular a low speed layer. This is a first step toward synthesizing and validating geoacoustic inversion results for a given shallow water environment.

7. ACKNOWLEDGEMENTS

We’d like to thank the scientific and engineering staff at the Woods Hole Oceanographic Institution for providing data acquisition, both acoustic and oceanographic, to support these experiments. John Goff at UT Austin provided the chirp seismic data and interpretation. Our appreciation is also extended to the crew of R/V Endeavor for towed source and CTD chain operations. M.S. Ballard was supported by a National Defense Science & Engineering Graduate Fellowship award. This work was also supported by an Office of Naval Research Grant No. N00014-08-1-0237.

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GEOACOUSTIC CHARACTERIZATION USING IMAGE SOURCES

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Abstract: In this communication, we present the first results of a new imaging technique that allows us to obtain the geometry and the geoacoustic structure of a seafloor in shallow water areas. For doing this, we use a broadband (100 Hz – 6 kHz) acoustic source towed by a ship and a vertical array moored on the seafloor, composed of 15 hydrophones. All the different acoustic paths are recorded but we only deal with the first reflected signal which is composed by the numerous reflections of the pulse signal on the layers that compose the seafloor. Because the signal is short in the time domain and because we use only short range, the reflected signal can be modeled as a sum of contributions coming from image sources relative to the seafloors layers. The position in space of these image sources being directly connected with the seafloor geometry and the sound-speed profile, the visualization of these image sources allows us to find these characteristics of the seafloor. The map of the image sources is obtained by back-propagating numerically the recorded signals in the domain where the seafloor has been removed. A simple backpropagation of the signals does not give very good results so, we have developed a more complete algorithm which allow us to get a very clean map of the image sources. This inversion method is consequently data driven and does not use any modeling of the propagation inside the sea column or inside the sea bottom. The first results obtained on experimental and synthetic data are very promising.

Keywords: Seafloor characterization, image sources, array processing, geoacoustic inversion.
1. INTRODUCTION

The knowledge of seafloor structure is essential for many applications. Complementary to the direct geophysics measurements, remote sensing by acoustics has proved its ability to get the geometry and the physical parameters of the seafloor. Most of the present techniques are based on an inversion process such as, for example, matched field methods [1] or inversion of backscattering strength data [2]. Recently, C. Holland and J. Osler have proposed a joint time-frequency method based on reflection coefficient measurement at short distance [3]. In their measurements, they used a single hydrophone on a vertical array and a towed omnidirectional broadband source (fig. 1). These data were also used recently to study the dependence of the coherence with the seafloor nature [4]. Based on this geometrical configuration, we propose here a method to invert the seafloor structure (layering and sound-speed profile). The main idea is to use the full array response to get the image sources and then to use the position of these image sources to get the required information. The data used in this paper are presented on section 2. Then, we describe the method with its hypothesis and algorithm. And finally, on section 3, we present results on synthetic and real data.

2. DATA AND MODEL

2.1. Synthetic data

To test our approach, synthetic data are obtained with a numerical model of our problem. This is done with a numerical evaluation of the Sommerfeld integral that is the exact analytical solution of the reflection of a spherical wave on a layered media [5]. The transfer function \( H_n \) of this reflection between a source located at \( r^s_0 = (0, z^s_0) \) and a hydrophone \( n \) located at \( r^r_n = (r^r_n, z^r_n) \) is:

\[
H_n (r^r_n, r^s_0, \omega) = \frac{ik}{\pi s^2 \omega} \int_{0}^{\pi/2} J_0 (kr^r_n \sin \theta) R(\theta, \omega) e^{iH(z^s_0+z^r_n)\cos \theta} \sin \theta d\theta ,
\]

where \( \theta \) is the incidence angle, \( \omega \) the angular frequency, and \( k \) the wave number. The exponents \( s \) et \( r \) stand respectively for a source and a receiver. Because this integral is the result of plane wave decomposition, the term \( R(\theta, \omega) \) is the plane wave reflection coefficient and can be computed for a large variety of structures [6]. The time signal composed of the direct and the reflected wave at the hydrophone \( n \) is:
\( s_s(t) = \text{TF}^{-1}\left[\left(\mathcal{G}_0(r_n^s, r_n^f, \omega) + H_s(r_n^f, r_n^s, \omega \right)) \times F(\omega)\right], \)

where \( F(\omega) \) is the source spectrum (fig.2b) and \( G_0(r_1, r_2, \omega) = \exp(ik_0|\mathbf{r}_1 - \mathbf{r}_2|/|\mathbf{r}_1 - \mathbf{r}_2|) \) is the Green function of an homogeneous medium with a wave number \( k_0 = \omega/c_0 \), \( c_0 \) being the sound speed. Note that the signal used for the synthetic data comes from the real data experiments (see next subsection).

**Fig. 2:** Signal emitted by the source (a) in the time domain and (b) in the frequency domain.

The choice of the geoacoustic structure for the synthetic data is driven by two opposite principles: on one hand, it should be complex enough to prove the validity of the method, but on the other hand, it should be simple enough to avoid difficult interpretation. We decide to simulate a seafloor composed of 9 sediment fluid layers covering a semi-infinite fluid basement (Tab. 1). The synthetic data are then obtained with equations 1 and 2, \( R(\theta, f) \) being computed with these parameters.

<table>
<thead>
<tr>
<th>Layer</th>
<th>Sound speed (m/s)</th>
<th>Density</th>
<th>Thickness (m)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Water</td>
<td>1500</td>
<td>1</td>
<td>-</td>
</tr>
<tr>
<td>1</td>
<td>1520</td>
<td>1.1</td>
<td>3</td>
</tr>
<tr>
<td>2</td>
<td>1540</td>
<td>1.2</td>
<td>3.5</td>
</tr>
<tr>
<td>3</td>
<td>1600</td>
<td>1.5</td>
<td>4</td>
</tr>
<tr>
<td>4</td>
<td>1630</td>
<td>1.7</td>
<td>2</td>
</tr>
<tr>
<td>5</td>
<td>1700</td>
<td>1.9</td>
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</tr>
<tr>
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<td>1720</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>7</td>
<td>1800</td>
<td>2.5</td>
<td>5</td>
</tr>
<tr>
<td>8</td>
<td>1900</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>9</td>
<td>1920</td>
<td>3.1</td>
<td>2</td>
</tr>
<tr>
<td>Basement</td>
<td>2000</td>
<td>3.6</td>
<td>-</td>
</tr>
</tbody>
</table>

**Tab. 1:** Geoacoustic parameters for the synthetic data. The media used here are non-dissipative.

**2.2. Data from Scarab experiments**

The method developed here is also compared to real data acquired in June near Elba Island in Italy (Fig. 3) as part of the SCARAB (Scattering And Reverberation from the sea Bottom) experiment series (see [3] for details).
We use the acoustic data obtained on site 2. This site is in 150 m water depth, with a flat and featureless seabed (from side-scan sonar data); bottom slopes are less than about 0.3°. Geoacoustic inversions from broadband reflection data [3] show sound speeds and densities consistent with a silty-clay fabric with intercalating sandy sediments (Fig. 3 right).

3. DETECTION OF IMAGE SOURCES

3.1. Image sources

The image source is a well known method to simulate wave propagation. It models the reflection of a wave on a plane as a wave emitted by the symmetric of the source relative to the surface: the image source. This method is generally used for room acoustics, propagation in a waveguide [3, 7] or to model the reflection on the floor of an emission from a radar antenna [8]. In these two last cases, the image sources are complex to take into account the angular variations of the reflection coefficient.

This method is usually used to model systems, but according to our knowledge, not for making inversion. The spatial positions of the image sources being directly linked to the thicknesses and the sound speeds of the layers that compose the seafloor, the localization of these image sources should allow us to obtain these parameters.

In our case, some hypotheses are necessary to model the reflection of the emitted wave as a collection of image sources:

- the water column and the geologic layers are homogeneous, the latter being all horizontal,
- the angle of incidence over an interface is smaller than the critical angle, and its angular variation (measured on the array for a given source-array distance) is small enough to neglect its influence on the reflection coefficient,
- only the first reflections are taking into account; multiple reflection between interfaces are considered as speckle.

In this case, each reflection of sound on an interface (fig. 4a) is identified by the receiver array as an image source which can be described in an equivalent system: the structure (water + sediment layers) above this interface and its symmetric (fig. 4b). So, we have a different equivalent system for each image. For an equivalent system, the places of the components (water and layer and their symmetric) have no consequences on the angle of arrival or on the
total travel time. It is then possible to merge all the equivalent systems in a single one which contains all the image sources (fig. 4c). In this system, all thicknesses are doubled and the image locations are on the interfaces. In the following we will only consider this system.

3.2. Detection principles

The geoacoustic inversion by the image sources method needs a very accurate localization of them. To solve this problem, we propose here to backpropagate numerically the recorded signals in the water without the sediment structure. This can be done in our case because we know when the pulse is emitted ($t_0$) and because the time resolution of signal is good enough to separate each reflection (Fig. 2a).

The imaging with backpropagation of signals is very close to the time reversal in a homogeneous media [9-10] but signals reflected by the seafloor will not focus only on the real source after backpropagation like in the physical time reversal, but will also focus on all image sources corresponding to the reflections on geological interfaces because we suppressed the sediment structure. This phenomenon has already been shown during time reversal experiment [11] but here, we use it to characterize the seafloor. Source and images are coherent because they emit the same signal (with different amplitudes) at the time $t_0$.

For $M+1$ monopole sources (the real one + $M$ images) emitting at the same time a short pulse with the spectrum $F(\omega)$ and with a different amplitude factor $\beta^m$ at $\mathbf{r}^s_m$ in an homogeneous and isotropic media, the received signal at the hydrophone $n$ at $\mathbf{r}^r_n$ is:

$$S_n(\omega) = \sum_{m=0}^{M} \beta^m F(\omega) \times G_0(\mathbf{r}^s_m, \mathbf{r}^r_n, \omega) + \eta_n(\omega) ,$$

where $\eta_n(\omega)$ is an additive noise, spatially white. In the following, the signal to noise ratio is supposed high.
To backpropagate signals at a coordinate $r$ where we want to know if there is an image source, we multiply the spectrum $S_n(\omega)$ by the inverse of the Green function $G^{-1}_0(r_1, r_2, \omega) = \left|r_1 - r_2\right| \exp(-ik_0|r_1 - r_2|)$ and then we compute the average over the $N$ hydrophones of the array. In order to reduce on a signal the influence of all sources and the noise, we propose to window the backpropagated signals around $t_0$ with a window $w(t)$ that have quite the same duration than the emitted pulse: $S_{wn}(r, \omega) = \left[S_n(\omega) \times G^{-1}_0(r, r', \omega)\right] * TF[w(t)]$. Thus, the energy that is mapped is:

$$F^{BW}(r) = \begin{bmatrix} \frac{1}{N} \sum_{n=1}^{N} S_{wn}(r, \omega) \end{bmatrix}^2 d\omega.$$  (4)

But even with the window, there is still speckle on $F^{BW}(r)$. Indeed, the pulse emitted by the source $m$ and received by the sensor $n$ draw on the map a circle centred on the sensor with a radius of $|r_m' - r_n'|$ and the intersection of the circles of the $N$ sensors will correspond to the source location. The problem is that it is possible for some isolated circles to be greater than a coherent intersection of circles corresponding to another source with lower amplitude. This problem is well visible on figures 5a and 5c (next section) where the amplitude of an image source is lower than the circles from the real source. In that way, it will be difficult to make an automatic detection of the sources. However, to detect sources, we are not interested in their amplitudes. Consequently, we can normalize the backpropagated and windowed signals and work in the orthogonal subspace of the signal by searching the minima of the functional:

$$F^{NS}(r, \omega) = N - \sum_{n=1}^{N} \sum_{q=1}^{N} \frac{S_{wn}(r, \omega) \times S^*_w(r, \omega)}{\|S_{wn}(r, \omega)\|_{2,N} \times \|S_{wn}(r, \omega)\|_{2,N}},$$  (5)

where $S_{wn}(r, \omega)$ is the column vector formed by the signals $S_{wn}(r, \omega)$ and $\| \cdot \|_{2,N}$ is the L2 norm of the signal vector.

The range resolution of $F^{NS}$ is directly linked to the window size $w(t)$. For making more accurate this range resolution, we use the knowledge of the emitted pulse $F(\omega)$ by comparing it with the mean of the backpropagated and windowed signals. For this operation, the functional to minimise is:

$$F^{AT}(r, \omega) = \begin{bmatrix} \sum_{n=1}^{N} S_{wn}(r, \omega) \end{bmatrix}^2 \|f_w(t)\|_{\infty,j},$$  (6)

with $f_w(t) = f(t) \times w(t)$, $F_w(\omega) = TF[f_w(t)]$, $S_{wn}(r, t) = TF^{-1}[S_{wn}(r, \omega)]$, and $\| \cdot \|_{\infty,j}$ is the infinite norm of the signal over the time. The choice of the minimum between addition and substraction is due to the fact that the reflection coefficient might be positive or negative.

Finally, the image sources are mapped with the functional:

$$F^{NSAT}(r) = 1 \int_{\omega} \left| F_w(\omega) \right|^2 F^{NS}(r, \omega) + F^{AT}(r, \omega) d\omega,$$  (7)

where $F^{NS}(r, \omega)$ is compensated in frequency with the power spectrum of the emitted pulse.

4. RESULTS
The functionals $I_{BW}(r)$ (equ. 4) and $I_{NSAT}(r)$ (equ. 7) are first computed for the synthetic data and the results are displayed on Fig. 5a and 5b. The map obtained with $I_{BW}(r)$ clearly shows the image source but also all the circles drawn by all the individual hydrophones as mentioned in section 3.2. The functional $I_{NSAT}(r)$ get rid of all of these circles and the image sources appears very clearly and with a very good resolution both in range and angle.

![Fig. 5: Focalisation on the image sources for the synthetic model (a: equ. 4 and b: equ.7) and for the Scarab data (c: equ. 4 and d: equ. 7).](image)

The same results are observed on Scarab data (Fig. 5c and 5d). The image obtained with $I_{BW}(r)$ shows even more artefacts. This might be due to the presence of another sound source (maybe a merchant ship) near the experiment area.

Compared to synthetic data, less image sources appear on the map although the geoaoustic structure of the seafloor in Site 2 is made of 20 layers with a total thickness of 150 m [3]. But some of the layers are very thin and their source images are merged in a single spot with image sources of the interfaces near them.

We can note on the map (Fig. 5b and 5d) that the image sources are not aligned on a perfect straight line; this is due to the refraction in the sediment layer (cf. Fig. 4b and 4c) that is not taken into account in the detection algorithm developed here and presented in the previous section.

5. DISCUSSION
The method presented here and its first results obtained on synthetic and real data are very promising. The hypotheses are not very restrictive and, therefore, the method can be applied to various data.

The method and results presented here are the first steps towards a complete inversion method. The next step is to use the obtained localization of the image sources to get information on the geoacoustic structure. We believe that the geometry (number of layers and their thicknesses) and the sound-speed profiles are quite easy to obtain from the presented results, e.g. from the map presented on figure 5b and 5d. Very first results confirm this impression and this will be our first work to develop in the future.

The main difference with existing method is that the inversion scheme presented here is not based on comparison with a model. It is, more or less, a data-driven approach. Consequently, we think that this procedure will be robust and quite fast. The presented results being the first ones, this work must of course be confirmed and extended to other data.

REFERENCES

PROPAGATION AND INVERSION OF AIRGUN SIGNALS IN SHALLOW WATER OVER A LIMESTONE SEABED

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\textbf{Abstract:} Limestone seabeds with thin or non-existent coverings of unconsolidated sediment are common around the southern Australian continental shelf and often provide strong coupling between the sound wave in the water and the shear wave in the seabed. Sound reflection from such seabeds is very weak except at the p-wave critical angle, which results in the acoustic energy transmitted to long range in the water column being dominated by high-speed Head waves. The characteristics of acoustic propagation in such an environment have considerable practical importance for the propagation of the sound produced by marine seismic surveys and are investigated in this paper, which compares measured signals from a commercial seismic survey with the results of numerical modelling. Techniques for inverting for the geoacoustic parameters of the seabed are also considered.

\textbf{Keywords:} Acoustic, propagation, limestone, calcarenite, geoacoustic inversion
1. INTRODUCTION

Most of the southern and western continental shelf of Australia is capped by calcarenite, a relatively soft type of limestone formed during past sea level low-stands when unconsolidated sediments with a high proportion of calcium carbonate were exposed to fresh water [1]. The fresh water partially dissolved the calcium carbonate, which then re-solidified, bonding the sediment grains together. There are few rivers of any significance along this coastline and those that do exist bring very little sediment to the ocean. The result is that over the majority of the shelf the calcarenite is covered by only a thin veneer (typically less than 1m) of unconsolidated sediment of mainly marine origin.

Calcarenite is a highly variable material, but the geoacoustic properties given in Table 1 appear to be typical [2].

The plane wave reflection coefficients of calcarenite seabeds covered by various thicknesses of sand are plotted in Fig. 1. Of particular note is the very rapid reduction in reflection coefficient with increasing angle that occurs for small grazing angles when there is no sand cover. This is due to the conversion of incident acoustic energy into shear waves in the calcarenite. This dip is progressively filled in as the thickness of the sand layer increases.

<table>
<thead>
<tr>
<th>Material</th>
<th>Calcarenite</th>
<th>Sand</th>
</tr>
</thead>
<tbody>
<tr>
<td>Density (kg.m⁻³)</td>
<td>2400</td>
<td>1800</td>
</tr>
<tr>
<td>Compressional wave speed (m.s⁻¹)</td>
<td>2800</td>
<td>1700</td>
</tr>
<tr>
<td>Compressional wave attenuation (dB/wavelength)</td>
<td>0.1</td>
<td>0.8</td>
</tr>
<tr>
<td>Shear wave speed (m.s⁻¹)</td>
<td>1400</td>
<td>-</td>
</tr>
<tr>
<td>Shear wave attenuation (dB/wavelength)</td>
<td>0.2</td>
<td>-</td>
</tr>
</tbody>
</table>

*Table 1: Geo-acoustic parameters used for reflection coefficient calculation.*

Fig. 1. Magnitude of plane wave reflection coefficient vs. grazing angle for seabeds comprising a calcarenite halfspace covered by sand of thickness 0λ (thick line), 0.1λ (dotted line), 0.2λ (broken line), 0.5λ (dash-dot line), and ∞ (thin solid line). λ is compressional wave wavelength in the sand layer, geoacoustic parameters are given in Table 1.

Another important feature of this plot is the sharp peak at 57°, which corresponds to the p-wave critical angle at the calcarenite interface. Sound incident at grazing angles fractionally
less than this will propagate along the interface at the calcarenite compressional wave speed, re-radiating into the water column. Such waves are called Head or lateral waves and are discussed in detail in [3].

The following sections present a comparison between measured data and modelling results for acoustic signals recorded in a shallow water environment with a seabed of this type.

2. COMPARISON BETWEEN MEASURED DATA AND MODEL OUTPUT

The measured data presented here were recorded during a commercial two-dimensional seismic survey that was carried out off Dongara, Western Australia, centred on 29°10'S, 114°45'E, in water depths ranging from 10m to 45m. The receiving system was a bottom mounted autonomous acoustic recording system that was left in-situ for the duration of the survey (12 days). A total of 27478 airgun array signals (shots) were recorded by this receiver during the survey, but only shots close to the 40m bathymetry contour have been included in the analysis to facilitate using the range independent wavenumber integration propagation model SCOOTER [4] for the comparison. The source depth was 4m and a total of 15001 shots were analysed with source -receiver separations varying from 1 km to 16 km. The Centre for Marine Science and Technology's airgun array model was used to obtained the source spectrum of the array in the direction of the receiver, and this was combined with the narrowband spectrum of the received signal to obtain the transmission loss as a function of range and frequency.

An initial Head wave arrival time analysis [5] was carried out in order to obtain a starting point for a geoacoustic model for the seabed. The Head waves from the higher speed layers were well defined, but the identification of a Head wave from the sediment layer was uncertain. The resulting geoacoustic model is given in Table 2. Compressional wave speeds and layer thicknesses are from Head wave analysis, shear speeds were taken as 50% of compressional wave speeds for elastic layers. Other parameters are typical values for sediments with similar compressional wave speeds. SCOOTER was then run to obtain transmission loss as a function of frequency and range to compare with the modelled data.

A comparison between measured and modelled results is shown in Fig. 2. At frequencies below 120 Hz there is a pronounced horizontal banding in the measured data, which is even more pronounced in the model results. This is similar to the effect expected from shearwave resonances in an upper sediment layer [6], however, the presence of the banding in the modelled data, which does not include an upper sediment layer, indicates that this is not the explanation in this case.

The model results capture the general characteristics of the measured data very well, including the broad maximum in the transmission loss that occurs at all ranges at just over 100 Hz, and the wedge shaped region of much lower transmission loss at higher frequencies, with its characteristic modal interference patterns.

An expanded view of the low frequency portion of the plot is given in Fig. 3. This shows good agreement between the frequencies of the lowest frequency horizontal bands, but progressively poorer agreement at higher frequencies. SCOOTER also predicts sharper, more distinct bands extending out to longer range than those seen in the data.
<table>
<thead>
<tr>
<th>Layer</th>
<th>Water column</th>
<th>Calcarenite</th>
<th>Limestone basement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Thickness (m)</td>
<td>42.5</td>
<td>448</td>
<td>∞</td>
</tr>
<tr>
<td>Density (kg.m⁻³)</td>
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<td>2400</td>
</tr>
<tr>
<td>Compressional wave speed (m.s⁻¹)</td>
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<td>2426</td>
<td>3550</td>
</tr>
<tr>
<td>Compressional wave attenuation (dB/wavelength)</td>
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<td>0.15</td>
<td>0.15</td>
</tr>
<tr>
<td>Shear wave speed (m.s⁻¹)</td>
<td>-</td>
<td>1213</td>
<td>1770</td>
</tr>
<tr>
<td>Shear wave attenuation (dB/wavelength)</td>
<td>-</td>
<td>0.3</td>
<td>0.3</td>
</tr>
</tbody>
</table>

Table 2: Geoacoustic parameters used for model comparison with measured data.

Fig. 2. Transmission loss in dB as a function of range and frequency from measured data (left) and propagation modelling (right).

Fig 4. Expanded view of low frequency portion of Fig. 2.
3. DISCUSSION

The striking features of the transmission loss plots shown in the previous section can be explained in terms of the calcarenite halfspace reflection coefficient plot given in Fig. 1. The horizontal bands occur at the frequencies where the in-water modes have grazing angles at the seabed that correspond to the sharp spike in the reflection coefficient at the compressional wave critical angle. The lowest frequency band occurs when mode 1 satisfies this condition, the second band when mode 2 satisfies it, etc. At these frequencies there is reinforcement between the in-water modes and the Head waves, and strong Head wave arrivals are observed.

As frequency is increased, the seabed grazing angle for a given mode reduces, so the mode can be thought of as traversing the reflection coefficient curve in Fig. 1 from right to left.

At low frequencies all the modes have substantial grazing angles and, because of the sharp dip in the reflection coefficient are strongly attenuated unless one happens to be at the compressional wave critical angle. At frequencies above about 120 Hz the grazing angle of the lowest order mode has reduced to the point where its reflection coefficient is high enough to allow it to make a noticeable contribution to the received field. As the frequency is increased further, the mode 1 reflection coefficient continues to increase, as do the reflection coefficients of the higher order modes so that they also start to contribute significantly to the received signal, giving rise to the modal interference patterns seen above about 170 Hz.

The reasons for the differences between model results and data for the frequencies, spectral width and strengths of the horizontal bands require further investigation, but could include variations in water depth and geoacoustic parameters with range.

3.1. Implications for geoacoustic inversion

The very different transmission loss regimes evidenced by this type of seabed present both challenges and opportunities for geoacoustic inversion. The prominent low frequency banding potentially provides useful information to aid the inversion process, but because of its narrowband nature, would be very easy to miss if inversion was carried out on the basis of a small number of discrete, pre-determined frequencies. For a simple calcarenite halfspace seabed, the frequencies of the horizontal bands correspond to the modal cut-off frequencies and can therefore be used to directly calculate the compressional wave speed in calcarenite. However, when this method was applied to the data presented here it was only found possible to match the frequencies of a few of the bands (Fig. 3). Further work is required to generalise this method to more complicated seabeds.

At higher frequencies more traditional geoacoustic inversion methods based on a comparison between modelled and measured transmission loss at a subset of frequencies should be effective and would be expected to be sensitive to at least some of the properties of the upper sediment layer.

4. CONCLUSIONS

The propagation of sound in shallow water over calcarenite seabeds typical of much of the southern and western sections of Australia's continental shelf displays two distinct frequency regimes. Low frequency propagation occurs in a number of narrow frequency bands
determined by the requirement that the grazing angle of a mode at the seabed corresponds to the calcarenite compressional wave critical angle. Much of this energy propagates through the seabed as Head waves. Through-water propagation corresponding to modes with much smaller grazing angles becomes more important at higher frequencies. These different regimes can potentially be exploited to aid the process of geoacoustic inversion, but this requires further work.

5. ACKNOWLEDGEMENTS

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Measurements of attenuation of low frequency sound in marine sediment

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Abstract: This paper describes a method of determining the compressional wave attenuation in marine sediment from short range measurements. The data were obtained during the Shallow Water 2006 (SW06) experiments at a site on the outer shelf break off the New Jersey coast. The experiment involved a vertical line array that received mid frequency chirp signals from a sound source at a range of 230 m. The frequency band of the chirp was 1.5 to 4.5 kHz. The close range geometry provided strong signals for the single bottom and sub-bottom paths. The sediment attenuation was extracted from the signal strength ratio of the sea bottom reflection to the sub-bottom reflection from the R-reflector at different frequencies from 1.75 kHz to 3.15 kHz. The analysis indicated a linear frequency dependence of the attenuation and weak sound speed dispersion, consistent with the assumption of linear frequency dependence of the attenuation. The estimated attenuation is lower than the values estimated previously from inversions of acoustic field data previously done in the vicinity. Reasons for the lower values are discussed, and the estimated values are compared to predictions from theoretical models for sound propagation in marine sediments. (Work Supported by ONR).
Structured Session 7

Asset Protection by use of Sonar

Organizer: Mathieu Colin
ADVANCED MOSAIC TECHNIQUES OF ACOUSTIC VIDEO IMAGES FOR UNDERWATER SURVEILLANCE AND DIAGNOSING DEGRADATION LEVELS OF HARBOR STRUCTURES

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Abstract: Using an acoustic video camera DIDSON, we took acoustic video images of a ship’s hull and an entire wharf of several hundred meter wide, and then created seamless mosaic images with 0.5cm and 1cm accuracies by correcting the distortion and adjusting the brightness of the video images. This method is effective in detecting suspicious objects around wharfs, ship’s hulls and underwater as well as in diagnosing the degradation levels of harbor structures. Since the video image captured by the DIDSON normally is brightest in the center and gets darker toward the edges, we made corrections to make the brightness and contrast uniform over the mosaic image. In combination with a GPS, a photonic inertial navigation system PHINS and an echo sounder, we estimated the position, view angle, and distance of the DIDSON to the wharf wall. Consequently we could make a precise seamless mosaic image of the wharf. In addition, an image navigation technique was developed for compensating blurring and saw-tooth distortion images, and estimating the camera position. Because of the working principle of the DIDSON acoustic camera, saw-tooth distortions often appear while the camera moves.

Installed on a patrol ship, the SeaBat7123 effectively observes underwater areas with as large a swath width as 300 meters. Using the same method mentioned above, we also created a seamless mosaic image of a sea bottom with 5cm resolution from hundreds of frames obtained with the SeaBat7123. It was possible to mosaic even the detailed images of sunken cars and wrecked ships at the sea bottom. In addition, we also eliminated effectively cross-
talk noises. For more convenient use, we also developed the method of overlaying mosaic images on aerial photo maps and 3-D model images.

**Keywords:** Acoustic video camera, Seamless mosaic image, Underwater inspection

1. **IMAGE NAVIGATION WITH DIDSON**

The research team of UT and Japan Coast Guard Academy for developing underwater security sonar system took acoustic videos of a ship’s hull continuously while a small ship, on which were mounted the StarFire GPS (measurement accuracy of nominally 10 centimeters) and an acoustic video camera DIDSON, was circling around the patrol ship. The patrol ship was anchored; however, the propeller was redundantly caught in the mosaic image using the GPS position due to the drift and motion of camera’s pole, and problems were encountered in the observation of ship’s hull. However, if a mosaic image is created by estimating the movement vector from the matching analysis of two continuous video images and obtaining the ship track of a camera by integrating the movement vector, warping of the propeller is eliminated and a clean mosaic image for observation of the ship bottom is created. [1] For security purposes, the authors researched image matching for correction of the camera shake and developed an image navigation technique based on image matching for the inspection of ship’s hull. This has the effect of sharpening moving objects, because of the focusing of multiple video images. If the position of the acoustic video camera is measured by GPS and the acoustic video image is overlaid on the sea bottom terrain model and an aerial photographic map, it is very effective in observation of the harbor and coast.

In addition, if air is melted in water up to an almost saturated condition, air bubbles are generated near the transducer in the lens frame and a cloudy phenomenon is generated in the video image. It is necessary to avoid this phenomenon. The sharpening of divers amidst bubbles, the removal of crosstalk, the removal of sea surface wave noise, the removal of crack video image phenomena and the removal of multiple ghost video images are the methods required for obtaining clear acoustic video images.

2. ** MOSAIC OF SHIP’S HULL BY IMAGE NAVIGATION**

Fig. 1 shows a frame image of acoustic mosaicking videos of a ship bottom continuously shot while a small ship, on which were mounted the StarFire GPS (measurement accuracy of nominally 10cm) and the acoustic video camera DIDSON, was circling around the patrol ship. The patrol ship was anchored; however, the propellers were redundantly caught in the mosaic image using the GPS position due to the drift and motion of camera’s pole, and several problems were encountered in the observation of ship bottom. If a mosaic image is created by estimating the movement vector from the matching analysis of two continuous video images, the warping of the propeller could be eliminated and a clean mosaic image for observation of the ship bottom could be created. [2] For security purposes, we researched a correction technique of the camera shake and an image navigation technique based on image matching for the inspection of ship’s hull. These have an effect of sharpening moving objects, because of focusing multiple video images. In addition, we measured the position of the acoustic video camera by GPS. As a result, we could overlaid real-time acoustic images on
the sea bottom features model and aerial photographic map. This technique is very effective in underwater observation of the harbor and coast.

Fig. 1: (left) Automatic mosaic by GPS position from the center of the ship bottom to the screw at the stern, (right) mosaic of ship bottom by image matching navigation [2]

Where air is melted in water up to an almost saturated condition, air bubbles are generated near the transducer in the lens frame with the result that a cloudy phenomenon comes out in the video image. Furthermore, air bubbles have gave birth of strong reflections and put divers out of sight. The sharpening of divers amidst bubbles, the removal of crosstalk (refer to Fig. 2), the removal of sea surface wave noise, the removal of crack video image phenomena and the removal of multiple ghost video images are methods required for obtaining clear acoustic video images.

Fig. 2: Diver sharpened by crosstalk removal filter; (left) original video image, (right) processed image
3. DEGRADATION OF HARBOUR STRUCTURE

A collaborative research team of UT and Civil Engineering Research Institute for Cold Region (CERI-CR) implemented the test for diagnosing the degradation and damage status of the quay side of the harbor, and accomplished the creation of precision mosaic video images of the quay side in Otaru port. The research team created detailed seamless mosaic images of quay sides 150 meters wide by 11 meters high and 265 meters wide by 8.5 meters high with 1 centimeter resolution by image matching analysis from approximately 8,000 images taken with the DIDSON. This technology is effective for degradation diagnoses of harbor structures and for examining whether any suspicious objects are attached to the quay side. Grooves of 3 centimeters, 2 centimeters, and 1 centimeter in width are engraved on the two 50 centimeters square concrete plates hung from the quay side, and grooves up to 2 centimeters could be checked from an image shot 5 meters away.

Fig.3: Seamless mosaic image of (a) a wharf 265-meter wide and 8.5-meter high in Otaru port with 1 centimeter resolution with a DIDSON after (b)(c) correction of saw-tooth distortion, (d) adjusting brightness and automatic mosaic, and (e) semi-automatic correction in horizontal position and brightness. A concrete plate 0.5 meters square lowered at a depth interval of 1.5 meters.

This mosaic image is created by estimating the velocity vector from the shot images of the previous or next frame to create a seamless image without position information from a GPS.
This velocity vector is also used for correcting warps in the DIDSON original image. As the wobbling of the support pole becomes larger and the warping of the image increases in the deep sections, this correction technique takes particular effect. In addition to the geometric correction of the image warping caused by the difference in distance of each part of the photogenic subject, the authors implemented brightness and contrast correction in order to homogenize the mosaic image, because of the tendency for the center of an image to become bright while the periphery becomes dark. And for 1 frame of DIDSON image, which divides all 96 beams into 12 sections and created the image with 8 pings to reduce the side-lobe crosstalk and create a clear video image. Consequently, if a moving object is shot or the camera is moved, a saw-tooth distortion appears in the video image as shown in Fig.3 (c). In order to correct this strain, the relative speed of the photogenic subject and camera is estimated and is used for correcting distortions. In some stages of the processing, the authors implemented elastic correction using GPS.

Fig.3 (e) shows a part of seamless mosaic image of a wharf 265-meter wide by 8.5-meter high at Otaru port with 1 centimeter resolution with the DIDSON after correction of saw-tooth distortion, adjusting brightness and automatic mosaic, and semi-automatic correction in horizontal position and brightness had been carried out in turn.

4. MOSAIC OF SEABAT7123

As a development in observational method using ship-based acoustic sonar SeaBat7123, [2] the research team of UT and JCGA has developed an effective moving observational method in which far distance observation to near distance observation can be implemented with one unit utilizing the characteristics of 3 frequencies of ship-based acoustic sonar. The research team also connected the detection sensors of precision GPS positioning and Photonic Inertial Navigation System (PHINS) to create a high-quality video image that is projected on the map in real time, and created a practical and integrated monitoring system through the development of software for the removal of noise caused by movement and tested for operational use in actual sea tests. The research team ran an observation test using multiple sonars to develop a practical monitoring system for use in harbor security at Kobe port, Yokoham port, and Tomakomai port.

The SeaBat7123 was equipped on the hydrographic survey ship "HAMASHIO" of the Japan Coast Guard as shown in Fig. 4 (a). When the ship sails in the sea, the sonar attached to the guide rail on the nose of the ship is lifted above the sea surface, and during observation, the sonar is lowered 1m below the water surface. The pan-tilt device is attached to the upper part of the sonar, so the pan control of ±90 degrees right and left and the tilt swing 0 to 90 degrees from horizontal to vertically downward can be implemented during sonar observation.

The Seabat7123 is a superior equipment that searches for suspicious objects in low-frequency long-range mode and can recognize what an object is with detailed acoustic video image in high-frequency short-range observation mode using three transducers and shared hydrophone arrays that support frequencies of next to 100 kilohertz, 200 kilohertz, and 400 kilohertz.

The Seabat7123 has superior beam resolution and acoustic imaging information; however, it is affected by the crosstalk noise particular to beam forming as shown in Fig.4 (b). Of course, the side lobe suppression is applied to; however, the crosstalk removal process is effective for places that have no echo, such as shade or a specular reflection. For example, when the video images captured sunken ships in the 100 meters range, the crosstalk noise appeared on the whole screen.
Fig. 4: (a) Ship-based 3-frequency observation sonar SeaBat7123 (arrow), equipped on the hydrographic survey ship "HAMASHIO" of the Japan Coast Guard. (b) Underwater acoustic image of sunken ships polluted by crosstalk noises.

Fig. 5: Seamless mosaic image in Yokohama port from SeaBat7123 after crosstalk removal, map-projection, and mosaic.

The authors mosaicked video images of many intentionally sunken ships (length: 25 meters, width: 6 meters) placed in front of the Yokohama hanging pier as fishing banks. Fig. 5 shows a finely mosaicked image in Yokohama port from SeaBat7123 after crosstalk removal, map-projection, and mosaic had been implemented.
In general, a side scan sonar can create an acoustic image of the sea bottom with two fan beams on the right and left; however, this sonar provides an acoustic video image of a large underwater area, utilizing 256 fan beams at once. This allows it to make a significant contribution in real-time observation and searches, and assisting divers and ROVs in the field. It enables effective placement of marker buoys near targets sea bottom location, and allows observation of objects that move in the water such as divers. The combination use of screen scale also enables checking of size, shape, and status of submerged objects. Comparing the real-time video images, the large range in status from the observing video image tessellated on the marine chart can be observed effectively.

In the tests at Kobe and Tomakomai ports, the Seabat7123 surveillance system easily found several vehicles submerged in the water near the quay side in Fig. 6. It displays the terrain data of Tomakomai port surveyed by SeaBat8125; then, the terrain model could check rises at the vehicle positions by comparing the sonar images. As the resolution of sonar video image is higher than that of the terrain model, the sonar video image proved to be effective for the identification of submerged objects, such as vehicles. At Kobe port, it could be attained to orient the acoustic video camera DIDS on to the target effectively by using the SeaBat7123, and the acoustic video image could check the tires and frames of vehicles that were upside down. It is difficult to effectively search vehicles at the sea bottom using acoustic video camera by itself.

![Fig.6: Two vehicles (arrows) sunken into the sea floor in Tomakomai port, easily founded by SeaBat7123 surveillance system. The surveillance system provides with 3-D navigation window of a real-time acoustic image projected on an aerial photographic map and seafloor bathymetric grids.](image)

5. CONCLUSIONS
In order to put acoustic video image technology into practical use, the research team of UT and JCGA have tried to develop a system that can be effectively used for various applications thorough various actual operation tests. The mosaic method and map projection system that use the acoustic video imaging machine are necessary methods for comprehending the whole, by creating clear acoustic video image. Tessellating the clear video image without warping and projecting the video image with the original deformations onto a map or flat surface exert an adverse influence on each other. However, the creation of detailed mosaic video images without warping has been accomplished by utilizing mutual advantages. The authors would also like to attempt in the future many more improvements to the clarity of images, such as the further removal of crosstalk, the removal of background during movement, the acquisition of targets by utilizing the characteristics of multifrequency sonar, and the development of a display method in which the viewing angle can be selected. Acoustic video imaging machines can be used in different fields of applied research, so observation methods and analytic methods suited to particular fields are required.

REFERENCES


Sonar Performance Variability in Shallow Harbours

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Abstract: Sonar is one of the best tools for detecting a variety of underwater threats to ships and man-made structures in shallow harbours. As part of a Maritime Force Protection project, a number of commercial sonar systems were evaluated in order to inform the Canadian Navy on intruder detection capability. Testing was done in local harbour waters in Halifax, Canada, and other sites. It was observed during the evaluations that sonar performance is strongly affected by both natural bathymetric features and by the presence, and type, of man-made structures nearby. This paper describes modelling of the performance fluctuations of a diver detection sonar using both sonar data and local environmental measurements.
Asset protection using autonomous underwater vehicles

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Abstract: Asset protection is becoming an important area of research for the underwater community. In recent years, we have been involved in several projects with the Office of Naval Research to tackle the problem of confined area inspection such as harbours and ship hulls. In this paper, the algorithms for detection and classification of targets are presented. High resolution video-rate sonar systems are increasingly used for such identification tasks such as the Blueview and DiDson acoustic cameras. Which such sensors, techniques that were previously only used on optical video data can now be used for acoustic data. We present three algorithms providing the key enabling technologies for large-scale survey of confined areas for IEDs and mine-like target. The first algorithm performs precise motion estimation from acoustic video streams to produce large scale mosaics of the surveyed areas. This enables to create a large scale image of the area that can be used by the operator to perform visual inspection. This also enables to ensure that no holidays are present in the data and to monitor coverage. Very large compression of the data flow are achieved making it suitable for transfer on low-bandwidth acoustic communication systems. The second algorithm performs target detection and tracking. It is based on a data fusion engine fusing several low-level detectors. The detection are fed into a tracking module which estimates the track of the objects in the video stream and removes false alarms. The final algorithm performs the identification of the targets. This uses machine learning techniques such as ada-boost (cascade classifiers) and rule-based classification tools. The algorithms are tested and demonstrated on real robots in at-sea experiments. The detection and identification performances are evaluated and reported in the paper.
Acoustic Monitoring of Terrorist Intrusion in a Drinking Water Network

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Abstract: The dependence of cities on their water supplies has been known for centuries. A well known technique, in the Middle Ages, to conquer a besieged city was to poison its water sources by throwing infected corpses in. In collaboration with KWR, Watercycle Research Institute (formerly known as Kiwa Water Research), TNO (Netherlands Organisation for Applied Scientific Research) has investigated the possibilities to detect and classify aberrant sounds in water networks, using acoustic sensors. Amongst the sources of such sounds are pumps, drills, mechanical impacts, which could, for instance, indicate a terrorist’s attempt to inject a toxicant into the water mains. In parallel, an important spin-off is recognized in the detection of sounds caused by leaks. The Acoustic water pipe monitoring project was carried out between 2003 and 2007 in three distinct phases. A first phase was carried out in 2003-2004 to gain knowledge on the subject. Models were developed for the sound propagation in water pipes and computer simulations were carried out to specify detection ranges. Subsequently, in 2005-2006, the knowledge created in phase 1 was applied for detection of leakage and terrorist attacks to the network. For this purpose a number of field experiments were carried out to characterize background noise and suspicious sounds. In the last phase, carried out in 2006-2007, a demonstrator aiming at terrorist detection was developed and tested on an operational water supply system. Overall, the TNO acoustic demonstrator, equipped with two sensors fixed on fire hydrants and two on service taps, has proved to be a suitable and promising system for building protection: it is capable of working real time and detected about 90% of terrorist intrusion sounds within a range of 110 m. Using algorithms developed for leakage detection in phase 2, an equivalent demonstrator could assess possibilities for leakage detection.
Structured Session 8

**Biosonar**

Organizers: Peter Dobbins & William Megill
SOUND RECEPTION IN BOTTLENOSE DOLPHINS (TURSIOPS TRUNCATUS) – RESULTS FROM 2-D AND 3-D MODELLING

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Abstract: Odontocetes use active sonar (echolocation) for hunting, navigation and socialisation. This sonar is characterised by narrow transmission and reception directivity patterns, over a variety of ranges and frequencies. While the process of sound transmission is already well understood, it is still not entirely clear how odontocetes receive sound. According to the prevalent model, dolphins receive sound through a thin region in their lower jaw (pan bone) and sound is then transmitted to the inner ear. A different model suggests however that the teeth in a dolphin’s lower jaw play also a role in sound reception, acting like an end-fire sonar array. Our work has modelled sound reception for a bottlenose dolphin (Tursiops truncatus) using the acoustic characteristics of the different components of the entire jaw. Earlier results showed the importance of multiple scattering, secondary peaks from neighbouring teeth mixing with the direct arrivals on individual teeth, as well as the strong influence from the relative positions and sizes of the teeth. Our later work shows that teeth cannot be considered as individual, point-like receivers. Enhanced directional hearing occurs for particular orientations of narrow-band plane waves coming in the plane of the jaw. The entire jaw has been modelled with a 3-D finite-element software (ANSYS), using the acoustic properties of teeth, skin, jawbone and tooth nerves measured on live and dead dolphins in a series of studies by other workers. The 3-D approach allows modelling of the reception of out-of-plane waves and their interaction with all elements of the entire jaw.

Keywords: Sonar, acoustic modelling, directional hearing, bottlenose dolphin
1. INTRODUCTION

Dolphins use active sonar for navigation, communication, foraging and hunting. The exact process of sound reception is still not entirely clear. Many researchers believe that sound is solely received through the acoustic window in the lower jaw of a dolphin and that it is then transmitted to the dolphin’s ear. The role of the acoustic window or pan bone in sound reception has already been demonstrated in many experiments [e.g. 1-7]. [3,4] for example conducted experiments with trained dolphins, placing a rubber hood over the lower jaw and teeth to test their navigational capabilities. The animals showed decreased directional hearing. A different view on sound reception in toothed whales is that teeth [8-11] or the ‘gular pathway’ in posterior mandibles [12] act as primary acoustic pathway.

This paper investigates the role of dolphins’ teeth in directional hearing. Observations show the teeth in the lower jaw of a bottlenose dolphin (Tursiops truncatus) are regularly spaced and placed at an angle of 10-20° with deviations, depending on individuals. Dolphins’ teeth are homodont, which means that all teeth have the same shape. [8] suggested that the teeth act as a passive resonant receiver, combined as two equispaced line arrays, with the nerves introducing progressive delays, as in a delay-line beamformer. The slow propagation of nerve impulses implied a progressively delayed response related to the position of teeth in the jaw. This topic was followed by [9,10], who suggested that combining the rows of the teeth in a monopulse configuration would yield accurate angular resolution with wide beams for rapid searching. [9,10] also suggested that an echo arriving from a direction along the axis of the row of teeth (end-fire) would combine constructively on all teeth, thus providing an enhanced signal for analysis by the central nervous system. We follow on these studies by investigating the role of tooth size and positions and how this impacts on the sound received at different directions.

2. METHODS

2.1. 2-D Modelling

As a first approach, the 2-D model of a dolphin’s teeth in its lower jaw was constructed using a cast from a real dolphin. Bottlenose dolphins have 44 teeth (22 on each side) in their lower jaw. These are set approximately 10 mm apart in average (Fig. 1). The diameters of the teeth vary with individuals, with their position within the jaw (larger toward the back and smaller toward the front) and with the point of measurement (i.e. near the tip or at the base). For the purpose of acoustic modelling, we have used tip and base diameters to create two different input datasets. Our model can account for small variations in teeth positions and sizes. The 2-D modelling of sound propagation in the dolphin’s jaw is implemented in Matlab, using a time-domain solution of the acoustic wave equation. This third-party program uses pseudo-spectral methods to calculate spatial derivatives and a staggered Adams-Bashforth method to integrate forward in time [13]. A perfectly-matched boundary layer is applied at the edges of the calculation domain. This model can include nonlinear propagation and frequency dependent attenuation, although this was not used here. The model has been run for different orientations of narrow-band plane waves moving in the plane of the jaw, at frequencies typical of dolphin’s echolocation signals (30-150 kHz) and with a grid spacing of \( \lambda/8 \). Dolphin teeth vary with individuals and with their position (larger toward the back and
smaller toward the front), and whether they are measured near the tip or at the bottom. Typical sizes vary between $\lambda/2$ and $\lambda/4$, depending on the signal’s main frequency. Models have used both tip and bottom diameters from the cast of a dolphin’s jaw. The speed of sound in teeth varies, but current measurements indicate transverse and longitudinal velocities of 2200 m/s and 3380 m/s respectively [14]. The density of teeth used is the nominal value measured for a fully filled tooth of 2035.4 kg/m$^3$ [15]. The model has been run looking at different orientations of waves coming in the plane of the jaw.

![Fig.1: Dental arrangement of a typical bottlenose dolphin [10], as used for modelling.](image)

### 2.2. 3-D Modelling

The construction of an accurate 3-D model of a dolphin’s jaw is made possible by recent advances in acoustically-related measurements of its constituents (e.g. [14, 16-18]) and in the understanding of sound production in dolphins and whales [19]. Laser Doppler Vibrometry measurements of dolphins’ teeth [14] have shown for example that they have strong resonances at 115-135 kHz. Inner variations in each tooth (e.g. density, tooth age) are related to their immediate surroundings. The skin of a dolphin’s jaw consists of three layers: the epidermis, dermis and hypodermis (blubber). The thickness of epidermis and dermis of six different areas was measured by [18]. The lowest measurement of the epidermis/dermis at the back of the dolphin was 0.17 cm, whereas the lowest thickness of the hypodermis was 1.54 cm. Since the skin around the dolphin’s mouth is assumed to be thinner than at the rest of the body, this lowest value can be used. Current estimates of skin sound velocity are close to 1600 m/s [18]. It is also less dense, close to 969 kg/m$^3$ in average. The tooth sockets are placed in gums, whose acoustic properties are assumed similar to that of skin in the absence of any published measurements. The jaw bone is an important factor in the modelling. Its acoustic attenuation factor is very high (1.2 dB/mm according to [14,15]), leading some to conclude that the hard material of the jaw bone is not the primary pathway for sound reception, although recent measurements show there is some coupling [14,15]. Our model uses sound velocity for bones varying between 1900-3300 m/s and bone densities around 1785 kg/m$^3$ [20], commensurate with other in situ measurements. Fig. 2 shows a typical arrangement of the teeth in a dolphin’s jaw. The nerves below the teeth are also part of our 3-D model. Although they are comparatively small, the low velocity of sound propagation along the nerves (100 m/s) has clear influences on how and where appropriate acoustic signals are picked up and how they propagate (e.g. [14]). These individual components can be combined in a 3-D model of the jaw and teeth with the different fatty tissues. This is done using ANSYS, a finite-element analysis software able to calculate both harmonic and transient solutions.
3. RESULTS

3.1. 2-D Modelling

Sound propagation will evidently vary with the size and position of teeth. Both types of simulations (with tip or base diameters) show that teeth cannot be considered as individual, point-like receivers (Fig. 3). Multiple scattering is important, at all angles of rotation of the jaw. Secondary peaks from neighbouring teeth mix with the direct arrivals of the incoming sound on individual teeth. In some configurations, individual clicks might become indistinguishable. It is possible to identify individual teeth reflections on the acoustic signals measured at each single tooth. The first and most important peak in the acoustic signal corresponds to the direct arrival of the wave on the tooth considered. Sound is enhanced immediately after reception of the original signal, attenuating with time. Subsequent peaks in the signal have amplitudes of 10% or less, and are identified from their time of arrival to be reflections on nearby teeth. Reflections from some of the immediately neighbouring teeth cannot be identified, either because they are subsumed into the signal scattered by the previous tooth or because they are masked. This will of course vary with the jaw geometry and with the angle of the incoming wave. This result shows the limits of considering dolphin teeth as point-like receivers, and analyses of their sonar reception abilities (or how to emulate them) should bear this point in mind.

These effects are exacerbated as teeth increase in diameter (e.g. when looking at the bottom of the teeth). They significantly modify the signals received by each tooth. Furthermore, these effects are only modelled here in the horizontal plane. 3-D propagation is likely to enhance these effects, especially as the model is refined and other acoustic scatterers are considered (blubber, skin, actual jawbone, etc.). Sound propagation was modelled for a plane wave going through the base (Fig. 4, left) and the tip (Fig. 4, right) of the teeth at different angles. The incoming plane wave comes from the bottom of each graph, and multiple scattering between teeth can be readily identified. The acoustic amplitudes are colour-coded and normalised to the same reference value. There are clear areas of enhanced sound, moving away from the jaw, and areas of reduced sound, their size extending toward the back of the jaw as the wave propagates. Teeth further away tend to be masked by teeth in the line of sight from the transmitter. Finally, there are clear indications of multiple scattering. Signals backscattering on teeth further back in the jaw create signals later in time on teeth already ensonified and, depending on teeth positions, it can be as high as 25% of the maximum level.
Fig. 3: Signals received on the left (L) side of a dolphin’s lower jaw at respective teeth numbers 2, 3, 4 and 5 (L2-L5), normalised to the incoming plane wave amplitude.

Fig. 4: Sound propagation varies with the angle of the jaw to the incoming wave (from top to bottom: 0° to 15° by 5° intervals, for a plane wave coming from the bottom of each graph). Sound amplitudes are colour-coded and normalised to the same reference value.
For 0° (end-fire configuration), the signal shows a symmetric distribution, with the highest amplitude at the front of the teeth. There are low amplitudes at the remainder of the teeth and no sound in the middle of the jaw between the teeth. For angles increasing by 5° (to 15°), the acoustic amplitudes are getting higher at teeth towards the end of the jaw and lower on the other side of the jaw. Sound on the tip of the teeth is higher than on the base of the teeth. The prediction indicates that the pressure at the teeth is highest nearest the incident wave; behind the teeth the amplitude of the sound is much lower. The complete simulations were run for 1° intervals for an incoming signal at angles from -90° to +90°. They show high variations with the relative angles between the jaw and incoming acoustic waves, with most of the directional hearing between approximately ±10°. This result supports the angular resolutions observed for dolphins (e.g. [7, 10]). The complete simulations confirm the relative decoupling of the left and right portions of the jaw, and significantly the importance of neighbouring teeth in attenuating and/or amplifying the signal received at other teeth. This directional hearing can be amplified or attenuated depending on the tooth configurations.

Fig. 5: 3-D models are sensitive to mesh size and pattern. Left: loose meshing (top) and inaccurate solution (bottom). Right: denser meshing (top) and accurate solution (bottom).

3.2. 3-D Modelling

The 2-D results correspond to the case of an acoustic wave exactly in the plane of the jaw; this might not always be a realistic situation. The 2-D model is also focusing on the teeth and neglecting the acoustic properties of surrounding tissues, the skin and the jawbone itself. It does not aim to explain how the signals can be picked up by the tooth nerves and integrated into a coherent echolocation. To extend these simulations, we use ANSYS. It has been well validated by different users for 2-D and 3-D cases [e.g. 21]. First investigations have aimed at
optimising the number of mesh divisions per wavelength (i.e. the spatial resolution of the model) and the integration time steps (i.e. the temporal resolution). Time steps have been fixed here at 10 μs. As a general rule, solid structures can be modelled with pre-defined element SOLID45 (8 nodes, 3 degrees of freedom at each node, user-defined density and sound speeds). Acoustic elements in contact with the solid parts can be modelled with the FLUID30 element and those further away with the FLUID130 element (absorbing the pressure waves and thus simulating the outgoing effects of a domain that extends to infinity). Seawater is assumed compressible, with only relatively small pressure changes. First results show the sensitivity of the solution to mesh size (Fig. 5); looser meshing for a spherical object with the acoustic properties of a dolphin tooth (top left) will reduce the accuracy of the acoustic solution for excitation by a plane wave (bottom left). Conversely, denser meshing (top right) will result in more accurate acoustic calculations (bottom right), at the expense of computing time and the overall number of mesh cells available for other elements.

4. CONCLUSIONS

Field observations show that dolphin teeth are aligned in the lower jaw in a pattern similar to hydrophones in an efficient end-fire array. These results have been used both to explain sound reception by dolphins and to design new biosonars. We have modelled the propagation of an acoustic signal (typical of a dolphin echolocation signal) in 2-D and 3-D. Our 2-D model uses measurements of base and tip diameters of teeth from the cast of a dolphin’s jaw, and acoustic velocities and densities averaged from field records of live and dead dolphins. We have in particular investigated the role of the angle of arrival of a plane wave, and the sound levels received on all individual teeth. The first result of these simulations is the multiple scattering between teeth, some being masked and others seeing the incoming signal amplified significantly. Teeth cannot be considered as point-like receivers, their sizes as well as their positions influencing the overall signal levels. Another main result is the directional hearing of a typical dolphin jaw, consistent with field observations. The 3-D results show the influence of simulation parameters on the accuracy of the solution: they will be used to investigate the contribution of other elements of the jaw (such as bone and tissue) to the acoustic signal received at different parts of the jaw and how it can be analysed by the dolphin’s central nervous system.

5. ACKNOWLEDGEMENTS

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REFERENCES


COMPARISONS OF COMMON WHISTLES IN TWO POPULATIONS OF GUIANA DOLPHINS, *Sotalia guianensis* (Cetacea, Delphinidae) IN SOUTHEASTERN BRAZIL

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Abstract

The common whistles of two populations of Guiana dolphins (*Sotalia guianensis*) were compared in two areas in Southeastern Brazil (Guanabara Bay, 22° 57’S, 43° 10’W, Rio de Janeiro state and Puruba, 23° 23’S, 45° 56’W, São Paulo state). The whistles were classified into different types, according to the contour similarity. Six acoustic parameters of each whistle were measured: duration, start frequency, end frequency, frequency ¼, ½ and ¾ of duration. A total of 399 common whistles were analyzed and the ascending form represented 78.69% (N=314) were classified into four types (T2, T3, T4 and T5). The spectral contour ascending-descending (T1) was found in two populations of Guiana dolphin. Whistle types presented similar characteristics, especially in relation to the parameters of frequency. The duration of whistles was significantly different in three types analyzed (T1, T3 and T5), indicating the importance of this parameter in determining the differences in the repertoire of the whistles among populations. Intraspecific variations and social function of whistles are relevant to the comparison of the acoustic repertoire of Guiana dolphin populations.

Keywords: whistles, Guiana dolphin, Sotalia guianensis, acoustic parameters
I. Introduction

The whistles of Odontoceti are related to different behavioral contexts, the socialization of the group and individual recognition (Caldwell et al., 1990; Tyack, 1999; Podos et al., 2002). Comparative studies indicate that whistles present variations in acoustic structure among populations of the same and different species (Wang et al., 1995a, b; Bazuá-Durán & Au, 2004). This variability may be related to such factors as: adaptation to different environments, the variation in body size and learning (Wang et al., 1995a, Mathews et al., 1999; Rossi-Santos & Podos, 2006).

Recently studies have shown an intraspecific variability of the Guiana dolphin whistles, Sotalia guianensis (Van Bénéden, 1864) (Azevedo & Van Sluys, 2005; Rossi-Santos & Podos, 2006) The Guiana dolphin is a small delphinid and is found in the South and Central America, from Southern Brazil (27°35’S, 48° 34’W) to Nicaragua (14° 35’N, 83° 14’W), including some records in Honduras (15° 58’N, 85° 42 W) (Flores & da Silva, 2009). The species is classified as insufficiently known by the International Union for the Conservation of Nature and Nature Research - IUCN (Reeves et al., 2003). The main threats for this species are represented by destruction of the habitat, pollution caused by human activities and incidental capture in fisheries (da Silva & Best, 1994; Lailson, 2000; Di Benedito et al., 2001). The aim of this study was the comparison of the common whistles between two populations of the Guiana dolphin in Southeastern Brazil.

II. Materials and Methods

A. Study sites

During the period of May 2002 and February 2004, the whistles of Guiana dolphin were recorded in Guanabara Bay, Rio de Janeiro state (22° 57’S, 43° 10’W) and the Puruba beach, São Paulo state (23°23’S, 45°56’W), in Southeastern Brazil (Figure 1).

![Fig.1: Location of two sample sites, Guanabara Bay (22° 57’S, 43° 10’W), Rio de Janeiro state and Puruba beach (23°23’S, 45°56’W), São Paulo state in Brazilian coast.](image)
The Guanabara Bay is an estuary with a diverse ecosystem, high rate urbanization and has large impacts related to human activities such as landfills, deforestation, traffic of vessels, industrial and domestic sewage (Amador, 1997). The Puruba beach is a coastal ecosystem formed by remains of Atlantic rainforest and mangroves, with low rate of human occupation, and fishing as the main activity.

B. Recordings and Whistles Analysis

Acoustics recordings were made with appropriate weather conditions (Beaufort sea state ≤ 2), using motor boats from 4 to 10 meters in length, with the engine off and were monitored using headphones. Different individuals, groups and behaviors were considered to avoid problems in independence of data. The recording system was composed of a High Tech hydrophone, model HTI-96-MIN, sensitivity: -175 dB, with frequency response from 5 Hz to 30 kHz (±1.0 dB, -165 dB re: 1 V/μPa) at 1 m depth, connected to a digital recorded SONY TCD-D8 with the upper frequency limit of 24 kHz (sampling rate of 48 kHz).

The recordings were redigitized using Coll Edit Pro 1.0 (Syntrillium Software), with sampling rate of 48 kHz (16 bit). The analyses of spectrograms were made with the software Coll Edit Pro 2.1 (FFT size of 256 points, using a Hamming window). In the acoustic analysis were selected only whistles with the same visual contour shape.

Whistles were separated visually by two observers to avoid possible errors of classification (Janik, 1999) and were classified into six contour categories (ascending, descending, constant, ascending-descending, descending-ascending and multiple) described by Azevedo & Van Sluys (2005). Six acoustic parameters were measured from each whistle: duration (DUR), start frequency (SF), end frequency (EF), frequency at ¼ of duration (F1/4), frequency at ½ of duration (F1/2) and frequency at ¾ of duration (F3/4). The whistle duration was measured in milliseconds and the frequency variables in kHz. These acoustic parameters were selected to be consistent with others studies of Guiana dolphin whistles (Simão & Azevedo, 2002; Azevedo & Van Sluys, 2005; Rossi-Santos & Podos, 2006) and others species of Odontoceti (Rendell et al., 1999; Wang et al., 1995b). Chi-square test ($\chi^2$, $P<0.01$) was used to compare the distribution of whistle types in two populations (Siegel, 1975). The descriptive statistics was applied to all acoustic parameters of whistles, including the mean and standard deviation. The Mann-Whitney test ($U$ test) was used to verify the differences in the acoustic parameters of common whistles between two Guiana dolphin populations.

III. Results and Discussion

In Guanabara Bay, the recordings were made for 10 days, with 70 dolphins and three behavioral categories (feeding, socialization and traveling) were observed. In Puruba beach the recordings were made during one day, with 30 dolphins and were documented two behavioral categories (feeding and traveling). In both areas, groups of $S. guianensis$ were composed by adults, juveniles and calves. The recording time analyzed in Guanabara Bay was 2hs 3min 28s and in Puruba beach was 1h 7min 46s.

A total of 621 whistles were selected (215 whistles in Guanabara Bay and 406 whistles in Puruba beach). Whistles with contour similarity corresponded to 94.5% of all whistles ($N=587$) and these were separated in 22 common whistle types (196 whistles of the Guanabara Bay and 391 whistles of the Puruba beach). Only types that were registered 10 or more times were included in the analysis for comparison of the acoustic parameters (120 whistles in Guanabara Bay and 279 whistles in Puruba beach). The Guiana dolphin presents a
varied repertoire of whistles but with a predominant contour form. The ascending whistles were predominant (78.69%) with 314 whistles classified into four types (T2, T3, T4 and T5) (Figure 2). The ascending whistles were also more abundant in other studies with this species (Simão & Azevedo, 2002; Pivari & Rosso, 2005; Rossi-Santos & Podos, 2006). One whistle type with ascending-descending contour shape (T1) was common in both populations of the Guiana dolphin (Figure 2).

Fig.2: Spectrogram representation of five common whistle types in two populations of Guiana dolphin in Guanabara Bay, Rio de Janeiro state, 22°57’S, 43°10’W and Puruba beach São Paulo state, 23°23’S, 45°56’W, in Brazil.

The distribution of five common whistle types was significantly different in two sites ($X^2=75.5$, df=4, $P<0.001$) (Table 1). The whistle T5 (74.35%) was predominant in Guanabara Bay while the whistles T1 (88.23%) and T4 (85.56%) were predominant in Puruba beach (Table 1).
Areas | T1 | T2 | T3 | T4 | T5 | Total
--- | --- | --- | --- | --- | --- | ---
Guanabara Bay | 10 | 44 | 23 | 14 | 29 | 120
Puruba beach | 75 | 89 | 22 | 83 | 10 | 279
Total | 85 | 133 | 45 | 97 | 39 | 399

Table 1: Number of common whistles for each type recorded in Guanabara Bay, Rio de Janeiro, 22º57’S, 43º10’W and Puruba beach, São Paulo 23º23’S, 45º56’W, in Brazil ($X^2=75.5$, $df=4$, $P<0.001$).

The average duration of five whistle types in the Guanabara Bay was higher than in the Puruba beach (Table 2). A total of 70 whistles (seven whistles of each type in each area) were randomly selected for comparison of acoustic parameters between two populations of the Guiana dolphins (Table 3). The duration of whistles was significantly different in three common whistle types (T1, T3 and T5, $P<0.05$). A recently study compared whistles of *S. guianensis* along Southern and Northern sites in Brazil and did not identify significant differences in the duration (Azevedo & Van Sluys, 2005).

<table>
<thead>
<tr>
<th></th>
<th>T1</th>
<th>T2</th>
<th>T3</th>
<th>T4</th>
<th>T5</th>
</tr>
</thead>
<tbody>
<tr>
<td>Duration</td>
<td>2.36*</td>
<td>NS</td>
<td>2.49*</td>
<td>NS</td>
<td>2.30*</td>
</tr>
<tr>
<td>Start freq</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>-3.13*</td>
</tr>
<tr>
<td>End freq</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
</tr>
<tr>
<td>Frequency ¼</td>
<td>NS</td>
<td>NS</td>
<td>2.45*</td>
<td>NS</td>
<td>-3.13*</td>
</tr>
<tr>
<td>Frequency ½</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>-3.13*</td>
</tr>
<tr>
<td>Frequency ¾</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>NS</td>
<td>-2.23*</td>
</tr>
</tbody>
</table>

Table 3: Comparison of acoustic parameters of five common whistle types ($N=70$) of the *Sotalia guianensis* in both areas of study ($P<0.05$) (NS: $P>0.05$) (Mann-Whitney test, *Z values).

Differences observed in the duration parameter can be explained by social interaction of groups of the Guiana dolphin and related to environmental characteristics. The traffic of vessels and the types of human activities in Guanabara Bay and Puruba beach are factors which may interfere in sound emissions of Guiana dolphin. The adaptation to different habitats may have effects in the repertoire of whistles among populations of the same species (Steiner, 1981; May-Collado & Wartzok, 2008). Effects of geographical location in the structure of whistles were also reported in some studies (Azevedo & Van Sluys, 2005; Rossi-Santos & Podos, 2006).

The frequency parameters of whistles were similar between two areas, excepted whistle T5 that showed differences in duration and in start frequency, frequency at ¼, ½, ¾ of duration (Mann-Whitney test, $P<0.05$) (Table 3). The similarity of frequency parameters may be related to the maintenance of acoustic characteristics of the species *Sotalia guianensis* (Azevedo, 2000).
<table>
<thead>
<tr>
<th>Types of whistles</th>
<th>Guanabara Bay</th>
<th>Areas</th>
<th>Puruba beach</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>T1 (N=10)</td>
<td>T2 (N=44)</td>
<td>T3 (N=23)</td>
</tr>
<tr>
<td>Duration</td>
<td>174.80</td>
<td>109.59</td>
<td>320.26</td>
</tr>
<tr>
<td></td>
<td>±93.35</td>
<td>±58.95</td>
<td>±89.08</td>
</tr>
<tr>
<td>Start</td>
<td>10.38</td>
<td>10.05</td>
<td>7.9</td>
</tr>
<tr>
<td>Frequency</td>
<td>±2.12</td>
<td>±4.22</td>
<td>±1.78</td>
</tr>
<tr>
<td>Frequency</td>
<td>±4.22</td>
<td>±3.15</td>
<td>±1.53</td>
</tr>
<tr>
<td>Frequency</td>
<td>9.48</td>
<td>11.40</td>
<td>10.65</td>
</tr>
<tr>
<td>¼</td>
<td>±2.58</td>
<td>±4.25</td>
<td>±2.15</td>
</tr>
<tr>
<td>Frequency</td>
<td>11.49</td>
<td>13.44</td>
<td>13.97</td>
</tr>
<tr>
<td>½</td>
<td>±4.06</td>
<td>±3.85</td>
<td>±2.69</td>
</tr>
<tr>
<td>¾</td>
<td>±4.87</td>
<td>±3.62</td>
<td>±1.85</td>
</tr>
</tbody>
</table>

Table 2: Descriptive statistics of acoustic parameters of the common whistle types of the Gutana dolphin in both areas of study. Values are mean and standard deviation. The duration was measured in milliseconds and frequency parameters in kHz.
In general, animals produce sounds that reflect adjustments for local conditions (Peters et al., 2007). According to Wang et al., (1995a) the differences of the environment, as background noise, can lead to changes in the frequency range of whistles of different species of Odontoceti. The repeated occurrence of whistles in the two populations may represent an important characteristic of this sound emission type of the Guiana dolphin. The acoustic emissions have been an important tool for the ecology and behavior studies of this species. The variation of acoustics parameters may reflect the isolation of populations, adaptation to the environment, the ability to modulate the acoustic repertoire of whistles and different behavior contexts (Oswald et al., 2003; Bazuá-Duran & Au, 2004).

V. Acknowledgements

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VI. References


Hearing during Echolocation in the False Killer Whale (Pseudorca crassidens)

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Abstract: Most echolocating whales and dolphins produce short intense pulses which must be followed very shortly by an ability to listen for a much quieter echo. We have measured the hearing of a false killer whale for both its outgoing pulses and the echoes that return from targets during an active echolocation task by examining the whale’s evoked auditory potentials. We have found that: (1) the whale may hear her loud outgoing clicks and much quieter returning echoes at comparable levels, (2) the whale has protective mechanisms and may hear her outgoing signals at a level about 40 dB lower than comparable signals presented directly in front of her, (3) when we substantially change the echo return levels, either by making the targets smaller or by placing the targets farther away, the animal (without changing the levels of her outgoing signals) continues to hear the echoes at almost the same level (4) if targets are made much smaller and harder to echolocate, the animal will modify her hearing to increase what she hears of her outgoing signal as if to heighten overall hearing sensitivity to keep the echo level hearable, and (5) hears pure tone signals 20 dB lower when an echolocated target is present as compared to trials in which targets are absent. These findings lead to the conclusion that the animal has an active auditory automatic gain control process.
TIME AND FREQUENCY SHIFTED CLICK DETECTION

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Abstract: Spectrogram correlation is an effective and widely used technique for detecting and classifying cetacean echolocation clicks. Because different species operate in different frequency bands, different spectral templates are needed to detect them. However, examination of the spectrograms from a wide range of whales and dolphins suggests they are generally similar in form but, as already said, occupying different frequency ranges and spread over different time intervals. This suggests that if the clicks from different species can be translated in frequency and temporally normalised, one detector will suffice for all species. This would have many advantages: the frequency shifting is easily achieved with simple analogue electronics; if all signals are translated to baseband, a sampling rate of 96kHz would probably be adequate, so off-the-shelf audio equipment can be used; lower sampling rates mean reduced computing requirements. This paper investigates these ideas and proposes a system design that can carry out the required processes, will detect all species of echolocating cetacean, and classify the species from the degree of frequency shift required.

Keywords: Click detection, Passive Acoustic Monitoring, Marine Mammal, Cetacean
1. INTRODUCTION

The Detection, Classification and Localisation (DCL) of marine mammals, especially cetaceans, in the ocean using Passive Acoustic Monitoring (PAM) is becoming established as a tool for investigating free-ranging animals in their natural habitats [1], as well as a means for warning of the presence of marine mammals for environmental impact mitigation procedures [2]. Such systems detect either the echolocation clicks used by most cetaceans for navigation and detecting prey [3], or the lower frequency communication vocalisations such as the whistles used by most dolphins [4].

Nevertheless, not all cetaceans produce communication vocalisations or whistles, including many species of particular interest for environmental impact mitigation, such as beaked whales and porpoises. So, to cover all species a click detector must be used. Commonly used techniques for click detection include simple energy detectors that rely on the characteristics of a particular species’ click trains to discriminate between clicks and other transient noises [5], and spectrogram correlators [6], which are more versatile and, potentially, can classify a singleclick. However, spectrogram correlation is a computationally intensive process, especially as some species’ clicks require raw data sampling rates in the order of 400 kHz or more.

It is apparent that users of PAM want reliable, robust systems that are easy to deploy and preferably autonomous [7]. This means that it is often required to deploy devices for weeks, months or longer. Autonomous devices must have the endurance to operate for such periods on the power from a battery, possibly assisted by a solar panel. Clearly, it would be advantageous to reduce the computing requirements for click detection.

An approach is proposed here that transposes the received signals to a 0 - 48 kHz baseband. This allows reduced sampling rates of 96 kHz, standard for off-the-shelf (OTS) audio equipment, leading to reduced computing power. Additionally, the frequency transposition can be carried out using conventional analogue circuitry, as used in bat detectors, requiring much less power than digital processing.

Yet another advantage of the proposed technique is that the degree of frequency transposition needed to baseband a signal can be used to discriminate between groups of cetaceans such as porpoises, beaked whales, dolphins, and so on.

2. ECHOLOCATION CLICKS

Odontocete cetaceans produce echolocation clicks, whistles, cries, chirps, and similar sounds. Echolocation clicks range in frequency from a few kHz to about 150 kHz, in bandwidth from a single frequency to 100 kHz and in pulse length from a few tens of microseconds to milliseconds.

Recorded clicks were used to test the system described in this paper and represent four groups of species: sperm whales, beaked whales, dolphins and porpoises. Specifically, they were: sperm whale, *Physeter macrocephalus*; Cuvier's beaked whale, *Ziphius cavirostris*; Atlantic bottlenose dolphin, *Tursiops truncatus*; and the harbour porpoise, *Phocoena phocoena*.

Figs.1 and 2 show typical waveforms, spectra and spectrograms for clicks from two of these species; the sperm whale and Cuvier’s beaked whale. It is immediately apparent that they are similar in form, although differing in scale.
Fig. 1: Waveform (top), spectrum (centre) and spectrogram (bottom) of a typical sperm whale (*Physeter macrocephalus*) echolocation click.

Fig. 2: Waveform (top), spectrum (centre) and spectrogram (bottom) of a typical Cuvier’s beaked whale (*Ziphius cavirostris*) echolocation click.
In both these cases, the spectrogram could be represented as a broad ellipse. Higher resolution analyses carried out on bottlenose dolphin, *Tursiops truncatus* and sperm whale, *Physeter macrocephalus* clicks using Short-Time Fractional Fourier Transform (STFrFT) show that these clicks can be decomposed as two independent FM chirps [8], and this may be true for other species. However, such detail is not necessary for the process described here and, furthermore, would raise the computational load.

It seems that a spectrogram correlator based on an ellipsoidal kernel could be used to detect all these signals by suitably scaling either the kernel or the spectrogram derived from the signals. In the example shown in Figs.1 and 2, the sperm whale could be detected using an ellipse about 0.6ms long and 10 kHz wide, centred on 10 kHz. The beaked whale, on the other hand, would require an ellipse about 0.3ms long and 20kHz wide, centred at about 35 kHz.

However, if the beaked whale signal were transposed downwards by 25 kHz, this ellipse could be centred on 10 kHz, the same as the sperm whale.

### 3. THE DETECTOR CONCEPT

Fig.3 shows the main components of the proposed detector and explains its operation. The signal is input at the left and, following signal conditioning, is fed to a balanced mixer. A local oscillator is also fed to the mixer and the oscillator frequency controls the degree of frequency downshift at the mixer output. For a single frequency input, the frequency of the output (assuming unwanted components have been filtered out) is just

\[ f_{\text{OUT}} = f_{\text{IN}} - f_{\text{OSC}} \]  

(1)

where \( f_{\text{OUT}} \) is the output frequency, \( f_{\text{IN}} \) is the input frequency and \( f_{\text{OSC}} \) is the oscillator frequency.

![Fig.3: Block diagram showing main components of the detector.](image)

For example, if the bandwidth of the beaked whale click in Fig.2 is taken as 25 – 45kHz and an oscillator frequency of 25kHz was selected, then the click would be shifted to the band 0 - 20kHz.

The output from the mixer is digitised and a spectrogram computed. This is passed to a correlator which uses a kernel selected by the same controller that selects the local oscillator.
frequency. Thus, if at regular intervals the controller steps through oscillator frequencies and matching kernels to suit the different groups of species, when a detection is registered, the species group is automatically classified simply by the selected oscillator frequency and no additional computation is required.

4. AN EXPERIMENT

4.1. Set-up

An experiment was carried out to test the detector concept. This was a simplified version of the block diagram in Fig.3 and selecting the oscillator frequency and kernel were carried out manually. A sketch of the experimental set-up is shown in Fig.4.

Fig.4: Experimental set-up.

The functions of signal conditioning and the mixer were carried out by a superhet bat detector with its own internal local oscillator disabled. Both recorded click signals and the local oscillator signal were played from a laptop running a simple NI LabView routine and output via a NI USB-6251 DAQ device. The bat detector output was digitised by an Edirol UA-25 24 bit audio capture device sampling at 96 kHz. The spectrogram correlation function was carried out in a second laptop running Ishmael.

4.2. Procedure

The procedure adopted went as follows: for each species group a selection of approximately 50 different pre-recorded clicks were played back in a continuous loop. The local oscillator and associated spectrogram correlator kernel were then stepped through the appropriate selections for each species group, pausing long enough for the full 50 different clicks to be tested. For each species group selection, the mean amplitude of the correlator output spikes (if any) was noted.

There was not adequate data in this simple experiment to produce statistics of probability of detection or false alarm rate, but the mean detector output gives a good indication of how reliably an oscillator/kernel setting detects its own target species group along with how strong, relatively, are the false detections for other species groups.

The results from this exercise were then compiled into a confusion matrix. At this point, it should be noted that, to date, no attempt has been made to optimise either the spectrogram correlation kernels, or the degree of frequency downshifting. With this in mind, it is likely that the results obtained can be improved.
4.3. Results

Fig. 5 shows the waveform, spectrum and spectrogram of the beaked whale click downshifted 25 kHz as in the example in Section 3. Comparison with Fig. 2 will show that the spectrogram is essentially the same but, as expected, transposed downwards.

![Waveform, Spectrum, and Spectrogram](image)

**Fig. 5:** Waveform (top), spectrum (centre) and spectrogram (bottom) of Cuvier’s beaked whale (Ziphius cavirostris) echolocation click downshifted by 25 kHz.

Detection performance is demonstrated in Fig. 6. Here, a beaked whale click and a sperm whale click have been combined in one waveform, seen at the top of the figure. Below that is the spectrogram and below that the spectrogram obtained after downshifting by 25 kHz. Note that the spectrograms for the beaked whale click appear different from previous examples because the sampling rate has been reduced and hence the frequency scale altered.

It is seen that some remnant of the sperm whale click remains in the downshifted spectrogram because the input filtering on the bat detector is not matched to this application. However, it can be seen in the spectrogram correlator output at the bottom of the figure, using a beaked whale kernel, that there is a strong response for the beaked whale but nothing visible for the sperm whale. The filtering problem is also visible in the confusion matrix shown in Fig. 7.
Fig. 6: From the top: waveform, spectrogram, downshifted spectrogram and correlator output with a beaked whale kernel for a combination of Cuvier’s beaked whale (left) and sperm whale clicks (right).

Fig. 7: Confusion matrix showing response of the four detector settings (columns) to the four input click types (rows). Response is shaded from black for 0% to white for 100%.
The confusion matrix, Fig. 7, shows the average response of the four detector types in the columns to the four input click types in rows. For example, the cell in the second row, third column represents the response of the dolphin detector to a beaked whale click. Response is shaded from black for 0% to white for 100% and refers to the height of the correlation peak averaged over approximately 50 different clicks from the same species.

As expected, responses in the main diagonal, representing the response of a detector to its target click type, are all close to 100%. The remaining cells all show a response in the range of 0 – 20%, except for the row representing a dolphin click input, where the non-target beaked whale and porpoise responses are about 50% and the sperm whale about 25%. This is almost certainly because the dolphin click occupies such a wide bandwidth that there is energy leaking into the bands occupied by other species, but to a lesser extent with the sperm whale whose clicks are mainly below the dolphin’s bandwidth.

5. CONCLUSIONS

The results presented in this paper demonstrate the feasibility of a cetacean click detector based on the concept of downshifting incoming signals by an amount depending on the target species’ frequency range. This allows the sampling rate to be minimised so computational load is reduced and OTS audio equipment can be used.

A simple experiment has demonstrated the ability to detect and classify four different species groups with reasonable detection and false alarm rates, although the system has not yet been optimised. In particular, the input filtering in the bat detector used for downshifting was inadequate for this application.

REFERENCES

MEASUREMENT AND DATA ANALYSIS METHOD ENABLING SONAR STUDIES ON MULTIPLE FREE-SWIMMING AND SPONTANEOUSLY ECHOLOCATING DOLPHINS

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Abstract

Analyzing data in object investigation sonar studies on individual dolphins have always been challenging, especially in setups where multiple free-swimming dolphins can echolocate spontaneously and concurrently at the sonar targets. A 47-element array, 0.75 m by 0.75 m, was constructed in order to generate spatially high resolved measurement data under such circumstances. The system was designed and optimized for burst mode sampling since continuous sampling of all hydrophones would generate unmanageable data volume flow rates. The method allowed simultaneous sampling of all individual channels with a sample rate of 1 MHz, 12 bit resolution and real-time analysis and visualization of data. Different ways of separating overlapping click trains originating from different individual animals were tested and evaluated. It is concluded that the presented measurement and data analysis approach makes spatially high resolved biosonar studies feasible also where multiple dolphins may echolocate with overlapping click trains at the same sonar target.

Keywords: Biosonar multiple free-swimming dolphins
Introduction

Most dolphin sonar studies have mainly strived to characterize properties of the beam axis while the animal under test performed various sonar tasks (Au 1993, Au and Hastings 2008), e.g. object detection or object discrimination tasks. In order to keep track of the beam axis, animals have either been trained to voluntarily station in fixed test rigs (Au et al. 1993) or to carry a recording device in its mouth (Sigurdson, 1996, Sigurdson 1997, Martin et al. 2005). These setups have only required one or a few hydrophones and have been used under the assumption that the beam axis orientation within the beam is fixed relative to the rostrum. However, recent findings on the beam steering and beam width manipulation ability of one Atlantic bottlenose dolphin (Tursiops Truncatus) (Moore et al. 2008) imply an increased need for sonar measurements with high spatial resolution within the beam and a comprehensive beam width coverage.

In addition, it is likely that the static research setups with stationary animals, performing sonar tasks under command and isolated from the rest of the pod has restricted the full dynamic use of the sonar (Sigurdson, 1996). Therefore, more research on free-swimming dolphins in groups is strongly motivated.

However, analysing sonar data acquired in setups with multiple free-swimming dolphins is challenging when echolocation click trains from different individuals overlap (Akamatsu et al.). It is especially challenging when the dolphins swim close to each other, so that acoustic source positioning algorithms are unable to resolve the separate positions of individual animas. This paper compares various methods for automatic and semi-automatic separation of overlapping click trains, based on the analysis of four click properties: peak frequency, amplitude-weighted mean frequency, relative energy within the cross-section of the beam and finally the correlation of the whole frequency spectrum in successive clicks. The sonar activity of a group of 19 free-swimming and spontaneously echolocating dolphins was recorded with a 47-element array optimized for high temporal and spatial resolution, comprehensive beam width coverage and long recording times. Methods both in the time and frequency domains were tested and compared based on the applicability for the specific measurement system used in this paper.

Method

The echolocation activity of a group of 19 free-swimming and spontaneously echolocation dolphins were recorded. The animals were kept in a large open sea pen with variable water depths ranging from 0 m to approximately 5 m. A 0.75 m by 0.75 m 47-element hydrophone array was used for recording the sonar activity. The hydrophones were arranged in a 7 by 7 element array, minus the two upper corners. See Fig. 1 and 2. All hydrophones were sampled simultaneously with 1MHz sampling rate and 12-bit resolution. The system was set to acquire data during a time window of 150 μs for each trig-event, ensuring that only the waveforms, and not the silent periods between clicks, were captured and stored to a hard drive. This data acquisition method is here referred to as burst-mode sampling and enabled a low data volume flow rate, as opposed to continuous sampling, and therefore also supported prolonged recording periods of the 47-element array system. A pre-trig of 40 μs and 47 individual filters and pre-amplifiers were used for the measurements.

Objects were held in front of the submerged array to stimulate the dolphins to spontaneously echolocate towards the hydrophones in the array. See Fig. 1. Recordings were made during a 17.5 minutes session. One specific echolocation scenario was singled out and
used as a type example of measurements where multiple dolphins echolocated concurrently towards the same sonar target (recording time 312.00-315.50 s).

Figure 1. A. The hydrophone array with sonar target in front of screen. B. The experimental setup allowed multiple free-swimming dolphins to spontaneously echolocate concurrently at the same sonar target.

The four click parameters peak frequencies of clicks, amplitude-weighted mean frequencies, relative measured energy (E) across the array and frequency spectral correlations (C) between successive clicks were evaluated for their usefulness in separating overlapping click trains originating from multiple animals.

The relative measured energy (E) across the array was calculated as

$$E_k = \frac{1}{T \cdot H} \sum_t \sum_h u_h^2(h,t)$$  \hspace{1cm} (1)

where

k = 1, 2, 3… N

N = number of measured clicks in the echolocation sequence

t = 1, 2, 3… T

T = number of samples in each click acquisition

h = 1, 2, 3… H

H = number of hydrophones in the array

u = Measured amplitude

The correlations were calculated as

$$C_k = \frac{1}{\max \left( \sum_i U(f_{k,i}) \cdot \sum_i U(f_{k+1,i}) \right)} \sum_i U(f_{k,i}) \cdot U(f_{k+1,i})$$  \hspace{1cm} (2)

where

k = 1, 2, 3 … N-1

N = number of measured clicks in the echolocation sequence

i = 1, 2, 3 … M

M = number of components in the frequency spectra

C_k = the correlation coefficient of the k:th click
The correlation was performed successively over the sequence of clicks, starting with the first click correlated to the second click, the second click correlated to the third click and so on. Correlation coefficients greater than 0.95 were assumed to belong to one and the same animal.

Results

The echolocation activity of the free-swimming and spontaneously echolocating dolphins, directed towards the object in front of the screen, was successfully captured with the system. The measurements produced 43 MB data during the recorded 17.5 minutes recording session. As an example, one single click measured with the simultaneously sampling array is presented in Fig. 2.

All successive figures show echolocation click parameters from the three second long sequence that was singled out for a detailed analysis. Fig. 3 A shows the peak frequencies of each click measured with one of the hydrophones (17 in Fig. 2) during the sequence. The weighted mean frequencies from each click, measured with the same hydrophone (17), in the sequence are shown in Fig. 3 B. The measured relative energies across all the hydrophones in the array are plotted in Fig. 3 C. The result from correlation calculations of successive clicks during the sequence are shown in Fig. 3 D. The calculated coefficient values of the correlations between the clicks are indicated with a square box or with red crosses (x'es) when the coefficient falls below the 0.95 limit (red dashed line).
Fig. 3. A. Peak frequencies measured with one hydrophone in the array during the studied echolocation sequence. B. The amplitude-weighted mean frequency of each click measured with one of the hydrophones during the three seconds long echolocation sequence. C. Normalized relative energies measured across the array throughout the echolocation sequence. D. Correlation result of the frequency spectra content of successive clicks calculated for one hydrophone throughout the echolocation sequence.

The time positions of the correlation results that indicated a change of echolocating animal during the sequence (red crosses in Fig. 3 D) were combined with the energy data in Fig. 3 C and are plotted together in Fig. 4. For the purpose of comparing the results in Fig 3C and D the vertical position of each red cross are moved to the mean value of the energy amplitude levels of the non-correlating clicks.

Fig. 4. Relative energies measured across the array plotted together with the red crosses where the correlation coefficients fall below 0.95.
Fig. 5 A shows the relative energies across the array, colour and marker shape coded based on the results of the frequency spectra correlations in Fig. 3 D. The clicks originating from one dolphin is coded as triangles and the clicks from another dolphin is coded as blue squares. Clicks with low correlations to all other clicks are marked as unfilled circles.

Discussion

The acquisition of the echolocation activity data in the group of 19 free-swimming and spontaneously echolocating animals were successful and are exemplified by the click in Fig. 2. The spatial and temporal resolution and the comprehensive beam width coverage resulted in many measured sequences where multiple dolphins echolocated simultaneously towards the array with overlapping click trains. Since previously used sonar recording systems with a relatively large number of hydrophones have been incapable of recordings during extended time periods, the specific acquisition method designed for this study was essential for this work since it made it possible to capture the high resolution recordings and sequences presented here. Dolphins did echolocate concurrently at the submerged object in front of the hydrophone array and there were several such sequences to use as type examples in this study.
The peak frequencies shown in Fig. 3 A does not alone facilitate a separation of overlapping click trains. There is, however, an indication that more than one dolphin was acoustically active during the sequence due to the frequency shifts, but impossible to determine which click that originated from which animal. This is also the case with the amplitude-weighted mean frequency data presented in Fig. 3 B, although it is possible to distinguish three isolated groups of clicks with different amplitude-weighted mean frequencies. The relative energies in Fig. 3 C seem to offer a clearer division between clicks, possibly originating from different animals, in parts of the sequence (313.25-314.10), but are unfortunately ambiguous in other parts (e.g. 314.10-315.25). However, the frequency spectra correlation results in Fig. 4 demonstrates a clear division between correlation coefficient when successive clicks correlate well (squares) and when successive clicks do not correlate well (red crosses).

By combining the data in Fig. 3 C and D into Fig. 4 it is clear that the correlation result and the information that the click energies supply, do agree well. Where the differences in energy between clicks are large, we also find a low correlation between the clicks, indicating that they originate from different animals. Using this information it is possible to distinguish two overlapping click trains as originating from two different animals. The result presented in Fig. 5 shows how all click properties from the individual animals can be followed throughout the sequence and analyzed as separate click trains. However, there are two clicks that correlate poorly with the other clicks in the sequence. These clicks are marked with unfilled circles (313.52 and 314.52). They are especially apparent in Fig. 5 C where it is evident that the peak frequency of the second of those clicks differs considerably from the peak frequency of the other clicks. The first one, however, is in practice impossible to distinguish with any of the described methods, except the frequency spectra correlation method. It is concluded that these clicks either originate from a third animal and possibly even a fourth animal.

The frequency correlation method can be implemented as an automatic algorithm for separating overlapping click trains. However, it does not take into account how many dolphins that may have echolocated simultaneously. Therefore, combining results from this with the energy data and peak and/or amplitude-weighted mean frequencies would offer a robust and reliable semi-automatic method for analysing sequences where more than two dolphins have been acoustically active at the same time.

Conclusion

The described measurement method proved to be a useful tool for recording the sonar beam cross-sections from dolphins spontaneously echolocating at sonar targets while swimming freely in large groups. It can be concluded that the frequency spectra correlation results facilitates detailed studies of multiple click parameters on individual dolphins even though they swim freely in large groups and at times echolocate concurrently at the same sonar target. The peak frequencies, the amplitude-weighted mean frequencies and the relative energies did not single-handedly bring unambiguous results for the separation of overlapping clicks but were meaningful in combination with the frequency spectra correlation results. It is suggested that this method of separating overlapping click trains is especially useful in scenarios where dolphins that echolocate concurrently at the same sonar target also swims closely together so that conventional source positioning algorithms can not resolve the separate positions of the two individuals. These measurement and data analysis methods are suggested to be useful in future studies on object investigation and echolocation behaviour in individual dolphins, swimming freely in large groups.
References


Structured Session 9

Detection/Classification of Underwater Targets

Organizer: John Fawcett
Abstract: The multi-mode pipe projector (MMPP) is a small, wideband transducer developed at DRDC Atlantic. Its large bandwidth, approximately three and one-half octaves, makes it ideal for examining the use of spectral or temporal scattering features in the acoustic detection and/or classification of targets. In this paper, the projector and its measured characteristics are briefly described. The measured output of the projector for a specified, impulsive input waveform can be used to compensate input waveforms to produce desired output waveforms. A series of experiments in the DRDC Atlantic acoustic calibration tank are described. Various targets, including dummy limpet mines, were placed upon a circular fibreglass or aluminum disc. The discs were then rotated below a projector/receiver pair and the echos recorded for a large set of projector pings. The disc provided a background echo. One of the objectives of the experiment was to discriminate the objects’ echos from the background and from one other. In one sequence of rotations, 3 small objects of identical shape but with different interiors were considered. The data set also provided an interesting example of synthetic aperture beamforming where the rotational motion of the disc served to form a virtual aperture.

Keywords: wideband, detection, classification, mines
1. INTRODUCTION

The ability to discriminate between geometrically similar objects on the basis of their response (temporally, spectrally, time-frequency, aspect dependence) to a wideband incident sonar pulse is a subject of much interest [1]. For example, a rock and a mine may have similar geometrical dimensions and be difficult to distinguish solely on the basis of a high frequency sidescan sonar image. However, their wideband structural response may be quite different. At DRDC Atlantic, a new type of efficient, inexpensive, wideband projector, the Multi-mode Pipe Projector (MMPP) has been developed [2-4]. We will first discuss this projector and its characteristics. We will then describe a set of experiments that were carried out in the DRDC Atlantic acoustic calibration tank using the MMPP as the acoustic source.

These experiments were designed to simplistically simulate a limpet mine scenario. The concept was that in addition to a background sonar response from a hull, a pier, etc. there could also be the responses due to small attached objects. Thus, one could envision scanning a hull or a pier with a wideband sonar. Small objects could be detected/classified on the basis of their echo time series or spectrum. This is not necessarily meant to be a stand-alone system but might be used to supplement, for example, a higher frequency imaging-type sonar. In order to carry out the experiments in a very controlled manner, the projector and a hydrophone were fixed, and the targets were mounted on a disc which was rotated below the projector and the time history of the pings was recorded. This was done in such a way that the plate response would be recorded for much of the time with the appearance of the target response as it swept underneath the projector. We will consider the detection/classification problem for these recorded time series. Various sets of targets, plates, and incident pulses were considered.

As will be seen, the objects manifest themselves in the recorded time series both before and after they are directly below the projector due to the projector and receiver’s beampatterns. The rotational history of the time series provided an interesting example of an application of synthetic aperture beamforming. In this case, assuming we know the motion of the rotating plate underneath the projector, we can beamform exploiting this information. An example of this processing is shown.

2. THE MMPP PROJECTOR

In Fig.1a we show the family of MMPP [2] projectors developed at DRDC Atlantic and in Fig.1b we show the small high-frequency MMPP used in the experiments reported in this paper. The MMPP produces a wideband acoustic output by aligning multiple resonances across a wide frequency band. At low frequencies, longitudinal cavity resonances predominate and at higher frequencies, the piezoceramic drive motor and endcap wall breathing resonances become dominant. This design was optimized by using finite element analysis. The advantages of the MMPP design are that it is compact and low cost. It has a scalable design so that it can be tailored for various frequency ranges. It is not sensitive to depth. This projector has been used in a variety of communication, propagation, and scattering studies ([3, 4]). The Transmitting Voltage Response (TVR) of this MMPP is shown in Fig.2 for the slot direction (that is, aligned with the mid-projector gap or slot) and along the axis of the projector (the endcap direction). It is the endcap direction which we will be using in the experiments of this paper. In the geometry of our experiments the endcap will be facing directly down upon the disc and it is expected that the echo will consist mostly of
specular energy and some off-axis energy as the target approaches the projector. As can be seen from the TVR curves of Fig.2, the output energy is fairly flat as a function of frequency although there are some relative nulls. It should be noted that the output levels are shown per 1 volt; for example, for 100 volts applied to the projector the output levels should be multiplied by 40 dB.

![Endcap face](image)

**Fig.1: The “family” of MMPP projectors with the small, high-frequency MMPP used for these experiments shown in detail**

![TVR curves](image)

**Fig.2: The TVR curves for high-frequency MMPP for: (left) slot direction and (right) endfire direction. The curves for 4 different high frequency MMPPs are shown with a Finite Element prediction.**

The frequency response of the projector must be accounted for if a particular output pulse is desired. We do this by a frequency spectrum compensation method: we first transmit an input waveform corresponding to a flat input spectrum over the band \([f_1, f_2]\) kHz, we then extract the portion of the received time series corresponding to the direct arrival and compute the received spectrum (amplitude, and phase), \(T(f)\). The desired output frequency spectrum is then divided by this weighting (with a scalar added to the amplitude to prevent division by zero) and the resulting spectrum is inverse Fourier-transformed to produce the input waveform.
In Fig.3, we show the results obtained from using a combined high- and low-frequency MMPP (from a different MMPP experiment) with the recorded output spectrum 1-73 kHz and the corresponding short pulse. As can be seen, a very flat output spectrum has been obtained over a very wide bandwidth.

Fig.3: The output impulse 1-73 kHz and its spectrum after compensation using a combined high and low-frequency MMPP

3. EXPERIMENTAL SETUP AND RESULTS

The experiments with the dummy limpet mines on the fibreglass and aluminium discs were carried out in the DRDC Atlantic acoustic calibration tank. Three slightly different configurations were used, one of which is shown in Fig.4. In all cases, the MMPP and hydrophone were inline with the hydrophone offset closer to the plate, except for a bistatic configuration where the hydrophone was also displaced horizontally. We used separations of 40, 60, and 70 cm between the hydrophone and the circular disc. The discs, 5.91mm thick aluminium or fibreglass, were 1m in diameter. Some of the targets considered are shown below in Fig.5. They are dummy shapes and are not realistic in terms of their interiors or composition. The 3 small shapes of the bottom of Fig.5 were constructed from an epoxy and are identical in size and shape but have different internal structures: (1) just the epoxy compound (2) with internal ball bearings and (3) with a small internal aluminium plate. The circular plate was rotated at a constant speed (except for a small time interval near the startup), with the projector pinging at approximately 5 pings/ second. The echo time series were sampled at 250 kHz. A set of ping time series were computed for a full rotation of the disc. Different rotation sequences were recorded for the different discs and target arrangements.
As the circular disc is rotated, the predominant acoustic echo is that from the plate (including any modal response of the fluid-loaded circular disc to the incident energy). Example ping time-series sequences are shown in Fig.6 for two of the targets, the polymer-bodied and right-cylinder of Fig.5. Scattered energy is evident even when the targets are not directly under the projector. The [8 90] kHz (these numbers denote the frequency band of the pulse) incident pulse was non-compensated, that is, the spectrum of the incident pulse includes the projector’s TVR frequency-dependence. The [17 57] kHz pulse was compensated resulting in a very short incident pulse. The time history is shown for a small interval, 0.4 msec, near the plate reflection which is more-or-less continuous throughout the sequence of pings. As shown, the reflection from the plate is not flat across the time history. This is because there was a slight tilt of the rotator pole and plate. The echo from the polymer-body (left) and the right-cylinder are evident. In the case, of the polymer-body, the interior of the mine shape is simply air, and the echo, for the shape directly under the projector is basically a negative version of the incident pulse. One can look at the echos and their spectra in detail and construct simple features which allow for the detection and classification of the echos. In Fig.7, we show the time series history for the 3 identical shapes (bottom of Fig.5) for 3 different incident pulses (compensated [17 57], [9 103], and [50 110] kHz). The leftmost target is the object with an interior aluminium plate, the second is just the epoxy compound and the third contains the ball bearings. The object with the aluminum plate clearly seems to have the strongest return. The second time-series sequence for a compensated [9 103] kHz incident pulse produced the most visually distinguishable echos. In Fig.7b we show the distribution of the first 3 of the 4 computed feature values for 100-point echo time series from each of the three...
Fig. 5: The top picture shows 3 dummy mineshapes, the bottom picture shows 3 identical shapes with different internal composition.

Fig. 6: Some example ping/time series histories for the polymer-bodied and right-cylinder shapes of Fig. 5. These examples are for the fibreglass plate and a variety of incident pulses. The frequency band is indicated above each subfigure.
targets (green (compound only), yellow (compound and ball bearings), and cyan (compound with aluminum plate)) and the plate blue. The four features are: (1) the ratio of the mean spectral levels in the bands [62.5 107.5] and [10. 60] kHz, (2) the variance divided by the mean squared of the 100-point echo time series, (3) the maximum absolute value of the envelope of the cross-correlation between the normalized echo and a reference plate echo (ping 2050) and (4) the sum of the squared spectral amplitudes divided by the mean amplitude squared over the interval [10 107.5] kHz. It can be seen that although the features do not form tight clusters, the regions of the 3 targets’ and the plate echoes are easily distinguishable. In Fig.7c we show the results of using a random 12.5% of each class’s echos for training and the result of a nearest neighbour classifier. As can be seen, the classification results are very good.

Thus far, we have considered only the echos as received at the omni-directional hydrophone. There is no physical array to use for beamforming. However, we know the approximate motion of the plate underneath the hydrophone and thus we can predict the ping/time trajectory for a point scatterer on the plate (at the horizontal range of 0.25m which lies underneath the hydrophone). Summing along these trajectories, we can perform a synthetic aperture beamforming. In Fig.8, we show the original ping/time series, a time-shifted version (to flatten the plate response) and the beamformed response for the compensated [17 57] kHz pulse and the polymer- and right-cylinder targets. As can be seen in the beamformed image, the targets appear much more spatially compact.

Fig.7: (a) the ping/time series for three incident pulses described in text (b) distribution of 3 feature values (yellow-class not visible in black and white) and (c) the resulting nearest neighbour classification for the compensated [9 103]kHz incident pulse
4. CONCLUSIONS

We have briefly described the MMPP projector which efficiently yields a very wideband output. The detailed spectral information of the echos produced by such a wideband system can be used, perhaps in conjunction with other sensors, to discriminate objects of interest from a general background. We continue to explore the use of this wideband technology in a variety of applications.

REFERENCES


AN AUTOMATED METHOD FOR CHANGE DETECTION IN AREAS OF HIGH CLUTTER DENSITY USING SONAR IMAGERY

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Abstract: Change detection is the process by which objects are detected by comparing current acoustic measurements with historical ones. This method is often the only viable option of detecting targets when the size and shape of the object is unknown, or when the clutter density is very high, causing a prohibitive number of false alarms. This paper presents a method and results for automatically performing change detection in environments with a high number of false targets, such as a port or harbour. Using an unmanned system, data was collected with a high-frequency sidescan sonar on two separate days in the winter of 2008 in a port. Target signatures are simulated at specific locations using ray-tracing and fused with the real data in order to generate controlled test cases, taking into account sonar orientation and navigation error. Then, geo-located images are automatically coregistered to each other in order to compensate for errors in positioning between surveys. Four methods are then compared for performing the change detection: the Kolmogorov-Smirnov test statistic, the Bray-Curtis distance, the relative entropy and a traditional approach using detection and association between contacts. Results are shown for a variety of situations, and detection and false alarm rates are discussed.

Keywords: Sidescan sonar, change detection, ports and harbours
1. INTRODUCTION

Change detection is the process by which objects are detected by comparing current acoustic measurements with historical ones. For high-frequency imaging sonars such as those used for mine countermeasures (MCM) and route survey, change detection is often the most practical method for detecting objects in cluttered environments such as ports and harbours. The reason is that in such areas there are typically a prohibitive number of target-like signatures which cannot be easily dismissed as non-targets. In order to avoid repeatedly prosecuting false contacts, it is important to be able to automatically dismiss targets which are historical. Change detection is the most common tactic for route survey operations, where maintenance of a historical database is possible.

Change detection has long been a topic of research in medical imaging, video and remote sensing [1]. There are fewer examples of systems using imaging sonar. Initially, operator aids were developed [2]; these relied on a human operator to perform the complex image matching. As the amount of data increased, such systems were automated, typically using automatic target recognition algorithms in order to detect targets from both surveys and performing some form of distance-based association to compare the surveys [3][4]. Some more sophisticated methods for associating based on features in order to reduce the probability of wrongly associating objects have also been developed [5] as well as some matching of the surrounding area to provide context and improve matches [6][7]. In this paper we examine the performance of change detection methods operating in the image domain which do not make explicit use of detection and association. Such methods require a high positioning accuracy due to the small targets of interest and the difficult, cluttered environment. We present and compare four methods for quantifying changes in the imagery: a statistical measure, one roughly based on texture and another from information theory as well as a traditional detect-and-associate method. After an initial step of normalization and geo-referencing, the sonar images are matched together using an approach without explicit control points. Once candidate changes have been identified, a local co-registration is performed in order to match targets and reduce the number of false alarms. The final change detection is then performed.

2. DATA SET

The data set used in this study was described earlier [8]. The data set was collected with the Canadian Interim Remote Minehunting and Disposal System (IRMDS) which is a semi-submersible drone towing a Klein 5500 sidescan sonar operating at a centre frequency of 455 kHz and a bandwidth of 20 kHz. Two surveys were undertaken roughly one month apart during the winter of 2008 in a port area. A standard mission was undertaken where an initial survey of the inner harbour was performed followed by the survey of a long channel leading out to open sea. The port area is heavily cluttered with a significant number of target-like signatures as well as large areas of complex seabed. The channel area is an established crab fishing area, with many crab traps being deployed and removed by local fishermen. In order to provide some ground truth, target signatures were simulated at deliberate locations which were deemed of specific interest to test the methods described here; this paper will concentrate on a particular subset of those test cases. The performance of the methods will be measured as a function of target type. Another factor which will be examined is the
resolution of the resulting georeferenced images. In general, the resolution of this type of (real aperture) sensor is not equal in range and azimuth, with the azimuth resolution typically being the limiting factor. For the sonar used in this study, the range resolution is about 3-5 times better than in azimuth. Therefore, two resolutions of the geocoded image are explored: one in which the better resolution is decimated to the lower resolution to create square pixels, and another in which the poorer resolution is interpolated to the higher one, again creating square pixels.

3. PROCESSING

3.1. Normalizing, georeferencing and target simulation

The images are first normalized and georeferenced. The normalization ensures nearly constant amplitude with range. While this is a standard processing step, it is especially important for the proposed method in order to remove potential false alarms due simply to ensonification at different ranges. Next, the data is georeferenced onto a grid and a flat bottom is assumed. Data samples falling into the same grid cell are averaged.

As mentioned above, in order to provide ground-truth targets, signatures are simulated and fused into the sonar imagery (see [8] for a description of the method). Three target types are capable of being simulated – a large cylindrical target and two smaller targets: an isotropic truncated cone and a more complex wedge shape.

3.2. Spatial co-registration

The next step is to remove some of the uncertainty due to positioning errors which is inherent when positioning this sensor underwater. This is accomplished by globally co-registering two images which contain overlap [9]. The typical approach is to first find features or targets of interest in both images and then to compute the 2D transformation required to match the features (see, e.g [7]). Unfortunately, such an approach can be susceptible to degraded performance when there are large areas with few or no interesting features to use for matching or when the position of the features are displaced from one survey due to the image distortions. Of course, there is always a risk of incorrectly associating features which is especially true in cases where many similar signatures are grouped together, such as in a cluttered port. The algorithm used to match the images considers differences over the entire overlapping area. Such differences are minimized following an iterative process in which local image correlation measures are used. When such correlation measures reach a maximum, scans from both surveys are considered to be co-registered.

3.3. Change detection

Several techniques are then considered to compute the degree to which two images are different and considered to have changed. All methods compute some function $\Delta(I_0, I_1) > \tau$ which quantifies the degree of dissimilarity between two images $I_0$ and $I_1$. A simple example is the absolute difference $\Delta(I_0, I_1) = |I_0 - I_1|$. If the value of the function exceeds some
threshold \( \tau \), then a (change) detection is called. Because of the multiplicative noise which is present in high-frequency, narrowband sonar images, difference operators tend to perform poorly. The first three methods suggested here are based on the direct comparison between two co-registered images \( I_0 \) and \( I_1 \) within two \( m \times n \) windows centred on the same point \((x,y)\) which represents the same geographic location in the two images. The last technique is a more traditional detection – association technique.

### 3.3.1. Kolmogorov-Smirnoff test statistic

The two-sample Kolmogorov-Smirnoff test statistic [10] is typically used in a non-parametric hypothesis test which provides a method of testing whether two samples are from the same probability distribution. The statistic is computed using:

\[
\xi(i, j) = \sup \left| F(I_0(i, j)) - F(I_1(i, j)) \right|,
\]

where \( F(I_v(i,j)) \) is the cumulative distribution of the estimated probability distribution of the pixels in image \( v \), estimated using the histograms of the pixels located in an \( m \times n \) window centred on the pixel \((i,j)\). The statistic is a form of distance between the two empirical estimates of the distributions of the pixels. Since \( \xi \) is computed using histograms and not by comparing analogous pixels in two images, it should be less sensitive to slight misalignments of the two images. In the examples given below the values of the window dimension \( m,n = 16 \).

### 3.3.2. The Bray-Curtis distance

The Bray-Curtis distance [11] between two images \( I_0 \) and \( I_1 \) is defined as:

\[
\beta_{01} = \frac{\sum_{i=1}^{n} \sum_{j=1}^{m} |I_0(i, j) - I_1(i, j)|}{\sum_{i=1}^{n} \sum_{j=1}^{m} (I_0(i, j) + I_1(i, j))},
\]

where \( n \) is the number of pixels in the images and \( I_v(i,j) \) is the location of the image at position \((i,j)\). It has been used previously as a similarity metric between two images; in the case of the sonar imagery, it is meant to provide a measure of texture divergence for the images; since it is computed pixel-wise, it will be sensitive to even slight misalignments between the two images. It is always between 0 and 1 for positive values of \( I \) and again is calculated for each image location within a \( 16 \times 16 \) window.

### 3.3.3. Relative entropy

The relative entropy (or Kullback-Leibler divergence) [12] between two distributions \( p \) and \( q \) over \( x \) is computed as:

\[
D_q(p \| q) = \sum_x p(x) \log \frac{p(x)}{q(x)}.
\]

Here, \( p(x) \) and \( q(x) \) are the probability distributions of the pixel amplitudes within an \( m \times n \) window of \( I_0 \) an \( I_1 \) respectively, centred at location \((i,j)\). \( D_q(p \| q) \) is another measure of the distance between two distributions and because it uses histograms, it should not be sensitive
to slight misalignments in the co-registered images. As with the other methods, it is calculated for each image location using a $16 \times 16$ window.

### 3.3.4. Detection and association

The final method in this comparison is a more traditional one which explicitly detects features using an automatic detector which performs a statistical test to find echo-shadow pairs of the size and shape of a generic target signature [13] and associates them using some transformation, in this case a simple distance threshold. Detected objects or features without a corresponding detection in the first survey are considered to be new. The detector is applied to the pre-georeferenced images, since it requires information about the sensor telemetry to create suitable target signatures.

### 3.4. Local co-registration

The first three methods, particularly the one based on pixel-wise comparisons, will have some degree of sensitivity to remaining misalignments in the positioning after co-registration. Although the global co-registration method presented above will match the entire images well, local, non-constant misalignments are inevitable due to violations in the simplifying assumptions of the georeferencing process (a flat bottom, a constant speed of sound) and short term changes in the accuracy of the position method (i.e. because of a sudden shift in sensor pitch angle). In order to reduce spurious false alarms due to such misalignments, there is also a local co-registration process which happens once an initial detection has been made. The normalized image cross correlation is used, which at a point $(i, j)$ is defined as:

\[
\gamma(i, j) = \frac{\sum_{xy} |I_0(x, y) - \overline{I_0}||I_i(x-i, y-j) - \overline{I_i}|}{\sqrt{\sum_{xy} |I_0(x, y) - \overline{I_0}|^2 \sum_{xy} |I_i(x-i, y-j) - \overline{I_i}|^2}}. \quad (5)
\]

The peak value of $\gamma$ over a $2m \times 2n$ window around the detection is used as a shift factor to align the two images, and the change detection metric is recomputed. If the value still exceeds some threshold $\tau$ then a detection is called.

### 4. RESULTS AND DISCUSSION

Table 1 shows the results for a variety of test cases using the four change detection methods. For each test case, a brief description is given as to the target type and the resolution of the geocoded imagery used in the test. For the detection/association method, the algorithm is run on the pre-geocoded imagery and the positions of those detections are corrected using the offsets provided by the co-registration step, resulting in more robust association for this method. Also given are some factors which cause false alarms such as a ship wake which, obviously, are not replicated between the two surveys. The table shows the number of targets detected out of the total number of targets present. It also shows the number of changes detected; this is the total number of changes which were deemed to be due to new objects which were not the ones simulated in the test set but could not be considered
false alarms. A detected change was considered to be a false alarm when it could not be attributed to a new object or feature which had a potential target-like signature; the total number is shown. Some sonar images of the first and second surveys are shown in Figure 2. The thresholds used for test cases 3 and 4 were 20% higher than the ones used in the other cases.

<table>
<thead>
<tr>
<th>Test Case</th>
<th>KS Test Statistic</th>
<th>Bray-Curtis Distance</th>
<th>Relative Entropy</th>
<th>Detection and Association</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Test Case 1</strong>: Cylinder within clutter, normal resolution</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>1</td>
<td>2</td>
<td>0</td>
<td>2</td>
</tr>
<tr>
<td>Changes detected</td>
<td>3</td>
<td>17</td>
<td>0</td>
<td>6</td>
</tr>
<tr>
<td>False alarms</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Test Case 1</strong>: Wedge within clutter, high resolution</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>2</td>
<td>2</td>
<td>1</td>
<td>2</td>
</tr>
<tr>
<td>Changes detected</td>
<td>25</td>
<td>27</td>
<td>4</td>
<td>6</td>
</tr>
<tr>
<td>False alarms</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Test Case 2</strong>: Three cones near objects, high resolution, ship wake</td>
<td>3/3</td>
<td>3/3</td>
<td>3/3</td>
<td>3/3</td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>2</td>
</tr>
<tr>
<td>Changes detected</td>
<td>24</td>
<td>3</td>
<td>44</td>
<td>2</td>
</tr>
<tr>
<td>False alarms</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Test Case 2</strong>: Cylinder near clutter, normal resolution, ship wake</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>2</td>
</tr>
<tr>
<td>Changes detected</td>
<td>2</td>
<td>9</td>
<td>17</td>
<td>0</td>
</tr>
<tr>
<td>False alarms</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Test Case 3</strong>: Cylinders at varying positions versus lateral range, normal resolution</td>
<td>4/4</td>
<td>3/4</td>
<td>4/4</td>
<td>4/4</td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>0</td>
<td>2</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>Changes detected</td>
<td>0</td>
<td>1</td>
<td>0</td>
<td>8</td>
</tr>
<tr>
<td>False alarms</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Test Case 4</strong>: Cylinder within clutter, normal resolution, propagation effects</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
<td>1/1</td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>0</td>
<td>1</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>Changes detected</td>
<td>32</td>
<td>15</td>
<td>45</td>
<td>11</td>
</tr>
<tr>
<td>False alarms</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Total</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Target(s) detected</td>
<td>11/11</td>
<td>10/11</td>
<td>11/11</td>
<td>11/11</td>
</tr>
<tr>
<td>Changes detected</td>
<td>4</td>
<td>8</td>
<td>2</td>
<td>10</td>
</tr>
<tr>
<td>False alarms</td>
<td>86</td>
<td>72</td>
<td>110</td>
<td>33</td>
</tr>
</tbody>
</table>

Table 1 Results for change detection for 6 examples.

From the results shown in Table 1, the following observations are made:

- The detection/association method shows the most consistent performance out of all the methods. The co-registration step essentially eliminates most of the uncertainty with matching structures of control points and a small (here 1.5m) and constant distance threshold for association can be used. In addition, this method allows one to categorise changes as a new object or one which has been removed. Only new objects are shown in Table 1, resulting in much less false alarms. However, the use of a detection method requires some a priori knowledge of the target signature in order to be able to detect an object in the first place. Although the detector is very generic and a low threshold was used in order to simulate the case where one does not precisely know the size/shape of the target, it is nonetheless less generic than the other methods.

- When the background is stable, the techniques based on pixel statistics (the KS test and the relative entropy) performed well. However, being based on statistics, they were very sensitive to changes in the background such as those caused by the wake of another ship crossing the path of the sonar, or a downward refracting
environment which causes irregular patterns in the sonar images. The false alarms of the relative entropy were generally caused by such occurrences.

- The micro-registration step reduced the number of false alarms and was absolutely necessary when using the Bray-Curtis distance which compares images in a pixel-by-pixel way. In general, the Bray-Curtis distance was not affected by different yet random effects such as ship wake, and in the case of the test case 2.1 worked very well; however this metric will more likely raise false alarms due to real objects which were slightly different between the two surveys.

- As was originally noted in [9], the small objects (the cone and the wedge) required high-resolution geocoded imagery to be used to detect the targets with the first three methods. Since the detection / association method works on pre geocoded images, the performance is the same for high and normal resolution cases.

- For the first three techniques, the detections were generally the same while the false alarms were generally uncorrelated. A simple fusion rule such as “and”-ing or majority voting will, for these examples, greatly reduce the false alarm rate while maintaining a high detection rate; however, the computational requirements will increase, therefore needing some optimization for use in an operational system.

![Figure 1 Example images. The first survey is shown on top and the co-registered image from the second survey is shown on the bottom.](image)

5. CONCLUSIONS

Several methods for automatic change detection in sonar imagery were presented. The use of the methods is made possible by a step which globally co-registers sonar images; the images can then be compared for changes with much less ambiguities due to positioning inaccuracy. Three image-based methods and a traditional control point matching method are compared using real data with simulated target signatures. In general, the performance of the detection / association method was more consistent but required some a priori knowledge of
the target signature, while the statistical methods performed well when the collected data was of high-quality and reported many more false alarms when effects such as ship wakes and poor environmental conditions were present. Of course, results would be different by simply redefining the false alarm – the ship wakes and other effects are seen visually as changes by the human eye, however the definition used here stated that anything that was not due to the actual presence of a new object on the seafloor was a false alarm. Future work will concentrate on better statistical characterisation of the performance and robustness of the change detection methods, using a larger dataset with real targets.

REFERENCES

A SYSTEM FOR AUTOMATIC DETECTION AND CLASSIFICATION FOR A MINE COUNTERMEASURE AUV

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Abstract: Autonomous underwater vehicles (AUVs) gain increasingly importance in civil and military applications. Especially in the area of reconnaissance and mine countermeasures (MCM) it is expected that AUVs will be integral part of seagoing vessels. A major disadvantage of a typical MCM AUV operation is the time delay between survey and evaluation of collected side scan sonar data which typically doubles the overall mission duration.

In order to exploit the full potential of MCM AUV missions, on-board data processing and the capability of online automatic detection and classification (ADAC) of mine-like objects is necessary. Once the sonar data has been processed, a reduced data set containing the classification results can be transmitted to the surface platform and are available immediately after the mission. It also opens up the possibility of online mission re-planning based on actual results and the prior mission plan.

In this paper, the ongoing development of an operational ADAC system is presented together with first results.

Keywords: automatic, detection, classification, MCM, AUV
1. INTRODUCTION

Autonomous underwater vehicles (AUVs) gain increasingly importance in civil and military applications. Equipped with a variety of sensors these vessels are used e.g. for search of natural resources, pipeline inspections, scientific missions and mine warfare applications such as reconnaissance and mine countermeasures (MCM). In particular the operations of dedicated MCM AUVs have strong advantages, e.g. an AUV keeps personnel from potentially dangerous zones or as a containerized AUV it can be deployed fast from nearly every platform allowing first MCM reconnaissance operations.

However, the time delay between mission accomplishment and evaluation of recorded sonar data is regarded as a major drawback of a usual MCM AUV operation. The main reason for this is the amount of collected data and hence the corresponding time for data transmission and evaluation. To avoid this disadvantage, onboard processing and the capability of real-time automatic detection and classification (ADAC) of mine-like objects is necessary. After ADAC processing the raw sonar data are reduced considerably to a set of relevant contact data, e.g. positions of mine-like objects, which can be transmitted to the surface platform and are available immediately. Furthermore, based on actual ADAC results and the prior mission plan, dynamically mission re-planning is possible.

In 2006 ATLAS initiated an ADAC project with the objective to develop an operational basic ADAC system suitable for high frequency side scan sonar images. It was part of the MCM AUV project of ATLAS and the German Federal Office of Defence Technology and Procurement (BWB), and it provides core functionalities of the MCM AUV operational sequence, covering mission planning, mission execution, transmission and display of ADAC results on the surface station. In the first stage of the development the emphasis was the operational chain i.e. planning, execution and display of results on the user interface rather than the performance of the ADAC processing algorithm itself. However, as discussed later on, this ADAC processing algorithm has a modular design in order to allow easy integration of extensions and improvements subject of future developments. In the following the ongoing development of this ADAC system with focus on ADAC processing algorithms is presented.

This paper is organized as follows: chapter 2 gives a short description of the ADAC system concept. In chapter 3 the ADAC processing modules are presented and subsequent in chapter 4 first results are shown. The paper ends with conclusions given in chapter 5.

2. ADAC PROCESSING ALGORITHM CONCEPT

The concept of the ADAC processing algorithm was determined by two items: firstly by the long-term objective, starting from a basic system, to gradually develop a reliable and high-performance system. Secondly by the fact, that the environmental conditions, bottom (seabed) type and the specific sonar system have impact on the appearance of objects in sonar images. Hence, it is not expected that there will be a single “best” ADAC processing algorithm that can cope with all these varying conditions but rather a combination of different algorithms may achieve the desired performance.
For the above reasons a multi-stage approach is chosen, each stage corresponding to distinct ADAC processing module, ensuring high flexibility in terms of easy exchangeability of individual modules to form a single system with good performance. In Fig. 1 a) the different components of such an ADAC processing algorithm is diagrammatically shown. It consists of the following four processing steps:

**Sonar image generation and pre-processing**: here, the raw sonar data are collected over a fixed time period and arranged into a sonar image. Afterward the image is enhanced by means of normalization and/or de-noising methods.

**Detection**: in this module the pre-processed image is searched for mine like objects often with a subsequent (weak) filtering of detected objects in order to eliminate obvious false alarms.

**Feature extraction**: the image sections of the remaining detected objects serve as input for the feature extraction step where relevant image and/or object features are extracted. These features represent condensed image and object information and form the basis for discrimination of mines and non mine-like objects.

**Classification**: the last step in the ADAC processing chain is the classification of the detected objects. At this stage, based on the extracted features each object is assigned to the class of mines (possibly even to a certain type of mine) or to the class of non mine-like objects. The resulting contact data, including positions, classifications results and other relevant information are then stored for later transfer to the surface station.

Moreover, the distinct processing modules open up the possibility of algorithm fusion at stage level, as indicated in Fig. 1 b) for the detection step. As reported in [1], this is particularly useful in the detection and classification stage since it gives the prospect of reducing false alarm rates and false classifications significantly.

Another way of reducing the false classifications is to fuse the results of multiple ADAC processing algorithms [2], as schematically shown in Fig. 1 c). Here, various independent algorithms process the same sonar data and only the classification results are combined to...
yield better overall classification performance. Both types of fusion are part of the ADAC system concept.

3. ADAC PROCESSING MODULS

The choice and design of appropriate processing modules for an ADAC processing system is determined by the specific target platform, a dedicated MCM AUV. Such vehicles are characterized by high sensor data rates, limited processing abilities and power resources. Therefore these modules must be computationally efficient as well as fast and robust in order to ensure reliable autonomous real-time operation of the ADAC system.

In a basic version only one single algorithm for each stage is used and the ADAC system is laid-out for detection and classification of proud mines with high signal to noise ratio and simple object structures (cylinder mine) on a flat bottom. In the following sections the algorithms in the different modules are described.

3.1. Image generation and pre-processing module

This module collects the raw sonar data and generates a geo-referenced sonar image. The image is then normalized in order to improve the contrast. Further image enhancements can be performed, such as histogram stretching and/or applying a median filter to remove noise. The described steps are exemplarily shown in Fig. 2 for a high frequency side looking sonar image.

3.2. Detection module

The automatic detection of mine-like objects is performed by segmentation of the sonar image in highlight, shadow and background regions and by finding corresponding highlight/shadow structures that can be related to a mine-like object.

Here, we use a segmentation method based on Markov Random Fields (MRFs). In this approach it is assumed that the observed image is a noisy version of the true data set with label fields corresponding to the classes highlight, shadow and sea bottom / background. The aim of the segmentation is to reconstruct the true label field by associating a class-label to each image pixel. The advantage of MRFs is the capability to incorporate prior information about the local spatial structure of the true image. Several MRF models have been proposed in the literature (e. g. [3] [4] [5]). They vary mainly in modelling the prior information about the spatial structure, the models of the noise distributions of the data-likelihoods and the approximation of the maximum a-posteriori estimation.

However, the main problem of automatic image segmentation is that the prior models are subject to variations of the data-acquisition conditions, of the sea bottom structure and on specific pre-processing of the sonar data. To obtain a robust segmentation solution, the Iterative Conditional Mode (ICM) algorithm has been chosen [6]. Here, a quadratic MRF-neighbourhood with equal weightings was used assuming a normal distribution for all classes in the (pre-processed) image. The initialization parameters have been chosen automatically ensuring that all relevant objects are detected (high detection probability), accepting at this stage a still relatively high number of false alarms.
Once the sonar image is decomposed in highlight, shadow and background zones the detected object filtered by means of pre-selection to reduce false alarms. In a first step, geometrical characteristics of the expected object are used to reduce clutter. Then, in a second step, “object-like” highlight-shadow structures are identified and stored for further processing. Together with the sonar images, these results serve as input data to the next ADAC steps.

Fig. 3 shows the detection result of the pre-processed sonar image of Fig. 2. In addition to the mine object a large number of false alarms can be observed (Fig. 3 left). Subsequent pre-selection leads to a significant false alarms reduction. However, the relevant object is preserved (Fig. 3 right).

3.3. Feature extraction module

Texture features are used, since previous investigations [7] indicate that these feature types are appropriate for the mine classification problem. Texture features are based either on
statistical or structural texture models. However, for sonar images the following statistical
texture models appear to be suitable [7]: co-occurrence matrix [8], grey level run length
matrix [9], neighbouring grey level dependence matrix [10] and the neighbourhood grey tone
difference matrix [11]. All texture matrices are functions of two parameters and for every
matrix four to five different texture features can be extracted. Many of the features describe
the same or comparable image characteristics and therefore may not be independent.
Moreover, for a real-time system running in an AUV, it is essential to select as much features
as necessary in order to prevent loss of information and simultaneously as few as possible in
order to limit the training and classification expenses. Thus, a reduction of the multitude of
above mentioned features was performed. The aim hereby was to eliminate correlated
features and simultaneously to determine the most significant features for discrimination of
mines and non mine-like objects. After comprehensive investigations 14 significant texture
features were identified. These relevant features are extracted from image sections (like the
inlay in Fig. 3 right) containing the detected mine-like objects. A principal component
analysis is applied and the first three components are used for classification, so that the
dimensionality of the feature space is further reduced.

3.4. Classification module

The aim of this module is to assign every detected object to the class of mines or non
mine-like objects based on the extracted features. This classification is performed using fuzzy
clustering. Clustering methods are in general fast and applicable even for limited pool of
training data, which is the case here. The fuzzy variants of these methods have also the
advantage that the result of the classification process is a number indicating the membership
(or confidence level) to the mine class. This membership is preferred over a hard two-class
classification since in sonar images the appearance of an object is not always apparent so that
a unique assignment to one class is not always possible.

Following classifiers have been analyzed for ADAC in real time application: fuzzy c-
means (FCM) [12], fuzzy k-nearest neighbour (FKNN) [13], fuzzy nearest prototype (FNP)
[13] and a modification of the last classifier, termed fuzzy single prototype algorithm (FSP).
The resulting membership of the FCM, FKNN and FNP classification is a function of the
weighted ratio of the (euclidean) distances of the feature vector and cluster centres
corresponding to a particular class. For the two-class classification case considered here the
cluster centres represent the feature vectors of mine and non mine-like objects. Especially for
the latter we faced the problem to represent a variety of non mine-like objects with one single
feature vector, which is not possible. To avoid this problem we developed the fuzzy single
prototype classifier, where only the cluster centre for the mine objects is used. The
confidence level is computed by an s-shaped membership function whose parameters are
adjusted according to the training results. As discussed in the next section, all classifiers show
similar performance with slight advantage for the FSP, which why FSP is actually chosen for
the AUV ADAC processing algorithm.

4. FIRST RESULTS

At the time this paper is written the development and integration of above described basic
ADAC processing algorithm onto an AUV was completed but it is still in the testing phase.
Therefore the results presented are from pre-processed high frequency side looking sonar data
of former trials of the ATLAS AUV SeaOtter MkI. The dataset includes in total 10 cylinder
mines and 13 non mine-like objects. The size of each processed sonar image is 1000 x 500 pixels, with a resolution of about 0.1 m x 0.1 m. The computation time for a complete ADAC cycle (without sonar image generation) is about 2s on a PC with 3 GHz processor.

The two prototypes for the FNP classifier were defined as the mean of typical mine feature vectors and the mean of typical non mine-like object feature vectors respectively. For the prototype of the FSP classifier also the mean of typical mine feature vectors have been used. For the FKNN classifier a 4 element trainings data set consisting of 2 mine feature sets and 2 non mine-like feature sets was used. The number of neighbours was three. An object is classified as a mine if the membership to the mine class exceeds 75%. An overview of the classification results using different classifiers is given in Table 1.

<table>
<thead>
<tr>
<th>Classifier</th>
<th>Mine classified as mines</th>
<th>Non mines classified as mines</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fuzzy c-means</td>
<td>100% (10/10)</td>
<td>85% (11/13)</td>
</tr>
<tr>
<td>Fuzzy k-nearest neighbour</td>
<td>100% (10/10)</td>
<td>62% (8/13)</td>
</tr>
<tr>
<td>Fuzzy nearest prototype</td>
<td>100% (10/10)</td>
<td>46% (6/13)</td>
</tr>
<tr>
<td>Fuzzy single prototype</td>
<td>100% (10/10)</td>
<td>39% (5/13)</td>
</tr>
</tbody>
</table>

Table 1: Classification results using different classifiers

Our main emphasis to ensure correct classification of all mines is met by all classifiers whereas the FSP method yields the best performance. However, all classification results suffer from high false alarm rates that are at this stage acceptable for the basic system that is subject of continuously improvements. Enhancement of false alarm reduction is expected with data fusion and increasing number of representative sonar data sets.

5. SUMMARY AND CONCLUSIONS

A system for automatic detection and classification (ADAC) for a MCM AUV is presented. The main objectives of this system are to obtain a fast and robust basic ADAC functionality under the conditions of limited resources of the on-board system and especially to ensure fully autonomous operation.

For the ADAC processing algorithms a multi-stage approach is used starting with sonar image generation and pre-processing, going to detection, feature extraction and ending with classification of the detected objects. The capability of data fusion on stage and algorithm level plays an important role in future development steps and is therefore incorporated in the ADAC system concept.

First results show that the prototype system indeed classified all mines correctly but at a high false alarm rate. In the future, when the classification can be based on a large number of representative sonar data we expect to improve the false alarm rate. Additional improvement of the classification results can be expected applying more sophisticated classifiers and data fusion.

6. ACKNOWLEDGEMENTS
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REFERENCES

3D TARGET SHAPE FROM SAS IMAGES BASED ON A DEFORMABLE MESH

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Abstract: The seafloor can nowadays be scanned with side-looking sonar that provides a very high resolution over a large swath, which has proved beneficial for underwater target detection and classification. For systems operated at hundreds of kilohertz, one may obtain centimeter resolution in the range and along-track direction. For these systems, the third dimension, height, is usually resolved much worse. Since the 3D shape is regarded a valuable clue in underwater target classification, it is important to extract height information as best as possible. In this paper a new method for deriving 3D information from non-interferometric SAS images is described. The method is experimentally applied to multi-view reconstruction of calibrated target shapes imaged by the MUSCLE vehicle, and the results are compared to the actual dimensions of the observed objects in order to quantify the reconstruction accuracy. The reconstruction algorithm is a shape-from-shading approach that uses a deformable mesh to preserve surface continuity while enforcing observational constraints. The technique is shown to have an important impact in object classification, both as a stand-alone method and combined with other 3D imaging techniques such as interferometry. Implications to target identification and vehicle autonomy are discussed.

Keywords: Synthetic Aperture Sonar, 3D reconstruction, Shape from Shading.
1. INTRODUCTION

Modern side-looking sonars provide a very high resolution over a large swath, which has proved beneficial for underwater target detection and classification. For systems operated at hundreds of kilohertz, one may obtain centimetre resolution in range and along-track directions. The third dimension, height, is usually resolved much worse, and as a consequence indirect methods—such as measuring shadow length—are frequently used to estimate the height of objects. Nevertheless, since not just the height but in fact the complete 3D shape of an object is an invaluable clue for underwater target classification, it is important to extract as much detailed height information as possible from the existing sonar image data. In this paper we present a new method for deriving 3D information from non-interferometric SAS images that uses a deformable mesh to solve the constraint satisfaction problem posed by a basic shape-from-shading approach.

2. 3D RECONSTRUCTION BASED ON A DEFORMABLE MESH

The underlying idea behind shape-from-shading [1] is that the configuration of a surface can be inferred from observing the way it reflects and scatters light (or sound in the case of sonar). And, in effect, the intensity of the reflected signal when illuminated by a sonar sensor is mostly dependent on surface slopes: the more directly the local normal vector of the surface faces the sonar, the brighter the image pixel for that location will be.

The simplest model for this type of behaviour is Lambert’s law [2], which assumes diffuse scattering and results in the returned intensity \( I \) from surface point \( p \) being proportional to the cosine of the angle \( \theta \) formed by the direction of observation \( r \) and the surface’s normal vector \( N \):

\[
I(x(p), y(p)) \propto r(p) \cdot N(p) = \cos \theta(p)
\]

Where \( x \) and \( y \) are the coordinates of the pixel where surface point \( p \) appears in image \( I \).

More complex models [3] can take into account the type of material forming the surface (its composition, granularity, etc). The inclusion of specular effects is another possibility, and is often implemented as a power function of the basic cosine law.

2.1. Local formulation in polar coordinates

Equation (1) provides a link between the observed intensities in a sonar image and the local orientation of the observed surfaces. To exploit this link in order to derive surface topography, working in polar coordinates has several advantages. The main one is the slopes being constant and independent of local elevation, which speeds up the iterative optimization algorithm. The derivation of the formulas is a bit more involved than when working on Cartesian coordinates (as in [2]), but the consequent simplification of the iterative optimization process more than compensates for it, especially given the associated speed increase and the ease of adapting the algorithm to a GPU implementation [4].

Sonar images are “range images”, with the across-track coordinate \( x \) corresponding to the distance \( r \) from the surface point to the sonar. This corresponds to a cylindrical coordinate...
system centered at the sensor’s position \( \mathbf{o} \) and with the cylindrical axis \( \mathbf{y} \) aligned with the direction of travel of the sonar. Two pixels adjacent in the range direction (Fig. 1) have ranges \( r_i \) and \( r_{i+1}=r_i+dr \), with \( dr \) being the range resolution (which in the case of SAS is constant).

Thus, for every range image pixel two of the three cylindrical coordinates are known (\( y \) and \( r \)), whereas the angular coordinate \( \alpha \) (depression or grazing angle from the sensor) is not. An estimation of this coordinate’s derivative for a given pixel can nevertheless be obtained from the image intensity at the pixel [2].

![Fig.1: Polar formulation of the geometry for the 3D reconstruction problem of a single scan line.](image)

In effect, with our choice of coordinates and assuming Lambert’s imaging model, the intensity at a point \( p_i \) directly depends on the angle \( \theta_i \) formed by a radius \( r \) and the surface normal \( \mathbf{N} \) at the point. The surface tangent at a point \( p_i \) is a 3D vector \((dr, dy, d\alpha)\) with \( dr \) the sonar’s range resolution and where we initially assume \( dy \) is zero. The value of \( d\alpha \) is given by the intersection of a secant line at \( p_i \) (tilted by \( \theta_i \) from the tangent’s direction) with the circle of radius \( r_{i+1}=r_i+dr \).

Computation of \( d\alpha \) is straightforward after noting that it does not depend on \( \alpha_i \). By a change of coordinates \((\alpha=\alpha+\alpha_i, \text{thus setting } \alpha_i \text{ to zero})\) we can use the simplified diagram on Fig.2 to derive the expression for \( d\alpha \).

![Fig.2: A change of coordinates simplifies the derivation of an expression for \( d\alpha \).](image)

Applying the cosine theorem to the triangle formed by \( r, r+dr \) and \( t \), we obtain:
\[ t^2 = r^2 + (r + dr)^2 - 2r(r + dr)\cos(d\alpha) \]  

(2)

Additionally, it is clear from Fig.2 that:

\[ dx = (r + dr)\cos(d\alpha) - r \]  

(3)

\[ dx = t\cos(\gamma) = t\cos\left(\frac{\pi}{2} - \theta\right) = t\sin(\theta) \]  

(4)

Combination of (2), (3) and (4) yields the final expression for the \( d\alpha \):

\[
\cos(d\alpha) = \frac{1}{r + dr} \left( -r\sin^2\theta + r + \sqrt{r^2\sin^4\theta + dr^2\sin^2\theta + 2rdr\sin^2\theta} \right)
\]  

(5)

The \( da_{ij} \) for every image pixel \((i, j)\) can thus be obtained substituting in (5) the value of \( \theta \) that corresponds to intensity \( I(i, j) \) according to the selected illumination model.

**2.2. Derivation of the global solution**

Knowing the altitude of the sonar over the seafloor (from the first-returns in the range image or from navigation data), the absolute values of \( \alpha \) for the image pixels at minimum range \( r_0 \) can be set, therefore allowing for the derivation of the \( \alpha \) coordinate of any pixel by simple integration:

\[
\alpha_j = \alpha(r_j) = \alpha_{o_j} + \int_{r_j}^{r_0} d\alpha_j(r) \approx \alpha_{o_j} + \sum_{k=0}^{i} da_{ij}
\]  

(6)

Unfortunately the \( da \) obtained from Lambert’s diffuse model are just an approximation and the 3D reconstruction resulting by just applying (6) to each image line is extremely noisy. And since the method works line-by-line the reconstructed surface gets jaggier as range increases. Results can be smoothed out by directly enforcing surface continuity \( (\alpha_{ij} - \alpha_{ij-1} \approx 0) \), ensuring that adjacent lines don’t diverge too much from each other—although this has the undesirable effect of also smoothing the details of the reconstructed surfaces.

In order to minimize the unwanted smoothing effects of enforcing continuity, we have weighted it using two simple heuristics that have provided good results:

- Adjacent pixels of similar intensity are likely to belong to the same surface patch. This follows from the natural assumption that edges in the image likely correspond to boundaries between objects in the scene.
- Dark pixels provide less information than bright ones, and in fact nothing is known about completely shadowed pixels.

The two rules can be incorporated in a weighted expression that determines the along-track relation between adjacent pixels:
Where $I_{ij}$ has been used instead of $I(i, j)$ for conciseness and where $f$ is a power factor between 0 and 1. Note that here we assume the image $I$ has been already normalized to the $[0, 1]$ range with 0 corresponding to the shadows (-5dB) and 1 to the highlights (35dB).

With the constraints given by equations (6) and (7) any $\alpha_{ij}$ can be related to its neighbors and a global solution can be found by solving a system of linear equations. Unfortunately the system is very sparse and nearly singular, which makes solving it problematic. A regularization scheme is required, which results in over-smoothed solutions and long computation times.

An alternative approach to finding the global solution is to formulate the reconstructed surface as a deformable elastic mesh (Fig. 3) and pose the constraints as forces that deform it. The mesh can be left to evolve over time until it converges to an equilibrium configuration, which will correspond to the sought reconstruction surface. Note that for simplicity Fig.3 shows a Cartesian mesh, where surface points $p_{ij}$ are only allowed to move in the $z$ direction in order to satisfy the constraints to their neighbors. In our cylindrical coordinates implementation the surface points actually displace in the angular coordinate $\alpha$.

Fig. 3: The constraint satisfaction problem posed as a deformable mesh, with the constraints implemented as forces acting on the mesh nodes.

The resultant force $F_{ij}$ acting on each mesh node consists of two across-track forces ($C_{ij}^-$ and $C_{ij}^+$) and two along-track ones ($L_{ij}^-$ and $L_{ij}^+$):

$$ F_{ij} = C_{ij}^- + C_{ij}^+ + k_L (L_{ij}^- + L_{ij}^+) $$

Where $k_L$ is a constant that gives more or less weight to the smoothing associated to enforcing continuity (we normally set it to 1/32) and where the forces are:

$$ C_{ij}^- = (\alpha_{i-1,j} - \alpha_{ij}) + d\alpha_{i-1,j}; \quad C_{ij}^+ = (\alpha_{i+1,j} - \alpha_{ij}) - d\alpha_{ij} $$

$$ L_{ij}^- = (I_{ij} I_{ij-1} (1 - |I_{ij} - I_{ij-1}|)) (\alpha_{ij-1} - \alpha_{ij}); \quad L_{ij}^+ = (I_{ij} I_{ij+1} (1 - |I_{ij} - I_{ij+1}|)) (\alpha_{ij+1} - \alpha_{ij}) $$

Since we are not really interested in the dynamic behavior of the mesh, once the forces are computed at each iteration of the optimization loop, the positions of the mesh nodes are updated by first order evolution:

$$ \alpha_{ij} = \alpha_{ij} + k_F F_{ij} $$
Where \( k_F \) is a constant that controls the evolution speed (normally set to 0.3). The system is left to evolve until the forces are small enough that an equilibrium configuration can be assumed. The whole algorithm is implemented in a multi-resolution fashion similar to that of [2].

3. RESULTS

The proposed reconstruction method has been applied to SAS images acquired with NURC’s MUSCLE vehicle. One example is shown in Fig.4, where the 3D has been estimated for a set of jars found in the wreck of the roman vessel Dolia.

![Fig.4: Single-view reconstruction results for a set of roman jars found in the Dolia wreck. Left: MUSCLE SAS image. Middle: textured 3D surface. Right: flat shaded 3D.](image)

Application of the method to multi-view reconstruction of underwater targets provides unprecedented shape details that should improve classification and identification performance [5]. Fig. 5 shows the results for the combination of 15 views of a cylindrical exercise target during the Colossus2 trials in Latvia (2008).

Combination of the 3D surfaces obtained from different views is nevertheless not trivial: navigation accuracy is not enough to just merge the geo-referenced reconstructions, areas within the shadow volumes must be discarded, and the reconstructions are only resolved up to a single-valued elevation map that is distorted by the shape of its imaging wave-front [6]. The result shown in Fig. 5 has been obtained by correlation of the different surfaces with the...
shadow areas masked out and weighting elevation values according to their echo strengths, but the method still needs further development for improved robustness and accuracy.

4. DISCUSSION

The proposed technique should have an important impact in object classification, given the amount of detail that it can provide (in principle up to sensor’s pixel resolution). Combination with interferometry should also give more accurate reconstructions.

We are now working on a GPU implementation of the iterative optimization algorithm which we expect should provide a speed increase of several orders of magnitude [4]. This should open the possibility for novel detection and classification applications that were until now unfeasible due to their prohibitive computational costs.

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IMAGE PROCESSING IN SIDE SCAN SONAR IMAGES FOR OBJECT DETECTION AND CLASSIFICATION

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Abstract: This paper presents recent and planned activities in the area of computer aided detection and classification (CAD / CAC) of mine like objects (MLOs) at the FWG with assistance of FU-Berlin and FGAN-FOM. These investigations are intended to support software for the analysis of side scan sonar images by an operator and to contribute to automatic target recognition (ATR) software in case of autonomous UUVs. The object detection and classification process is divided into a couple of processing steps starting with pre-processing and screening for regions of interest over the reduction of false positives and object classification up to the fusion of detection and classification results of different algorithms. A first CAD system was presented 2008 [2] including four different screening methods (statistical analysis, highlight / shadow analysis, correlation with a 1d-template and a modified Maximally Stable Extremal Regions (MSER) approach) as well as four different algorithms for the reduction of false positives (snake algorithms, correlation with a set of 2d-tamplates and iterative segmentation). In the meantime investigations have been made to study the influence of pre-processing on the detection and classification process, additional algorithms have been implemented (e.g. k-means and higher order...
statistical based segmentation, co-occurrence matrixes based entropy and energy, ... ) and first results were achieved. All algorithms have been or will be tested using a SSS data currently covering 25 km² of the seafloor. These data were in part collected by the SeaOtter MK I AUV from Atlas Elektronik gathered in the Baltic Sea and the Mediterranean Sea.

**Keywords:** computer aided detection (CAD), computer aided classification (CAC), side scan sonar, automatic target recognitions (ATR), UUV

1. INTRODUCTION

Side scan sonar (SSS) systems are today the state of the art sensor for mine hunting, because they need a relatively short time to map large areas of the seafloor with a relatively high resolution.

Images from these SSS systems are normally analyzed by an operator. The time for this task, which may include complex scenes is very limited. Therefore the operator needs support by a computer aided detection (CAD) / computer aided classification (CAC) system to flag potential mine like objects (MLOs).

By the fact that an operator is in the loop a limited number of missed MLO detections or many false targets does not harm a CAD/CAC system. A very different situation is given for an automatic target recognition (ATR) system for an AUV where no operator is in the loop. For such occasion, the requirements for a computer aided detection and classification system must be set much higher.

In order to develop existing CAD/CAC methods towards an ATR system a study has been carried out at FWG with the assistance of FU-Berlin and FGAN-FOM [1]. Based on this study the existing object detection and classification software has been improved in the mean time. Primly two new screening algorithms were developed, implemented and tested in order to improve the detection rate. In addition, a different type of contrast enhancing filter has been implemented and compared with the already-existing one and some investigations with classification have been made.

2. OBJECT DETECTION AND CLASSIFICATION SOFTWARE

2.1. Overview

Two years ago, the development of image processing software for automatic object detection and classification has been intensified at FWG [1, 2 and 3]. This work initiated as Master Thesis and has developed into a PhD Thesis at FWG in cooperation with the Free University Berlin, Department of Computer Sciences and the Research Institute for Optronics and Pattern Recognition (FGAN-FOM). The developed software processes the data in four steps (see fig. 1).
At first, images are pre-processed. Pre-processing includes normalization, corrections for distortions such as height estimation plus slant range correction and geo-referencing. During the second processing step a screening is performed. So-called regions of interests (ROIs) are identified by a single or several combined screening algorithms. Currently a set of six screening algorithms are available in the system, which base on statistical features within a sliding window, highlight / shadow analysis after threshold segmentation, normalized 1d-cross correlation with a template, a modified Maximally Stable Extremal Regions (MSER) approach, k-means based segmentation or higher order statistic based segmentation.

To reduce the number of false alarms after screening a set of four false alarm reduction algorithms is available, which can be used stand alone or in combination. So far a single snake algorithm for the combined highlight and shadow area, a coupled snake algorithm with different coupled polygons for the highlight and the shadow area, a 2d-cross correlation with object templates and a algorithm using an iterative fuzzy segmentation followed by a classification process utilizing the existence of parallel lines for the object shadow contour have been implemented [2, 3].

2.2. New Approaches

In the next subsections some new approaches are discussed, which were developed implemented and tested in our object detection and classification software. We have compared a logarithmic-filter with a power-filter for contrast enhancing in the pre-processing, added two new screening algorithms a k-means based and threshold segmentation with neighborhood information and for the classification we have started as a first approach with two simple classifiers a K-nearest neighbor (KNN) and a Probabilistic Neural Network (PNN) for the classification.

2.2.1. Pre-Processing

SSS images may be quite dark and may show only low contrast. To overcome this problem for the processing steps a nonlinear logarithmic spreading was suggested in [3] as a standard technique to enhance the contrast. By this type of filter the pixel values in darker image areas are stretched more in comparison to the pixel values in bright image areas, so that a good contrast enhancement is achieved.

But, also other techniques to enhance the contrast exist and are in use for the processing of SSS images like the in [4] suggested power-filter, which can be modified by the parameter $r$ and follows the equation
\[ y = x^r \text{ with } 0 < r < 1. \] (1)

To get a fair comparison of the ability of the two filters for the pre-processing of a SSS image a test was performed comparing the found mine like objects (MLOs) after pre-processing and screening [5]. As screening algorithm, the 1d cross correlation-based screener was chosen due to the robustness of this algorithm. Fig. 2 shows that the logarithm-filter has the same performance or outperforms the power-filter for all values of \( r \) for our data. In addition, the running times of the algorithm were compared and found to be approximately equal. The difference in all tests were less than 1%. The comparison of the false alarms gives generally similar result. Only for values of \( r \) where the rate of found targets broke down the false alarm rate of the power-filter decreases significantly.

2.2.2. Screening Algorithms

Normally objects in SSS image appear as highlight-shadow pairs. These highlight-shadow pairs can be extracted automatically by segmentation. For the segmentation simple approaches like threshold segmentation perform segmentation based on the image histogram. This leads to a poor robustness against speckle and other noise. However, SSS images are typically noisy. Therefore methods that are more robust are required.

Fig. 2: Logarithm-/Power-filter (detected MLO by Correlation)
1: Logarithm-filter, 2: Power-filter (\( r = 0.03 \)), 3: Power-filter (\( r = 0.05 \)),
4: Power-filter (\( r = 0.1 \)), 5: Power-filter (\( r = 0.15 \)), 6: Power-filter (\( r = 0.2 \)),
7: Power-filter (\( r = 0.25 \)), 8: Power-filter (\( r = 0.32 \))

We recently added two new algorithms to our image processing software: a modified k-means based algorithm and a segmentation algorithm using neighborhood information.

The iterative k-means based screening algorithm uses block processing. Each block has the size of a typical averaged mine like object and is processed in an iterative way by the following steps: The algorithm puts in the beginning the center of the object highlight in the middle of the left half and the center of the shadow in the middle of the right half of the block (see fig. 3).
As starting value for the highlight pixel average the local maximum, for the shadow pixel average the local minimum and for the background average the mean of the maximum and minimum. The variance for the highlight and shadow areas base on the difference of pixel values to the mean value and the distance between the pixel position to the center of the highlight or the shadow cluster. Pixels in the image are assigned to the segments highlight, shadow and background based on the lowest variance [6].

Fig. 4 shows a typical result for the realized k-means algorithm. On the left side the original and segmented ROI including a target and on the right side a ROI without a target are shown. After the segmentation, the ROIs are defined by the cluster sizes.

The second recently implemented algorithm is a segmentation algorithm using neighborhood information. This is done by performing threshold segmentation based on a higher order histogram. Each new dimension in such a histogram represents an additional neighbor pixel. An example of such a histogram is shown for a pixel and the first neighbor in fig. 5. The x-axis represents the normalized pixel value for the associated neighbor pixel. The frequency that a certain combination of pixel values exists is color coded (from blue to red).
The red lines are indicating the segmentation boarders. This technique leads to fast segmentation for images since, in principle, the resolution of the SSS image is reduced by using the neighborhood for segmentation instead of each pixel value [6].

In fig. 6, the results for a normal threshold segmentation and neighborhood segmentation are compared. In both images the two MLOs can be identified, but the image generated by the segmentation algorithm using neighborhood information contains much less noise. The ROIs have been selected by finding highlight and shadow clusters of a certain size close to each other.

![Fig. 6: Results of the implemented segmentation with neighborhood information algorithm (left threshold segmentation without neighborhood, right threshold segmentation with a 14 pixel neighborhood)](image)

To assess the performance of the two new algorithms in comparison with the already existing ones a test on the SSS image database has been performed.

For the test parameter sets for the new algorithms were needed. To get them a parameter study over a small selected set of SSS images which included examples of targets (mines), man-made objects and false targets of all measurement campaigns were used, for optimization of these parameters. For the data from all measurement campaigns the same parameter set for each screening algorithm has used.

Fig. 7 shows the excellent performance regarding found targets for the screening algorithm using neighborhood information during segmentation, which holds over all considered measurement campaigns a detection rate of 90% or better and is, therefore, currently our best screening algorithm.

However, fig. 7 also demonstrates that the detection rate of the k-means based screening algorithm is significantly lower than the detection rate of the other algorithms. This could be

![Fig. 7: Screening results for the different algorithms](image)
processing, an unfavorable location of the object in terms of the dividing process could lead to segmentation failure. A sliding window approach would solve this problem. The implementation of such a sliding window will be one of the next steps.

2.2.3. Classification

Automatic classification is an important task in the processing queue. For first tests two simple classifiers were used, a Probabilistic Neural Network (PPN) and the K-Nearest neighbor (KNN) algorithm. The KNN classifying algorithm performs the classification by attributing the input sample to the class where the k-nearest training samples belong while the PNN approximates the probability density functions of all classes in the feature space during the training and attributes the sample to the class for which the estimated probability functions gives the highest probability. This algorithm achieved Bayes optimal classification.

![Fig. 8: Classification results for different parameters](image)

Fig. 8: Classification results for different parameters 
triangle green: MLO KNN, triangle blue: MAN KNN, triangle red: NOT KNN 
circle green: MLO PNN, circle blue: MAN PNN, circle red: NOT PNN

The following features for the classification were used, the highlight and shadow size and the pixel values inside and outside the highlight and the shadow area.

As a first approach we have chosen three classes for classification, which are mine like object (MLO) including all types of mines, man made object (MAN) and no target (NOT). This class structure will be changed in the next step to decide between different mine types.

Fig. 8 demonstrates the performance of the implemented classifiers. The k-values on the x-axis correspond to the number of used neighbors for the KNN classifier and to the used \( \sigma \) value for the PNN classifier (see table 1).

<table>
<thead>
<tr>
<th>k-value:</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
</tr>
</thead>
<tbody>
<tr>
<td>sigma:</td>
<td>0.05</td>
<td>0.10</td>
<td>0.15</td>
<td>0.20</td>
<td>0.25</td>
<td>0.3</td>
<td>0.35</td>
<td>0.40</td>
<td>0.45</td>
</tr>
</tbody>
</table>

Table 1: Dependents between the on the x-axis in fig. 8 given K values and the parameter \( \sigma \)

The optimal choice of the free parameters \( K \) and \( \sigma \) result directly from fig. 8, they are \( K = 1 \) for the KNN and \( \sigma = 0.3 \) for the PNN. It is obvious that the PNN shows better results in terms of right classified MLO and MAN than the KNN, but the classification performance in both cases is not good enough for automatic classification. Therefore, further
improvements like the use of support vector machines for the classification and new features are essential to achieve better classification results.

3. CONCLUSION AND OUTLOOK

In this paper, we have presented some new approaches in our object detection and classification software. A comparison between two commonly used contrast-enhancing filters, a logarithmic- and a power-filter was performed. The result is that the performance of the two filters is nearly equal, provided the parameter $r$ was chosen correctly. If that is not the case, the logarithmic-filter performs much better. Secondly, we have discussed two new screening approaches, a k-means based approach and threshold segmentation with neighborhood information to perform the segmentation in a higher order histogram. Here the k-means based algorithm has shown some need for improvements e.g. the use of a sliding window instead of the piecewise overlapping processing. The segmentation with neighborhood information algorithm shows an excellent performance regarding found targets over all datasets compared to the other algorithms. In the third part, two simple classifiers were presented as a first approaches for classification, a K-nearest neighbor algorithm (KNN) and a Probabilistic Neural Network (PNN). The PNN shows the better performance, but the obtained result is only a first step and must really be improved. Therefore we want to use Support Vector Machines as classifier. The simple set of classes which we used were only for the first tests and will be changed, to differentiate between different mine types. By doing this we will get some training problems, which we will hopefully overcome with parametric models for the training.

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ADAPTIVE ALGORITHM FOR SEA MINE CLASSIFICATION

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Abstract: The simultaneous increase of coverage rate and resolution makes the sonar image analysis a challenging problem for the classification operator. Over the last twenty years, algorithms have been developed to automatically detect and classify sea mines in sonar images. The detection task consists in selecting the image locations that potentially contain a mine. The detection algorithms have reached a performance level suitable for implementation on operational systems. The automation of the classification task is more difficult to achieve. It is commonly admitted that multi aspect fusion and high resolution, obtained either by synthetic aperture beamforming or by very high frequency transducers, are the key elements for a successful automation of the classification task. However, when the target’s environment is not cooperative, it is still difficult to compete with a trained operator using hull mounted sector scan sonar on board a MCM vessel. After a review of existing techniques the paper presents new approaches to introduce adaptive behaviour in the automated classification. The performance of this new algorithm is mostly assessed on modelled data.

Keywords: Sidescan sonar image processing, automatic classification, Dempster-Shafer evidential reasoning, neural networks.
1. INTRODUCTION

The sea mine countermeasure is mainly achieved by using two techniques: mine sweeping and mine hunting. Mine sweeping’s goal is to neutralize the mine by simulating ship signatures. On the opposite, mine hunting requires to search for all the mines in an area before to dispose them. Mine hunting is generally decomposed in four stages: detection, classification, identification and disposal. This paper addresses the challenging problem of the classification stage which consists in recognizing 3D shapes from 2D images. The consequence is that the geometrical conditions under which the target is acquired play a significant role in the classification performance. In the following, a particular geometrical relationship between the target and the sonar will be called an aspect of the target. Conventional mine hunting vessels rely on a sequence of images acquired by a hull mounted sector scan sonar to achieve multiple aspect classification. To better adapt to the environment the hull mounted approach can be improved by either adding variable depth capability or by mounting the sonar on a remotely operated vehicle (ROV). These improvements give more accurate images but do not significantly change the classification process based on the cumulative analysis of a sequence of images on the target. When the classification task is conducted manually, the sonar operator classifies a target both using the aspect diversity from several images in the sequence and the continuous evolution of target acoustic signature during the whole sequence.

Limited in the last decades to surveillance missions (i.e. change detection), the side-scan sonar is now able to increase its operational field to address mine hunting tasks, particularly for the detection and classification stages. The increased use of side-scan sonar is dictated by two major evolutions. The first is that the preferred pace of torpedo shaped autonomous underwater vehicles (AUV) is linear. The second evolution is the arrival on the market of higher resolution side-scan sonar using either synthetic aperture beam forming or very high frequency transducers. However, despite the resolution improvement, the limits of single view based classification are rapidly reached when the target’s environment is not cooperative. The most common perturbation is due to seabed micro structure (e.g. sand ripples, pebbles) and occlusion by other objects (e.g. stones, rocks).

To acquire multi aspect images of a target, side-scan sonar can be designed with multiple beams pointing to different directions. An example of operational use of this technique is the DUBM-44 side-scan sonar [1] operated by the French Navy. The sonar transmits in three directions with angles of -30, 0 and +30 degrees relative to broadside direction and the receiving array spans the whole 60 degrees. The sonar bandwidth is divided in three sub bands, one for each direction. Another way to gather multiple aspect images is to take advantage of the manoeuvrability of autonomous underwater vehicles or semi submersible drones. The multiple aspects of the target to classify are obtained by mission planning. Compared to the multiple aspect side-scan sonar, the mission planning approach is more time consuming but offers more freedom in the number and the directions of the views.

2. MULTIPLE ASPECT CLASSIFICATION

In the mid nineties, multiple aspect classification techniques [2,3] have been developed to improve the target classification capability of side-scan sonar. Multiple aspect classification, also called multiple view classification, can be implemented with two main approaches: input merging (IM) and output fusion (OF). In IM, the classifier receives on its input the
combination of the information from all the aspects. The classifier algorithm used for single view can also be used for multiple views. However, to ensure a reliable classification, one must take into consideration that the dimension of the input space increases linearly with the number of views and that the number of elements in the training set remains constant. A reduction of the dimension of the input space can practically solve this problem. The IM method expects that all the views are available and that the relative angles between the views are identical to those used for the training of the classifier. The performance of the classifier will be significantly affected if one view is missing or if the relative angles between the views are changing. Conversely, it must be noticed that this approach is well suited for side-scan sonar with built-in multiple aspects capability. The second approach (OF) is a fusion-based classifier which operates in two stages: a standard classifier which separately processes each aspect and a fusion algorithm aimed in combining the results from the first stage classifiers. This method offers more freedom in the number of aspects to be fused and in the relative angles between aspects. The classification stage can be achieved by a wide range of classifiers. In a first attempt [2,3], several kind of neural networks have been tested: a classifier (MNK) combining multilayer perceptron and Kohonen self organized feature map, a radial basis functions (RBF) classifier [4]. The fusion was achieved by logical combination or by simple averaging of the classifier’s outputs. The RBF was performing better than MNK and the fusion methods, logical combination and output averaging, were less accurate than the IM approach. A second attempt [5] to implement OF was to keep the RBF classifier for the first stage and to introduce Dempster-Shafer (DS) evidential reasoning [6] for the second stage. Compared to conventional decision theory using the Bayesian approach, DS reasoning offers several properties well suited for our fusion problem. An interesting property is that the support to decision is the power set, or frame of discernment, encompassing all combinations of sets including the class singletons. This allows for example to support belief to the set \{A,B\} when for a given view the RBF classifier is not capable to distinguish between class A and class B. Another property of interest is the Dempster’s rule of combination which allows the fusion of belief from two different views.

3. ASPECT FUSION USING EVIDENTIAL REASONING

3.1. Basis of the fusion method

Considering $\Omega$ the set of N classes, $\Omega = \{L_1, L_j, \ldots, L_N\}$, the power set of $\Omega$, also called the frame of discernment of $\Omega$, is defined by:

$$P(\Omega) = \{\emptyset, \{L_1\}, \{L_j\}, \ldots, \{L_N\}, \{L_1, L_j\}, \ldots, \Omega\}$$

(1)

The belief is assigned to the sets in $P(\Omega)$ through mass functions or support functions, $m(A_j)$, also called basic belief assignment (BBA) and defined as follows:

$$m(\emptyset) = 0$$

$$\sum_{A_j \in P(\Omega)} m(A_j) = 1$$

(2)
The mass functions, established from the results of the RBF classifiers, are fused by the Dempster’s rule of combination, also called orthogonal sum, and defined as:

\[
m_{1,2}(\emptyset) = 0
\]

\[
m_{1,2}(A) = \frac{1}{(1-K)} \sum_{B \cap C = A \cap \emptyset} m_1(B) m_2(C)
\]

(3)

(1 – K) is a normalization factor where K represents the degree of conflict and is defined as:

\[
K = \sum_{B \cap C = \emptyset} m_1(B) m_2(C)
\]

(4)

A very basic interpretation of the fusion results is to consider only the singletons, \{L_1, L_2, \ldots, L_N\}, of \(\mathcal{P}(\Omega)\) and to use their associated belief, \(m(\{L_1\}), m(\{L_2\}), \ldots, m(\{L_N\})\), to take the decision. However, they are other ways to get a more accurate estimate of \(P(A_j)\), the probability of \(A_j\). A commonly used tool is the pignistic probability \([7]\) defined by:

\[
BetP(A_j) = \sum_{A_k \subseteq A_j} \left( \frac{1}{|A_k|} \right) m(A_k)
\]

(5)

where \(|A_k|\) denotes the cardinality of \(A_k\).

3.2. Translation of classification outputs to fusion inputs

The classification output from the RBF cannot directly be assigned to belief. Therefore, a translation has to be defined. If the classification exhibits some uncertainty between two classes, the mass function of the corresponding set will get high support, but no support will be assigned to the corresponding singletons. If none of the RBF output is activated, support will be given to the full set of classes, \(\Omega\), which represents ignorance. If \(C_1, C_2, \ldots, C_N\) are the outputs of the RBF classifier, ranked in descending order, if \(L_1, L_2, \ldots, L_N\) are the corresponding classes, the automated translation method (used in [5]) of assigning belief from RBF classifier output can be summarized as:

\[
m(\{L_1\}) = C_1 - C_2
\]

\[
m(\{L_1, L_2\}) = C_2 - C_3
\]

\[
\vdots
\]

\[
m(\{L_1, L_2, \ldots, L_{N-1}\}) = C_{N-1} - C_N
\]

\[
m(\Omega) = 1 - m(\{L_1\}) - m(\{L_1, L_2\}) - \ldots - m(\{L_1, L_2, \ldots, L_{N-1}\})
\]

(6)

The above translation from RBF classifier outputs to support functions is achieved for each view and the Dempster’s rule of combination is applied transitively.
3.3. New OF classifier design

The method described in the previous subsection has been assessed [5] during SACLANTCEN GOATS’00 experiment with three target aspects collected by EDGETEC DF 1000 side-scan sonar mounted on OCEAN EXPLORER autonomous underwater vehicle (AUV) from Florida Atlantic University. Three classes were defined with two mine-like object classes (Cylinder and Truncated Cone) and a class for non-mine mine-like bottom objects (Rock). These results showed that the classification performance was slightly improved by acquiring and processing additional aspects. However, during this experiment, the number and the orientation of aspects were defined prior to start the mission of the AUV. The classification was conducted onboard the mother ship at the end of the mission after downloading and processing the sonar data. In the current work, the capability to fuse the classification results during the mission is studied. The idea is to collect the minimum number of aspects to remove classification ambiguity by choosing, during the mission and according the automatic classification results, the most favourable aspects. The new OF classifier design is based on the classifier used in GOATS’00 experiment. The classification stage, still completed by a RBF neural network, has been improved to better feed the input of the fusion stage. The number of classes has been augmented to take into account that for some aspects the feature vectors of two different targets can be similar. In this case a new class is defined as the set containing these two classes. Therefore, instead of learning ambiguously two singleton classes, the RBF will be trained to recognize the doublet of classes. The translation from RBF output to mass functions becomes extremely simple. If $C_i$ is the highest output of the RBF classifier and if $A_i$ is the corresponding set, the automated translation is simply:

$$m(A_i) = C_i$$

$$m(\Omega) = 1 - C_i$$

(7)

The automated translation in eqn. (7) is both a simplified and an augmented version of the BBA defined in [8]. It is simpler as no distance is required and it is augmented because it applies not only to unions of singletons but to all sets of the frame of discernment. Theoretically all the $2^N$ sets belonging to the frame of discernment $\Omega$ should be outputs of the RBF classifier. In practice only few elements of the training set exhibit ambiguities. To select only the relevant sets of the frame of discernment that have to be learnt by the RBF, a clustering algorithm is applied to the training set. Three clustering approaches have been studied: K-Means, Fuzzy-ART [9] and a simple similarity measurement based on the Euclidian distance. The later method analyzes the neighbourhood of each element of the training set. If the classes of the neighbours are different from the class of the element, a new class is defined as the set of the different classes encountered in the neighbourhood. This new class is created only at the first occurrence of this particular set of classes. The neighbourhood is a hypersphere centred on the element and the extent of the neighbourhood is controlled by the radius of the hypersphere. For K-Means and Fuzzy-ART methods, the elements of the training set are grouped into clusters and for each cluster the class is determined by the set of initial classes represented in the cluster.
4. CLASSIFICATION PERFORMANCE

The new OF classifier is tested on modelled data and the results are compared to the former design of OF classifier. The modelled sonar is a KLEIN 5000 side-scan sonar operating at 455 kHz. Considering that the multiple aspects are acquired by multiple tracks with different headings, the distance to the object must not be too large and, therefore, has been set to 25 meters. The modelling technique is based on ray tracing to accurately define the acoustic shadow of the targets. As the major perturbation to shadow contour is the roughness of the seabed, or the fine scale variations of bathymetry, a fractal model has been used for the seabed. The fractal dimension varies from 2.0 (flat seabed) to 2.5. The classification experiment considers 3 man-made objects, a 2 meters long 0.5 meter diameter cylinder, a 0.5 meter diameter sphere and a truncated cone (height 0.5 meter, base diameter 1.2 meter, top diameter 0.6 meter). Natural objects are stones generated by a fractal model. A training set of 1008 object’s aspects is defined and every element of the training set is described by a two dimensional feature vector and a class. The sonar image is segmented and the shadow’s contour is extracted. If we consider North the along track direction and East the across track direction, only the Eastern part of the contour is used. The first feature describes the symmetry of the contour and the second the flatness. The dimension of the feature vector is deliberately kept low to better assess the behaviour of the new OF classifier. The classes are C (Cylinder), S (Sphere), T (Truncated Cone) and R (Rock). After search for ambiguities in the training set by the clustering algorithm, 6 new classes have been created corresponding to the following set of the frame of discernment: \{C,S\}, \{S,T\}, \{C,R\}, \{S,R\}, \{T,R\} and \{C,S,R\}. The performance of the classifier is assessed using a set of 346 objects (1730 aspects). For every object, five aspects are modelled with a relative azimuth angle of 0, 30, 45, 60 and 90 degrees to the heading of the first aspect. Sample images from the test set are shown in figure 1. The performance of new OF classifier is compared to the previous OF classifier using 3 views (0, 45 and 90 degrees). The performance of the single view classifier is also recalled. The classification results are the average of 20 experiments (i.e. 20 different learning by changing randomly the order of the training elements). The standard deviation does not exceed 1.5 % and results differing of less than 1.5 % are considered similar. The previous OF design shows a good level of correct classification but does not perform well on mine shapes. The fusion tends to privileged rocks. The new OF design reach the performance of the previous OF design for the level of correct classification but outperform the old design on the percentage of mine correctly classified (increase of about 28 %). These good results come from the additional RBF classes \{C,R\}, \{S,R\}, \{T,R\} and \{C,S,R\} which better describe the possible ambiguity, for a given aspect, between rocks and mine-like shapes. The new design can be used to confer an adaptive behaviour to the classifier. If the fusion of two views raises enough difference between pignistic probabilities, there is no need to acquire and process additional views. On the opposite, if the result is ambiguous after fusing two views, additional views are required. A very simple adaptive scheme has been tested consisting in predefining the sequence of aspects. The fact that for some aspects the similarity between two mine shapes give insight about the orientation of the target has not been yet introduced in the adaptive scheme. Two sequence configurations, defined in terms of orientation in azimuth relative to the first view have been assessed \{0°, +90°, +30°, +45°, +60°\} and \{0°, -15°, +15°, -45°, +45°\}. The fusion process is stopped when the difference between the two highest pignistic probabilities is at least 0.5. The percentage of correct classification is higher (+ 3 %) but the classification tasks is slower as the average number of view goes from 3 to 3.31 and 3.27, for configurations 1 and 2, respectively. If the minimum difference between the two
highest pignistic probabilities is decreased to 0.2, the average number of views goes below 3 views (2.39 and 2.46) but the percentage of correct classification drops down to 77.8% and 80%, for aspect configurations 1 and 2, respectively. It must be noticed that the two configurations lead to almost similar results.

<table>
<thead>
<tr>
<th>Classifier</th>
<th>% non classified</th>
<th>% misclassified</th>
<th>% correctly classified</th>
<th>% mines correctly classified</th>
</tr>
</thead>
<tbody>
<tr>
<td>Adaptive OF design – config2</td>
<td>0.0</td>
<td>17.0</td>
<td>83.0</td>
<td>90.4</td>
</tr>
<tr>
<td>Adaptive OF design – config1</td>
<td>0.1</td>
<td>16.9</td>
<td>83.0</td>
<td>89.3</td>
</tr>
<tr>
<td>New OF design</td>
<td>0.2</td>
<td>19.8</td>
<td>80.0</td>
<td>88.7</td>
</tr>
<tr>
<td>Previous OF design</td>
<td>0.0</td>
<td>21.5</td>
<td>78.5</td>
<td>60.9</td>
</tr>
<tr>
<td>Single view RBF</td>
<td>0.3</td>
<td>28.1</td>
<td>71.6</td>
<td>62.5</td>
</tr>
</tbody>
</table>

Table 1: Classification results

5. DISCUSSION AND ONGOING WORK

A new classifier design has been proposed for output fusion (OF) of single aspect classification results. To better link the classification and the fusion stages, additional classes have been defined to represent the possible ambiguities between object’s shapes for some aspects. The new design performs globally as well as the previous design and even better if only mine shapes are considered. The additional classes and the Dempster-Shafer fusion allow for adaptive behaviour. Starting with 2 views, the OF classifier can use up to 5 views to improve the classification’s performance. The clustering algorithm plays a key role in defining the additional classes. The simple similarity measurement performs better than Fuzzy-ART and K-means on modelled data but the tests on experimental data will be helpful to choose the proper clustering algorithm.

The results presented and discussed in this paper consider a target description limited to two features. Further tests will be conducted with input space of higher dimension and a larger number of classes, corresponding to actual mine shapes. Taking into account that ambiguity between two shapes can give information about the orientation of the target and therefore plays a significant role on deciding the angle of the next aspect. The OF classifier will be enriched to consider this improvement. In particular, smarter strategy will be studied to reduce the number of views without significant loss in the classification performance. The assessment of this new classifier on actual experimental data from GESMA’s RADE2001 experiment, from DRDC-GESMA CAMARET 2003 joint experiment and from NURC-DRDC-GESMA CIDATEL’05 joint research project has started and will give more accurate conclusions on the interest of this approach.
Fig.1: Examples of 8 modelled targets used in the assessment of classification performance. For each target, 5 views are modelled with aspect angle of 0\(^\circ\), 30\(^\circ\), 45\(^\circ\), 60\(^\circ\) and 90\(^\circ\) relative to the first aspect. The 8 targets are, from bottom left to top right, a cylinder, 5 rocks, a sphere and a truncated cone.

REFERENCES


BISTATIC SCATTERING ON SEABED AND TARGETS:
COMPARISON OF SCALED TANK EXPERIMENTS WITH FULL-SCALE SEA TRIALS

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Abstract: Detecting/classifying seafloor targets requires a better understanding of seabed/target interactions. We compare here analyses of scaled (10:1) tank experiments and full-scale sea trials conducted as part of the EC-SITAR project. Tank experiments in Bath used a 238-kHz sonar (10° beamwidth, 5° sidelobes) imaging a silt seabed at 45°, with multiple scattering angles (16°-70°) and bistatic angles (±40° by 2.5° steps). Our analyses showed a good agreement with the APL-UW model for bistatic scattering on a silt seabed. We have now quantified the importance of the sidelobes in determining the instantaneous bistatic scattering area and the exact conditions (i.e. seabed tilt, sediment mixing). The scattering from different target types (filled or empty, aluminium or steel) can now be assessed by comparing with the influence from the bare sediments. These results are usefully compared to full-scale experiments in muddy sediments. The SITAR’03 sea trials were conducted over a toxic waste dumpsite in Möja Söderfjärd, Stockholm Archipelago (Sweden). A TOPAS-120 parametric array (3-4° beamwidth, negligible sidelobes) was flown on the FOI-PLUMS ROV, imaging the seabed (and potential targets) 15 m away. Scattering was recorded on a hydrophone chain 15 m further. DGPS and a LBL transponder net gave accuracies of 1° for angles and 0.1 m for positions. The sonar was tilted at 3 different angles, and the hydrophone chain allowed access to 5 distinct scattering angles. Bistatic angles were varied and the ROV was flown to match the linear and rotation scans of the tank experiments. Similar analyses were made, including the effects of experimental uncertainties. The synergy between the two types of experiments is used to assess the importance of sidelobe contributions, and how seabed contributions can be used in target analyses.

Keywords: bistatic sonar, tank experiments, sea trials
1. BISTATIC SONARS AND BURIED WASTE

The seafloor has been one of the most convenient sites to dump bulk chemical ammunition, toxic waste and radioactive products, from industrial or military sources, during most of the previous century. This is now prohibited by the London Convention (1975), not always respected. Because of pressing environmental concerns, such as contaminants leaking into the surrounding biota or immediate risk from exposure to the waste (e.g. when trawling), there have been growing efforts to locate these dumpsites and assess the risks they cause. The location problem is compounded by the fact that most of the waste has undergone partial or complete burial. Even when properly documented, it is not always located where it was laid, either because of dispersal during the dumping process or because of bottom currents and sediment redistribution. In many areas, the sheer number of identified objects (e.g. the Farallon Island Radioactive Waste Dump, with 47,800 barrels scattered over 1,400 km² [1] or the Möja Söderfjärd dump site, where 450 targets have been identified in a few hundred m² [2]) adds to the scale of the problem.

Traditional tools such as sidescan sonar can produce accurate maps of objects proud on the seabed or partly buried if imaged in appropriate conditions, e.g. [1,3]. Many theoretical studies and a significant number of experiments have showed the information necessary to identify and distinguish targets on/in the seabed is contained in the full 3-D scattered acoustic field, natural objects (e.g. rocks and boulders) yielding distinct acoustic signatures from man-made objects (like mines or dumped objects). The European SITAR (“Seafloor Imaging and Toxicity: Assessment of Risk caused by buried waste”) project has used this 3-D structure to investigate the imaging of target(s) in a well-documented dumpsite with bistatic sonars, using different source and receiver geometries [2]. Scaled experiments were used to prepare for the sea trials and identify the optimal geometries. We compare here the analyses of both datasets, looking in particular at the role of experimental uncertainties (e.g. transducer sidelobes, sediment mixing and seabed tilt), at the comparison of seabed bistatic scattering with a recognised and validated model, and at how seabed contributions can be used in target scattering studies. Section 2 focuses on the main results from the scaled tank experiments, whereas Section 3 presents the first results from the full-scale sea trials. Comparisons between the two datasets are discussed in Section 4.

2. SCALED TANK EXPERIMENTS

2.1. Setup

The setup was designed to be a scaled version of the conditions expected at the SITAR’03 sea trials site in Möja Söderfjärd (Sweden), with a scaling factor of approximately 10:1 (Fig. 1). The underground tank used is 5.00 m long, 1.54 m wide and 1.80 m deep. The water depth was kept constant at 1.45 m. The sediment tray used in this study was filled with thoroughly degassed silt, 14 cm deep. For a scaling factor of 10:1, this matches the soft muddy sediments found in Möja Söderfjärd [2]. Careful preparation ensured all sediments were water-saturated and their surfaces were as smooth as possible [4]. The imaging transducer transmitted at 238 kHz and was tilted at a fixed angle (θ) of 45°, 0.5 m from the target. The scattered signal was measured with an omnidirectional hydrophone mounted on a robotic system. To complement previous measurements [5], the range of scattering and bistatic angles was significantly
extended [6]: scattering angles ($\theta_s$) varied between $\sim16^\circ$ and $\sim70^\circ$ (50 distinct values); bistatic angles ($\Phi$) varied $40^\circ$ either side of in-plane with a $2.5^\circ$ step (33 distinct values).

Fig.1: Scaled tank experiments (left) were used to prepare for later sea trials (right). See text for details.

A similar set-up was used in earlier experiments in 1999-2001 [5,7]. Bistatic scattering from silt was then compared with predictions from the APL-UW model [8] intended for 10-100 kHz but successfully tested at 240 and 455 kHz by other workers [9,10]. Although the model’s predictions for in-plane scattering matched experimental results very closely [7], they seemed to overestimate out-of-plane scattering. Analyses at the time identified the actual interface roughness and the approximation of scattering areas as constant as the most likely causes for these differences. The SITAR experiments, conducted in 2002-2005, extended these experiments by using more seabed types (Fig. 1), bare and with a series of targets at different angles and burial depths, over a larger range of scattering geometries [6,11]. Howey and Blondel [12] showed the local tilt of the sediments and a slight amount of sediment mixing could explain small deviations between experiment and model, and that most of the deviations come from the calculation of the actual bistatic scattering area, combining the main beam and the two sidelobes of the transmitter with the receiving pattern of the receiver. Section 2.2 shows the extension of this work.

### 2.2. Main Results

For comparison with the APL-UW model, the experimental measurements were converted into scattering strengths using the bistatic sonar equation. The expected strengths were calculated using the mean grain diameter of the sediment surface. Although more complex models exist, the APL-UW model was selected because of its relative simplicity and ease of use, as the use of only the mean grain size was more adapted to tank measurements needing not to disturb the layers of sediments and to sea trials with limited time on site. Initial comparisons between bare silt measurements and modelling showed regular discrepancies. These were shown by [12] to result from the combination of variations in the sediment grain size distribution (its measurement with optical microscopes indicated some contamination from the surrounding sediment trays); averaging of several realisations of the APL-UW model with the actual grain size distributions yielded much more accurate predictions.

The bistatic sonar equation depends on the exact and accurate calculation of the instantaneous bistatic scattering area. In these experiments, the transducer used had a circular main beam with a 3-dB beamwidth of $10^\circ$. The main beam intersected the sediment in the shape of an ellipse with an area of $0.0171$ m$^2$, though due to the short pulse length of $39.2$ $\mu$s, a maximum of $0.0096$ m$^2$ was ensonified at any one time. The transducer’s sidelobes are
small and symmetrical at 12° from the main beam’s centre, with 3-dB beamwidths of 5° and 6° respectively, and levels 14.5 dB and 16 dB lower than the main beam [7]. The intersection of the side lobes and the sediment correspond to ellipses with areas of 0.0041 m² and 0.0060 m² respectively. Calculating the scattering area for a transmitter with sidelobes is not straightforward: [13] give an approximation in the backscatter case and [14] also mentions the effect of short pulse lengths on bistatic scattering strengths. Here, analytical derivations of the scattering area in a bistatic configuration were calculated for elementary areas of 1 mm² over the entire tray and the contributing scattering area was derived for each angular combination used in the experiments. Fig. 2 shows typical variations of the contributing scattering area for one angular combination; one sidelobe is completely ensonified and makes up a significant amount of the overall scattering area.

Fig. 2: Left: times of ensonification (in ms) for all points in the beam pattern. Right: instantaneous scattering area, with a return time accounting for the transmitted pulse length (areas in white are not contributing for this particular geometry)

Initial comparisons of the experimental results with the APL-UW model assumed a flat surface, but evidently any tilt in the sediment surface will affect the actual bistatic geometry (angles $\theta_i$, $\theta_s$, and $\Phi$). This in turn will affect the projection of the beam pattern onto the sediment, the calculations of the instantaneous scattering area and the small changes from the transmission loss, i.e. all the main factors of the bistatic sonar equation. The sediment tilt could however be back-fitted to the data by accounting for all reasonable variations from the general tilt. In this particular case, the tilt of the sediment in the vicinity of scattering was approximately 10° towards the hydrophone. This is supported by vertical monostatic depth sounding data taken just prior to the experiments. Fig. 3 shows how the data compares to the APL-UW model before correcting for the actual sediment tilt and after back-fitting.

This approach was made necessary by the relative imprecision of the measurements of the scattering surface and exact bistatic geometry of the experiments, and it can be readily extended to measurements at sea. Sediment grain size distribution in the tank was not exactly as expected, because of contamination from neighbouring trays. At sea, sediments are rarely homogeneous, and this approach could account for different distributions around a mean value. The tilt in the sediment surface was greater than expected, and this could also be accounted for by looking at deviations from this value at local angles. This is of obvious interest for sea trials, where local surfaces will be affected by sediment redistribution and (at very small scales) bioturbation or other processes. Finally, the exact calculation of the instantaneous bistatic scattering area can fully account for the transmitter sidelobes and the deviations in the local scattering geometry. Identifying exact seabed contributions can then be used to better understand the contribution from targets on a similar seabed (e.g. [15,16]).
3. FULL-SCALE SEA TRIALS

3.1. Setup

This approach can be used advantageously in sea trials, in this case the SITAR’03 sea trials, carried out in 2003 over a toxic dumpsite in Mäija Söderfjärd, Stockholm Archipelago (Sweden) [2]. The bistatic part of the trials was conducted aboard HMS Färönsund (Royal Swedish Navy) and consisted in deploying near specific targets the FOI PLUMS Remotely-Operated Vehicle (ROV) fitted with a parametric sonar, doing line scans (flying toward a target) and rotation scans (around a target). The ROV was tracked and positioned accurately using a long-baseline transponder net. While taking measurements, the ROV was locked to hold constant position and heading as well constant angles of roll, pitch, and yaw; accuracies of 1° for angles and 0.1 m for positions were reported. The imaging transducer was a TOPAS-120 parametric array (3-4° beamwidth, no sidelobe); it used several signals during the sea trials and could be tilted at different angles to the seabed. Bistatic scattering from seabed and target(s) was measured with a hydrophone chain, deployed at a fixed distance (Fig. 4) and positioned with ship DGPS. Real-time checking of signal quality revealed after deployment that several hydrophones were not working properly, and only 5 hydrophones were operational. They were respectively 17, 15, 13, 11, and 9 meters above the seabed allowing for 5 distinct scattering angles from each ROV position. The total range of incidence and scattering angles used in the trials is shown in Table 1: bistatic angles were varied by moving the ROV along an arc circle loosely centred on each target.
Table 1: Total range of incidence (θ) and scattering (θ_s) angles, with number of values N.

3.2. First Results

These trials yielded a large quantity of measurements (~17 GB for bistatic data), and we focus here on 3 of the 81 bistatic configurations investigated. These are particularly suited for comparison as they were consecutive and taken in a time span of ca. 10 minutes, resulting in comparable environmental conditions. They also used the same incidence angle (33°), bistatic angle (0°), ROV depth (61 m) and transmission signal, a 20-kHz Ricker pulse long enough to ensonify the entire area around the target. This means only the area ensonified and scattering angles can change. The target was a rectangular metal box with square edges, half buried in soft muddy sediments. The first configuration ensonified the seabed patch 5 m in front of the target (and therefore 24.5 m from the hydrophone chain); the second configuration ensonified the target itself, and the third configuration ensonified sediments 5 m behind the target (14.5 m from the hydrophone chain). 100+ pings were recorded at each position. As both the ROV and hydrophone chain moved slightly due to water movement, directly averaging pings loses both signal amplitude and clarity. Thus the envelopes for all pings were computed and averaged before analyses. The signals received in all 3 configurations are shown in Fig. 5, along with geometrically expected times of direct arrival and first return.

Fig. 5: Signal envelopes received at the 5 working hydrophones for 3 distinct bistatic geometries; vertical lines show the expected times of direct arrival (dashed lines, well before the first seabed/target returns) and of first bottom return (solid lines).
The expected times of arrival are offset from the actual measurements: this is due to a general tilt of the seabed in this area (ca. 7° as inferred from monostatic depth soundings). In all 3 configurations, the return from the seabed and/or target shows a first peak, with a more widely spread group of second peaks (scattering within the soft muddy sediments, with gassy inclusions and/or scattering within the target). Despite the lack of correction, the scattering strengths can already be compared to the APL-UW model (Fig. 6); sea measurements are more scattered than the tank ones but show the same pattern of over-estimation by the model. Current efforts are now looking at the exact tilt, using all 81 bistatic configurations measured in this location, and identifying the exact target returns.

Fig. 6: Scattering strengths (in dB) of the first (+) and second (x) return peaks from the sediment, the first return peaks from the target (*), and predictions by the APL-UW model

4. CONCLUSION

Laboratory experiments and measurements at sea are complementary, and this article shows how the analysis of scaled tank experiments can be used with sea trials data. The SITAR tank experiments were used in first instance to define the best surveying strategies at sea [2]. To separate the bistatic scattering from the seabed and individual target responses, theoretical expectations from the APL-UW model [8] were compared with experimental measurements. Following on the earlier results of [12], three distinct areas of improvement were identified: (1) the unanticipated mixing of sediments can be accounted for by measuring grain size distributions of samples and weighing contributions from different model realisations; (2) the tilt of the local scattering surface can be back-fitted to measurements; (3) the instantaneous scattering area has the strongest influence on the scattering strength, and it is affected by the actual geometry (including local surface tilt) as well as signal length and transmitter/receiver beam patterns. Understanding the exact variations of seabed scattering can then be used to understand the actual contribution of targets in/on a similar seabed (e.g. [2]). This approach was used with the first analyses of the complex dataset acquired during the bistatic part of the SITAR’03 sea trials (Moren et al., in [2]). In the few examples presented here, they show how the tilt of the seabed can be detected and derived. They also show the overall comparison with the APL-UW model and target scattering at different angles.
5. ACKNOWLEDGEMENTS

The scaled laboratory experiments were conducted by PB, N. Jayasundere and M. Cosci (U. Bath) and the sea trials measurements were acquired by the crew and scientists aboard HMS Fårösund, as part of the European Union project SITAR (contract #EVK-3-CT2001-00047). Prof. N.G. Pace (U. Bath) assisted in the calculation of the instantaneous scattering areas and offered helpful suggestions to relate these to other studies in similar conditions.

REFERENCES

ESTIMATION OF DETECTION/CLASSIFICATION PERFORMANCE USING INTERFEROMETRIC SONAR COHERENCE

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Abstract:

This paper proposes a new performance metric for interferometric side-looking sonar. With this metric the probability of target detection and classification is reported as a function of coherence instead of across-track range, which is traditionally used. The coherence values are estimated from interferometric sonar processing, and can be converted into equivalent signal to noise ratios. The metric ensures that system performance is estimated based on sensor data quality of the actual mission, and as local values instead of large area averages. The method is demonstrated on synthetic aperture sonar data from HISAS 1030, showing increased consistency between shallow and deep water results, compared to the traditional metric.

Keywords: Target detection/classification, Performance estimation, Minehunting, Interferometry, Sonar coherence.
1. INTRODUCTION

Reliable performance estimates are required for consistent use of the results from target detection/classification systems, as well as for operation planning. For underwater minehunting it is necessary to estimate the total effectiveness over the mission area, based on all individual sonar views of the seafloor. Minehunting operations are to an increasing extent performed with Autonomous Underwater Vehicles (AUV) equipped with high-resolution side-looking sonar, either traditional Side-Scan Sonar (SSS) or Synthetic Aperture Sonar (SAS) [1]. The performance of side-looking sonar systems is typically reported as the probability for correct target detection and classification as function of range, called P(y) curves [2]. These curves depend on the combined properties of the target, sensor, data analysis system (manual or automated), bathymetry, seafloor characteristics and sonar environment.

The detection and classification capacity of minehunting systems has traditionally been estimated from a large statistical sample gathered through systematic and comprehensive sea tests with various deployed targets in different environments. Minehunting is usually performed in harbours and littoral waters, where sonar conditions are highly variable. System performance established for other environments may then be inaccurate. Multiple P(y) curves can be prepared to handle expected variations in e.g. seafloor roughness and clutter density. However, preparing curves for every possible combination of the environmental parameters (bathymetry, wind speed, sound velocity, seafloor reflectivity, etc) that affect the sonar performance is impracticable. Also, the mission environment may be complex and rapidly changing in space and time, and perhaps insufficiently known. It may thus be difficult to select the appropriate P(y) curve for a given underwater location.

In [2] P(y) curves for a mission were estimated by injecting synthetic target responses into the real sonar images and calculating the performance of Automatic Target Recognition (ATR) algorithms on these ground truth data. The target responses were simulated based on results from a sonar performance prediction tool. Although this approach represents significant progress as it uses the actual mission data to estimate performance, it is still assumed that the local sonar environment can be accurately specified and that the mission area can be divided into a few distinct regions, each with a representative P(y) curve.

We propose a new performance metric: probability of correct target detection/classification as function of sonar coherence. The coherence is obtained from interferometric processing of the data from two sonar receivers mounted with a vertical displacement. The coherence can be converted into an equivalent signal to noise ratio (SNR), making it well suited as a system performance parameter. The SNR is commonly used as a sensor performance parameter in simulations [3][4]. The correspondence between coherence and detection/classification probability, P(SNR), can be established through the same approaches as for P(y) curves. Multiple P(SNR) curves may be produced to handle different seabed characteristics. This metric ensures that the performance is estimated based on sensor data quality of the actual mission, and as local values instead of large area averages.

2. SONAR COHERENCE
The coherence, $\gamma$, is defined as the magnitude value of the zero-lag normalised cross correlation [5]

$$\gamma = \frac{|E\left\{s_1 s_2^*\right\}|}{\sqrt{E\left\{|s_1|^2\right\} E\left\{|s_2|^2\right\}}}, \quad 0 \leq \gamma \leq 1$$  \hspace{1cm} (1)

where $s_1$ and $s_2$ are two co-registered, zero-mean Gaussian random sequences (This is not to be confused with the spectral coherence function). Assuming that $s_1$ and $s_2$ are delayed versions of a Gaussian random sequence in additive uncorrelated Gaussian noise

$$s_1(t) = s(t) + n_1(t)$$

$$s_2(t) = s(t + \tau) + n_2(t)$$  \hspace{1cm} (2)

the signal to noise ratio, SNR, can be derived from the coherence [5][6][7]

$$\text{SNR} = \frac{\gamma}{1 - \gamma}$$  \hspace{1cm} (3)

The spatial coherence can be calculated from two displaced sonar receivers observing the same seafloor scene. Two receivers with a vertical displacement (baseline) can form an interferometer such as the HISAS 1030 interferometric SAS [8]. In this case, two types of spatial coherence can be calculated: 1) SSS coherence; 2) SAS coherence; There are fundamental differences between the two, in particular regarding along-track resolution. In addition, the temporal coherence can be calculated from ping to ping overlapping elements in the phased array receiver, used in sonar micro-navigation in SAS processing [9].

When baseline decorrelation is compensated for [5][7], the SNR (3) can be used as sonar image quality measure.

In shallow waters, the received signals can be contaminated by unwanted multipath where the sonar signals are reflected by the sea surface. Multiple propagation paths to two spatially displaced receivers cause decorrelation or loss of coherence. Similarly, the coherence can only be high for sonar signals without multipath. Hence, the SNR (3) can be used to map the signal to multipath ratio [10]. In areas where the sonar performance is not limited by multipath contamination, the SNR (3) will map signal to additive noise (either self-noise, ambient noise or interference). Excessive noise level or loss of signal (e.g. in acoustic shadow zones) yields low SNR.

In this work, we chose the SSS interferometric coherence as sensor quality measure. From observations in shallow waters, this coherence has proven to be a reliable multipath tracer [10]. This measure is also available in the standard processing of the HISAS 1030 SAS. A potential better choice for ATR performance would be SAS interferometric coherence, since this is estimated in the same coordinate system as the SAS image and incorporates possible image artifacts (defocus) induced during SAS processing. This measure will increase the processing time, though.

3. DATA PROCESSING
The sonar raw data was processed by FFI’s FOCUS toolbox [8] to produce default SSS bathymetry and SAS imagery.

The target detector was based on the deformable match filter [11]. It convolves the magnitude normalised SAS image with a generic target signature consisting of a highlight mask followed by a shadow mask. The output is a weighted sum of the two mask responses. Adaptive threshold values are used to segment the match filtered image and connected segmented pixels are clustered into a single detection. Clusters with too few pixels are discarded.

The classifier was a Support Vector Machine (SVM) using only two features as input. The first feature was the segment size in the match filter image and the second was the highlight/shadow contrast relative local background variations in the normalised image. The classifier outputs a confidence value between 0 and 1 indicating the detected object’s degree of “mine-likeness”. Targets with confidence value above 0.5 were accepted as correct classifications. This simple classifier was originally developed as a discriminator stage between detection and classification, but was selected for this study as focus is on coherence for performance estimation rather than classifier development.

The positions of the target responses were mapped from SAS to SSS data coordinates and coherence values were extracted using a 3 m x 3 m median window for smoothing. The window thus covered both the target and some of the adjacent seafloor.

4. EXPERIMENTAL RESULTS

The proposed performance metric is demonstrated on data from the HISAS 1030 sonar mounted on the Royal Norwegian Navy’s HUGIN 1000-MR AUV [1]. The data was obtained during six individual missions at two different test ranges in Norwegian fjords. Three of the missions were performed with one MP80 exercise mine placed on the seafloor in deep water (73 and 197 m depth), while the other three missions were performed in shallow water (10-25 m depth) with one MP80 exercise mine and four mine-like cylinder objects (1.5-2.1 m length) as targets (Table 1). The surrounding seafloor of all the targets was fairly smooth, but the reflectivity was higher in shallow water due to seafloor pebbles.

<table>
<thead>
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<th>Mission no</th>
<th>No of targets</th>
<th>Total no of target views</th>
<th>Target depth [m]</th>
<th>Sensor altitude [m]</th>
<th>Bottom type</th>
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<td>197</td>
<td>20-30</td>
<td>Mud</td>
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<td>38, 77, 82</td>
<td>11-24</td>
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<td>5, 6</td>
<td>1</td>
<td>18, 18</td>
<td>73</td>
<td>25-35</td>
<td>Mud</td>
</tr>
</tbody>
</table>

Table 1: HISAS data sets used in this study.

Figs. 1 and 2 show 10 m x 10 m cropped SAS images of the MP80 target at various slant ranges for deep and shallow water, respectively. The image resolution is approximately 4 cm x 4 cm. Dynamics of the grey-tone scale is set to 45 dB for all images. All six displayed target responses were detected and classified as mine-like.
In deep water, the target highlight and shadow are easily distinguished from the background seafloor at all ranges, although the shadow contrast is somewhat reduced at far range (179 m). In shallow water, however, the shadow contrast is reduced already at 76 m slant range and is almost indistinguishable at 121 m range. This is due to multipath signals gradually decreasing the SNR with increasing range. Initially, only shadows are degraded, but eventually also highlight contrast is reduced due to the increased background signal level.

Fig. 3 shows an interferometric SSS coherence image for a 100 m x 200 m seafloor area in deep water. The image is projected onto ground range with an image resolution of approximately 0.5 m x 0.5 m. The MP80 target at 62 m slant range in Fig. 1 is visible as a small region of lower coherence at along track distance 75 m. Also visible is a larger region of low coherence near the lower image edge. This is caused by an abrupt hollow in the seafloor, approximately 2 m deep, introducing an acoustic shadow along the leading hollow edge. The narrow, horizontal lines with low coherence values are due to the vehicle’s acoustical links interfering with the sonar. This illustrates how the coherence can be used as a measure of sonar quality, as low values indicate inferior SNR. Overall, the coherence is highest between 40 m and 130 m range, and then slowly decreases towards maximum range.

Fig. 4 shows a corresponding interferometric SSS coherence image for shallow water, with the target at 31 m slant range from Fig. 2 visible at approximately 50 m along track distance. The water column is shorter than in Fig. 3, as the sensor altitude was reduced in shallow water (Table 1). The coherence is highest from 15 m to 60 m range, with a maximum value similar to that of Fig 3. However, coherence then drops drastically towards 100 m range and from there slowly decays towards maximum range. This is consistent with the rapid degradation of shadow contrast in Fig. 2.
Figs. 3 and 4 present the single-view results from automatic detection and classification of the targets. Undetected targets were assigned zero classification confidence. The target classification confidences in Fig. 5 are high at short ranges for both deep and shallow water, but decrease at long range. However, the decrease starts earlier and is much more severe in shallow water. From 80-100 m range there is a large discrepancy between the two $P(y)$ curves. This result corresponds with the coherence images in Figs. 3 and 4.

Plotting the target classification confidences and probabilities as function of SNR (Fig. 6) instead of range, yields a better correspondence between the performance curves in deep and shallow water. In both cases, confidence values decrease with decreasing SNR and the response distributions overlap for similar SNRs. This suggests that a large data base of $P(SNR)$ from both deep and shallow waters can be used to estimate performance at any depth, while using a similar $P(y)$ data base would yield estimation errors for both deep and shallow depths due to averaging.

The two deep water performance curves in the right plots of Figs. 5 and 6, exhibit a distinct knee at far range and low SNR, respectively. This can be attributed statistical variations due the small number of observations in these parameter intervals. As evident from
the left scatter plots, only two deep water target responses failed to be correctly classified. These responses were contaminated by interference and highlight artefacts.

A question can be raised whether the poorer classification performance in shallow water could be caused by the rougher seafloor, i.e. pebbles vs. mud. Indeed the $P(y)$ curve for shallow water (Fig. 5) is slightly lower even at short range, but the difference is small compared to the large differences at long range. We thus conclude that multipath is the dominant parameter in our experiment.

The proposed performance metric, $P(SNR)$, can also be used for interferometric SSS. It is particularly well suited as an alternative to $P(y)$ for SAS, though, as theoretical along-track resolution in SAS images is independent of range. The fundamental parameter bounding the performance for a given SAS system is thus SNR rather than range.

![Fig. 5: Scatter plot of target classification confidence as function of slant range (left) and $P(y)$ curves for indicated confidence threshold (right).](image)

![Fig. 6: Scatter plot of target classification confidence as function of SNR (left) and $P(SNR)$ curves for indicated confidence threshold (right).](image)

5. CONCLUSIONS

We have proposed a new performance metric for interferometric side-looking sonar: the probability of correct target detection and classification as function of sonar coherence. The sonar coherence provides an in-situ measurement of the signal to noise ratio, which is a fundamental parameter bounding achievable sensor performance.
This metric has been shown to give more consistent results for a combination of deep and shallow water HISAS 1030 sonar data than the traditional P(y) curve. This preliminary study will be followed by further investigations on the relation between coherence and detection/classification performance regarding estimate robustness and whether SSS or SAS coherence should be used. We will also evaluate methods for incorporating varying seafloor characteristics into the performance estimates.

6. ACKNOWLEDGEMENTS

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Abstract: Automatic Target Recognition is a challenging task as the response from an underwater target may vary greatly depending on its configuration, sonar parameters and the environment. Template matching avoids this problem by creating a set of templates that covers the variation in the target response. However, many parameters can affect the target response, and in practice only templates that are sufficiently different are included in a template database, to avoid computational overload. Thus, it is important that the similarity metric, which is applied for comparing a template to a sonar image, is robust with respect to minor changes in the target response. Furthermore, a similarity metric should be sensitive to small differences so that it may distinguish between objects which have related shapes, but belong to different classes.

In this paper, we examine a selection of similarity metrics and evaluate their performances. This evaluation is performed on simulated and real Synthetic Aperture Sonar images.

Keywords: Template matching, automatic target recognition, synthetic aperture sonar

1. INTRODUCTION

Object classification is often performed by segmenting the object response into a highlight and a shadow area. Geometric, statistical and texture features are then computed from the segments. The features are finally used as input to a supervised learning algorithm, which assigns the object to one of a predetermined set of classes. This approach requires, however, that the features are invariant to typical changes in the object response. Developing such features may not be possible, as the response of an underwater target may vary greatly depending on its configuration (e.g., orientation, immersion into the seafloor), sonar
parameters (e.g., range to target and sonar altitude), and the environment (e.g., the seafloor, multi-path response).

Model based approaches provide a solution to this problem. These approaches typically apply a simulator to create 2D template images from 3D models of selected targets. A sonar image of an unclassified object may be compared to the template images by computing a similarity score between a template and the image. The object may then be assigned to the class of the highest scoring template. These approaches avoid the problem with the variation in the object response by creating a set of templates under many different conditions for each target class.

Several similarity scores for template matching have been reported in the synthetic aperture sonar (and radar) literature, including the cross-correlation [3;4], likelihood scores [1] and probabilistic models [4]. Some scores require that the image and the template have been segmented into highlight and shadow regions and measure the similarity of the segments. Reed et al. [7] use for example the Hausdorff distance to measure the largest distance between a pixel on the shadow perimeters to its closest neighbour on the other shadow perimeter. Fawcett and Myers [3] use several features to measure the fraction of highlight or shadow pixels in the template that match corresponding pixels in the image.

In this paper, we evaluate three similarity scores (cross-correlation, expected utility, segment based matching), and study their discriminative abilities and their robustness to environmental conditions. There is a large number of conditions that may possibly affect the target response and in order to avoid computational overload, only parameters that cause major differences in the target response may be handled by template creation. The similarity metric must consequently be robust to minor changes in the target response.

This paper is structured as follows: Sec. 2 describes the different matching methods used in the paper. The results are given in Sec. 3, and we finish with conclusions in Sec. 4.

2. MATCHING METHODS

2.1. Correlation

The most popular metric for measuring the similarity between an image and a template is the correlation coefficient (e.g., [3;4]). This is defined as

\[
    r(i,j) = \frac{\sum_{k=1}^{m} \sum_{j=1}^{n} (x_{i+k,j+l} - \bar{x}_{i,j}) (t_{k,l} - \bar{t})}{\sqrt{\sum_{k=1}^{m} \sum_{j=1}^{n} (x_{i+k,j+l} - \bar{x}_{i,j})^2 \sum_{k=1}^{m} \sum_{j=1}^{n} (t_{k,l} - \bar{t})^2}}
\]

(1)

Here, the template origin has been aligned with position \((i, j)\) in the image. It is assumed that the template consists of \(m\) by \(n\) pixels and that the image is larger than the template. \(t_{k,l}\) and \(x_{k,l}\) denote the value of the pixel at \((k, l)\) in the template and image (respectively). \(\bar{t}\) denotes the mean template value, and \(\bar{x}_{i,j}\) denotes the mean of the subimage with the same size as the template and origin at \((i, j)\) in the image. This score depends on the alignment of the image and the template, and to make it shift invariant it is necessary to maximize it over all possible shifts. This can be computed efficiently using the Fourier transform since the correlation matrix \(c\) with \(c_{i,j} = \sum_{k=1}^{m} \sum_{j=1}^{n} x_{i+k,j+l} X_{k,l}\) can be obtained as \(c = F^{-1}[F(x) * F(t)]\) where \(F\) and \(F^{-1}\) denotes the Discrete Fourier transform and its inverse (This correlation-based alignment was also used for the other scores in our experiments).
2.2. Probabilistic matching

It can be shown that maximization of the correlation coefficient leads to maximization of the likelihood when the noise in each pixel is normally and independently distributed [6]. The noise in a sonar (amplitude) image follows, however, a Rayleigh distribution (which is defined as \( f(x; \sigma) = \frac{x}{\sigma^2} \exp(-\frac{x^2}{2\sigma^2}) \)). Hence, the correlation coefficient may not be the optimal score in this case. One alternative would be to apply a maximum likelihood (ML) approach where the noise is assumed Rayleigh distributed. However, the Rayleigh distribution is skewed, and it becomes almost flat as \( \sigma \) increases. The approach would thus be more sensitive to mismatching pixels in the template shadow region than in the highlight region.

This problem can be avoided with a Bayesian method (based on Groen et al. [4]). Let \( x \) and \( t \) denote an image and a template with pixels \( x_i \) and \( t_i \) \((i=1,\ldots,n)\). The vector \( \theta \) is a possible segmentation of the image into echo (e), shadow (s) and background (b) regions, and the vector \( \delta \) is the segmentation the template. Given a utility function defined as \( u(\theta,\delta) = n^{-1} \sum_{i=1}^{n} v(q_i,d_i) \) where \( v(q_i,d_i) = 1 \) if \( q_i = d_i \) and 0 otherwise, the expected utility is (see e.g. [5])

\[
E_{\theta|x}(u(\theta,\delta)) = \frac{1}{n} \sum_{i=1}^{n} p_{\theta|x}(\theta) = \frac{1}{n} \sum_{i=1}^{n} p_{\theta|x}(d_i|x)
\]

where \( p_{\theta|x}(q_i|x) \) is the marginal posterior distribution of \( \theta_i \). In Bayesian estimation \( \delta \) would usually be an estimator, but here it is the segmentation of the template so that the expected utility measures how close the segmentation of the template corresponds to segmentations of the image. In this way, we may use the expected utility as a similarity score. This score requires, however, the marginal posterior distribution. We estimate it with Bayes' theorem by assuming that \( \theta_i \) depends only on \( x_i \) and that the prior distribution is uniform. For the parameter \( \sigma_\theta \) in the Rayleigh pdf, we use the template pixel value if \( x_i \) corresponds to the \( \theta \) region in the template, or the mean value of the \( \theta \) region otherwise. Moreover, we scale the templates values such that the maximum and minimum values in the image and the template are equal (This operation is applied after normalization of the image and the template so that the background values becomes approximately equal to 1).

2.3. Segment Matching

Another approach to template matching is to compare segments derived from the image and the template and ignore the actual amplitude levels. This approach is considered by Fawcett and Myers [3]. They divide the image and the template into highlight and shadow segments. From these segments they compute several features that measure the portion of shadow and highlight pixels in the image that match shadow and highlight pixels in the template. In order to compare this approach to those described above we combine these features into one score which is defined as \( m = \frac{|I_s \cap T_s| \cup (I_s \cap T_h)|}{|I_s \cup I_h \cup T_s \cup T_h|} \). The sets \( I_H \) and \( I_S \) contains the highlight and the shadow pixels in the image, and the sets \( T_H \) and \( T_S \) contains the highlight and the shadow pixels in the template.
3. RESULTS

The methods were tested on both real and synthetic images. The real data sets were obtained with the HiSAS sonar onboard HUGIN 1000-MR in a location outside of Horten in Norway. An MP80 exercise mine and a Manta dummy were deployed as test objects.

The simulator SIGMAS was applied to create templates and simulated data sets. The simulated data sets contained images of a cylinder mine, a Manta mine, a Rockan mine, a sphere, an oil drum, a tractor wheel, a tire and a rock. Templates were only created for the cylinder and the Manta where the altitude (10-50m, step=5m), the range (40-200m, step=5m) and the aspect angle of the cylinder (0-170 degrees, step=10 degrees) were varied. More details on SIGMAS can be found in [2;4].

3.1. The effect of the background

The two first scores included the background, while the last did not. This gave them a disadvantage as the background will typically contain more pixels than the highlight and the shadow regions so that it may dominate the score. Moreover, the background may contain objects that should not be considered and may thus reduce the score. Thus, we examined the performance of the two first scores when the background was included and when it was not. In order to remove the background we segmented the template and image and computed the score over their combined highlight and shadow areas.

The effect of removing the background is shown for the correlation coefficient and the expected utility in Fig. 1 on simulated data with 3 different bottom topographies (flat, ripple and complex seabeds). The removal of the background gave in all cases a better performance. The performance gain was especially high for ripple and complex seabeds. For correlation without background the performance was slightly worse for ripple seabeds than for the other seabed topographies. This reduction was due to the segmentation algorithm as this would sometimes include the ripples in the highlight and the shadow area. A similar reduction was also apparent for the expected utility, but in this case the performance was also worse for complex seabeds. This effect may be due to the scaling of the template values that may not have been robust to the background.

We performed the same analysis on real data and obtained much better performance without the background also in this case. However, we observed that the correlation coefficient could give higher values to some non-targets if they had a highlight, but no shadow. In these cases, the image highlight covered the template highlight region, and most
of the image background pixels were covered by the template shadow region. Thus, the image background would take the role of the shadow.

3.2. Discriminative ability

We examined the discriminative ability of each score on the MP80 and manta templates. The results for the real data are shown in Fig. 2. The correlation coefficient achieved the best performance in most cases with the expected utility score as number two. Segmentation-based matching yielded the worst performance. One exception occurred with the manta on the real data where the expected utility score outperformed the correlation coefficient. However, this data set contained only 8 manta instances so that it is not possible to draw any firm conclusions from this result.

3.3. Robustness

The performance of the scores was also examined with respect to several environmental parameters. In order to control these parameters, we created images with the simulator where these parameters were varied at random. The parameters were the seabed type, the seabed slope and the burial level.

Different seabed types yield different backscatter strength, which may influence the performance. However, correlation worked well on all seabed types, and there was no visible difference in performance between them. The performance of the segment-based matching varied slightly with the seabed, but the performance was not related to the difference between seabed type used for the templates (clay) and the image seabed type. This effect was probably due to the other parameters that were varied. For expected utility it was necessary to scale the template highlight and shadow values in order to make the method more robust (as mentioned in Sec. 2.2).

The seabed slope affects the shadow. Thus, we varied the slope of the seafloor between -20 and 20 degrees. Correlation decreased with increasing absolute slope. A similar reduction did not occur for non-targets so that the discriminative ability decreased. This was noticeable for slopes with absolute value larger than 5 degrees (see Fig. 3a). We observed similar reductions in performance for the other scores as well.

The objects on the seafloor may become partially buried, and this may influence the highlight and the shadow area in the image. We examined the effect by selecting a burial level between zero and half the height of the object. The performance of correlation decreased with increasing burial level as shown in Fig. 3b. Similar results were also obtained for the other scores. Hence, none of the methods was robust to this parameter.
4. CONCLUSION

We have examined the performance of three scores for measuring the similarity of an image and a template. Our results suggested that correlation was the best score. The expected utility outperformed segment matching, which seemed to be very dependent on accurate segmentation. Since the accuracy of the segmentation cannot be assured, the other two scores should be preferred as these seemed more robust to segmentation errors.

The results implied that it was important to remove the background before the correlation coefficient and the expected utility were computed. However, this strategy could fail in some cases when the shadow was too weak to be segmented. Thus, it may be necessary to detect this situation and apply another score, which only considers the echo.

The scores were robust with respect to the seabed type, but the seabed slope and the burial level had a negative impact on the performance. Thus, a template database should include templates over a range of these parameters. This would increase the number of templates that must be matched to each image, but this can be avoided if the parameters can be estimated from bathymetric information. We will examine this in our future work.

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STUDY OF ROBUSTNESS OF SINGLE-ELEMENT-MIRROR TIME REVERSAL METHOD TO REVERBERATION AND CLUTTER: SIMULATIONS AND TANK MEASUREMENTS

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Abstract: Previous results suggest that iterative time reversal (TR) using a single element mirror is a promising, simple and inexpensive means to enhance the backscatter signature of elastic objects and to simultaneously focus on its dominant resonance frequency. Motivated by these findings, this work presents a systematic explanation of the mechanisms thought to be responsible for the TR signal-to-noise ratio (SNR) enhancements under clutter and ambient noise limited conditions.

When clutter or reverberation limited, it is shown that in the frequency domain the magnitude of the TR echo spectrum after \( n \) iterations is simply a product of successive \( n+1 \) clutter and target echo transfer function measurements. Taking the logarithm of this product allows the interpretation of TR as an incoherent average of logarithms of spectral magnitudes and simplifies analysis of its performance as a resonance detector. The calculation of its asymptotic relative efficiency shows that TR suffers only a small loss (~2.2 dB) relative to optimum square-law integration when the signal is weak. In the case when ambient noise limited, the TR amplification and focusing mechanisms and SNR gains are explained by a simple non-linear recurrence equation approximation.

The method was also experimentally applied to scattering measurements from a flush buried spherical shell in sandy sediment in presence of clutter. These results motivate that single channel TR is potentially a useful technique for enhancing backscatter signatures in clutter and noise.

Keywords: Time reversal, resonance enhancement, reverberation suppression
1. INTRODUCTION

Iterative Time Reversal (TR) using an array of several transceivers (transmitter-receiver unit) to receive, reconstruct and finally reverse-propagate wavefronts of interest has been successfully applied to the detection of scatterers in highly reverberant environments [1-3]. Iteratively the method converges to the eigenstates of the TR operator associated to a certain scatterer. Recent numerical and experimental results [4,5] showed that iterative time reversal with a TR mirror reduced to a single element is a promising, simple and inexpensive means to enhance the backscatter signature of a man-made, elastic object and to simultaneously focus on its dominant resonance frequency.

Mathematically, the Fourier transform \( Y_{k}(w) \) of the received echo after \( k \) single-channel TR iterations is

\[
Y_{0}(w) = S(w) \cdot (H(w) + C_{0}(w)) + N_{0}(w)
\]

\[
Y_{1}(w) = S(w) \cdot (H(w) + C_{1}(w)) \cdot \alpha Y_{0}^{*}(w) \|Y_{0}(w)\| + N_{1}(w)
\]

\[
\vdots
\]

\[
Y_{k}(w) = S(w) \cdot (H(w) + C_{k}(w)) \cdot \alpha Y_{k-1}^{*}(w) \|Y_{k-1}(w)\| + N_{k}(w)
\]

where \( H(w) \) is the scatterer transfer function and \( C_{k}(w) \), and \( N_{k}(w) \) are Fourier transforms of the \( k^{th} \) measurement of clutter and ambient noise respectively. The scale factor \( \alpha \) and division by \( \|Y_{k}(w)\| \) represent the gain control and normalization of the received waveform that a real sonar system must perform prior to re-transmitting it.

An important question concerning single-channel time reversal is its potential for enhancing (relative to one ping conventional processing) the signal-to-noise ratio (SNR) in terms of the minimum SNR needed to reliably detect a resonance. Earlier work by Pierson [6] and Waters [7] using computer simulation analysis showed SNR enhancements against noise occurring, but the specific mechanisms responsible were not identified. Motivated by these results, this work presents a systematic investigation based on an analytic formulation and statistical analysis of the potential gain that an iterative single-element TR mirror approach can provide under either reverberation-limited or noise-limited conditions, compared to conventional approaches.

The method was also experimentally applied to scattering measurements by man-made, elastic objects flush buried in coarse sandy sediment in presence of clutter, namely under reverberation-limited conditions. This scaled experiment was conducted in a water tank in the broadband frequency range 200-700 kHz at the LMA laboratory, Marseille, France, in 2008. The experimental results obtained show the ability of the iterative, single-element TRM method to distinguish between the target (namely a hollow, steel, spherical shell) and clutter, and the SRR achieved after a limited number of iterations.

2. CLUTTER LIMITED CASE

When the measurement noise is dominated by bottom clutter or reverberation, i.e. \( \|C(w)\| >> \|N(w)\| \), the time reversal iterations in the frequency domain become
where the initial measurement is \( Y_0(w) = (H(w) + C_0(w)) \cdot S(w) \). It is easy to show that after \( n \) TR iterations, the echo spectral magnitude \( |Y_n(w)| \) is a product

\[
|Y_n(w)| = \beta_n \cdot |S(w)| \cdot \prod_{k=0}^{n} |H(w) + C_k(w)|
\]

(2-2)

of the magnitudes of successive \( n+1 \) clutter-corrupted transfer function measurements \( H(w) + C_k(w) \) where \( \beta_n \) represents a scaling introduced by the sonar’s transmitter and receiver gain control. Taking the logarithm of formula (2-2) (there is no loss in information since the logarithm is a monotonic function) now allows the interpretation of the right-hand part of

\[
\log|Y_n(w)| = \log \beta_n + \log |S(w)| + \sum_{k=0}^{n} \underbrace{\log |H(w) + C_k(w)|}_{z_n(w)}
\]

(2-3)

as a detection statistic that is an integration or average of the individual log-magnitudes of \( n+1 \) noisy transfer functions measurements.

To better understand the SNR enhancement provided by TR, regard \( z_n(w_R) \) in (2-3) as a test statistic for detecting the presence of an object resonance at frequency \( w_R \) in the presence of fluctuating clutter where the \( C_k(w) \) at frequencies \( w \) are modelled as IID sequences of zero-mean proper complex Gaussian random variables (the magnitude \( |C_k(w)| \) is then Rayleigh distributed). Hansen [8] analyzed the post-detection integration loss of a logarithmic detector for a non-fluctuating signal (constant \( H(w) \) in Rayleigh clutter) by calculating its asymptotic relative efficiency (ARE) [9,10] and showed that its loss was only -2.2 dB with respect to the optimum square-law or energy detector.

Since the ARE is defined as the ratio \( \eta_{TR} / \eta_{e} \) of the number of data samples \( \eta \) needed by each detector to achieve the same operating characteristics asymptotically when the signal is weak [10], the result of Hansen tells us that TR is indeed providing SNR enhancement against fluctuating clutter and furthermore, TR asymptotically under weak signal conditions needs only 1.64\( \eta \) iterations to achieve the same level of performance as a square-law detector that uses \( \eta \) measurements.

Of course, this analysis is not completely realistic since the clutter levels and sonar gain scalings are never known beforehand to allow setting of detector thresholds. If a signal-free clutter reference TR measurement \( \tilde{Y}_n(w) \) can be obtained, say by time gating or using adjacent frequency cells, a scale-invariant test statistic such

\[
g_n(w_R) = \log |Y_n(w_R)| - \log |\tilde{Y}_n(w)|
\]

(2-4)

can be constructed. This statistic is equivalent in distribution to
where $X_k(w_T) = H(w_T) + C_k(w)$ if the target is present and $\widetilde{C}_k(w)$ are clutter-only measurements.

The ARE of the scale-invariant TR detector (test statistic (2-4)) is now calculated with respect to the normalized square-law detector

$$
g_n(w_R) = \sum_{k=0}^{n} \log|X_k(w_R)| - \sum_{k=0}^{n} \log|\widetilde{C}_k(w)|
$$

(2-5)

Recall that the ARE is the ratio of the individual detector efficacies [10] where the efficacy of a detector with test statistic $T_n$ (under appropriate regularity conditions) is defined as

$$
e = \lim_{n \to \infty} \left[ \frac{d}{d \theta} E[T_n \mid \theta] \right]_{\theta=0}^2 / n \sigma_{T_n}^2
$$

(2-7)

where $\sigma_{T_n}^2$ is the variance of the test statistic under the null hypothesis. Using the results of Hansen [8] it is easy to show that the efficacy of (2-4) is $3/\pi^2$. The efficacy of (2-5) can be shown to be $1/2$ by plugging the formulas for the expectation and variance of centrally F and singly non-centrally F distributed random variables [11] into (2-7), differentiating with respect to the centrality parameter and taking the limit. Finally, the ratio of the efficacies or ARE of the normalized TR detector (2-4) is found to be $6/\pi^2$ or -2.2 dB which is the same as before.

2.1. Discussion

Although the ARE analysis shows that TR compares favourably with the optimum square-law detector, a limitation of the ARE is that it is primarily informative about a detector’s large sample weak-signal performance. To provide further intuition, a computer simulation analysis was performed using the complex Gaussian clutter model described earlier comparing the scale-invariant TR detector (2-4) against the optimum normalized square-law detector (2-6), a normalized coherent detector (coherent average of pings), and a one ping square-law detector by evaluating the probability of detection versus SNR at a false alarm rate of .005 (see fig. 1). At a probability of detection of .8, scale-invariant TR provided SNR gains of 6.4 dB and 7.3 dB over single ping processing for 15 and 30 iterations respectively. However, coherent processing outperforms TR. These results also confirm that scale-invariant TR performs nearly identically to the normalized square-law detector even at small numbers of iterations and provides gain over single ping square-law detection, thus verifying the ARE prediction.
Fig.1: Probability of detection vs. SNR evaluated at a false alarm rate of .005 obtained from 15000 independent simulation trials. The solid blue line is scale-invariant TR, the dashed red line is the normalized square-law detector, the black dotted line is normalized coherent detector, and the green dash-dot line is the normalized square detector using only a single ping measurement. (a) 15 time reversal iterations. (b) 30 time reversal iterations.

3. AMBIENT NOISE LIMITED CASE

In the ambient noise limited situation, i.e., $\|N(w)\| >> \|C(w)\|$, the TR iterations in the frequency domain take on the form of a non-linear recurrence equation

$$Y_m(w) \approx H(w) \cdot \alpha Y^*_m(w) \|Y_{m-1}(w)\| + N_m(w)$$  \hspace{1cm} (3-1)

The non-linearity from the normalization in (3-1) makes direct analysis of the convergence properties difficult. To simplify matters, consider an idealized case where the iterations are performed using M point discrete Fourier transforms (DFTs) $\tilde{Y}_m$ of the measurements and the DFT of the target’s transfer function has the form $\tilde{H} = [00\ldots\lambda\ldots00]^T$, that is, excites only a single DFT bin. As before, the objective is to detect the presence of a resonance in white Gaussian noise, therefore its bin location is assumed known. When $M$ is large the Frobenius norm in (3-1) can approximated as ($y_m$ represents the DFT bin at the location of the resonance)

$$\|\tilde{Y}_{m-1}\| \approx \sqrt{(M-1)\sigma^2 + |y_{m-1}|^2}$$  \hspace{1cm} (3-2)

where $\sigma^2$ is the DFT bin noise variance. Using approximation (3-2) in (3-1) we obtain

$$y_m \approx \frac{\alpha}{\sqrt{(M-1)\sigma^2 + |y_{m-1}|^2}} y^*_m + n_m$$  \hspace{1cm} (3-3)
This is still difficult to analyze, but useful insights into the convergence properties of (3-3) and (3-1) can be obtained from the ratio of successive iterates

\[ r_m = \frac{y_m}{y_{m-1}} \alpha \sqrt{(M-1)\sigma^2 + |y_{m-1}|^2} + n_m \]  

(3-4)

or more simply from its conditional expectation with respect to \( n_m \) given \( y_{m-1} \):

\[ R(y_{m-1}) = E\left[ \frac{y_m^2}{y_{m-1}^2} \right] = \frac{\alpha^2 - |y_{m-1}|^2}{(M-1)\sigma^2 + |y_{m-1}|^2} + \frac{\sigma^2}{|y_{m-1}|^2} \]  

(3-5)

Formula (3-5) represents the gain or amplification that TR provides over the previous iterate. When \( R > 1 \), then on average the power of the next iterate increases whereas if \( R < 1 \), the power must decrease on average.

To demonstrate these characteristics, formula (3-5) was evaluated using a unit signal as the initial excitation and plotted in fig. 2a as a function of \( |y_{m-1}| \) for parameters \( \alpha = 25 \), \( \lambda = 1 \), \( M=100 \) and \( \sigma^2 = 1 \) (these values of \( \lambda \) and \( \sigma^2 \) provide an initial SNR of 0 dB). In this example as long as \( |y_{m-1}| < 22.96 \) is satisfied, the sequence \( y_m \) on average should grow in magnitude and converge since \( R > 1 \). For small values of \( |y_{m-1}| \), the convergence rate should be rapid since the amplification \( R \) is large. However, when \( |y_{m-1}| \) exceeds 22.96, \( R \) becomes less than 1 and the sequence must decrease in magnitude. This implies that after convergence \( |y_{m-1}| \) is approximately bounded by the point \( p \) where \( R(p) = 1 \). This bounding behaviour and rapid convergence are observed in fig. 2b by plotting a realization of \( |y_m| \) obtained from a single computer simulation.

The non-resonance DFT bins are unaffected by the TR and are merely sampling the ambient noise with variance \( \sigma^2 = 1 \) at each iteration. Therefore the focusing on the resonance by TR provides an SNR increase by 22.96 over the single ping SNR of one.
Although this analysis was done for an idealistic single bin resonance, it nevertheless sheds light on the amplification and focusing properties of single channel TR and its convergence rate. For example, the final amplification \( p \) at convergence can be determined by solving \( R(p) = 1 \) using (3-5).

4. TANK EXPERIMENT AND RESULTS

The objective of this tank experiment was to examine the signal enhancement properties of single-element mirror iterative TR in the presence of clutter from realistic sediment. The target used was a void, steel, thin-walled spherical shell that was flush buried in highly reverberant sediment (due to surface roughness and buried stones). The sphere has a relative thickness \( b/a \) of 96% and an outer radius of 1.5cm. When loaded by the tank sediment it is expected to have \( S_0 \) Lamb-type resonances around (140, 195), 250, 305, 360 and 415 kHz. The \( A_0 \) Lamb-type wave is expected to be evident beyond 500 kHz where, though, the loss due to propagation in the sediment becomes strong. As source and receiver, two identical transducers were used one close to the other in quasi-monostatic configuration. This configuration is acceptable due to the relative large radiation beam that the sphere backscattering is expected to have in the selected band. The source has 400 kHz of bandwidth at -6 dB, centered at 500 kHz. The 3 dB beamwidth is about 7° at the center frequency; hence it is narrow enough to avoid multipath reflections from the tank interfaces.

![Fig.3: Iterative TR process (tank data). (a) Time series. (b) Target-to-reverberation ratio vs. iterations in decibels (dB). (c) Reverberation spectrum. (d) Sphere spectrum.](image)

Both source and receiver were tilted at about 35° in order to insonify the sediment at a supercritical angle (the sediment consisted of a very fine sand of mean grain size of 250 \( \mu \)m with a critical angle estimated to be around 30°). The horizontal range was about 40 cm. The system was calibrated in order to estimate the filter to apply for the equalization of the received signal at each TR iteration.

The iterative active TR process consisted in transmitting first a Ricker pulse centered at 500 kHz, receiving the response (iteration 0), selecting a time window centered on the buried...
target, equalizing, normalizing and finally retransmitting it time-reversed. The same procedure was applied to the same time window at each reception (iteration 1 and following). Only a limited number of iterations was possible as the procedure was not fully automatic. The received time signals from iteration 0 to 11 are shown in Fig. 3a where the red lines bound the TR window including the sphere response, while the black lines bound a window of the same duration including reverberation and taken as a reference to determine the gain achieved in term of SRR. Figures 3c-d shows the reverberation and sphere spectral response as the iteration number increases, while Figure 3b plots the gain in SRR. After 11 iterations, the SRR has increased to about 10 dB.

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UNDERWATER DETECTION, CLASSIFICATION AND TRACKING USING WIDEBAND SONAR

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Abstract: The Ocean Systems Laboratory is developing bio-inspired wideband acoustic sensing methods for underwater target detection and tracking. In this paper we explore what we expect to gain from wideband sonar used alone or in combination with ubiquitous sidescan and forward-look imaging sonars. The wideband sensors themselves are based on bottlenose dolphin sonar, covering a frequency band from around 30kHz to 150kHz and having a frequency dependent beamwidth considerably larger than conventional imaging sonars. The entire system is relatively compact and is suitable for mounting on a variety of platforms including small scale autonomous underwater vehicles (AUVs). In contrast to high resolution image processing techniques which we might apply to sidescan or forward-look sonar returns, detection and classification in the wideband system are based on pattern recognition methods applied to the echo spectra. We demonstrate the principles using example returns from multisaspect data for a range of natural and manmade target types and further responses for a variety of underwater cables measured in free water and against fine and coarse sediments. These show how features extracted directly from the echo spectra can be integrated over multiple pings in a hidden Markov model (HMM) framework to provide discrimination capability between targets and to provide further information such as estimates of surface roughness or target orientation relative to the sonar platform.

Keywords: wideband, sonar, autonomous vehicles, detection, classification, tracking
1. INTRODUCTION

The Ocean Systems Laboratory (OSL) is developing a bio-inspired wideband sonar system for autonomous underwater vehicles (AUVs) for improved detection and recognition of subsea objects [1]. Current research applications for the new system include multiaspect target recognition and the autonomous tracking of underwater cables. The wideband approach offers potential for improved detection and recognition from differences in targets' spectral responses and is expected to provide more robust tracking for low target strength, small diameter cables in cluttered environments. In this paper we present findings from research in which the bio-inspired system has been tested against a number of different targets and cables under various test conditions. These experiments aim to demonstrate the validity of the wideband approach in target and cable recognition and to explore the impact of reverberation and consistency of wideband responses measured against different background sediments.

2. BACKGROUND

Dolphins are known to have excellent capability in many of the tasks we would like to perform using manmade sonar. These include two general applications examined in this paper, target recognition and tracking.

The wideband sensors used in this research are based on bottlenose dolphin (Tursiops truncatus) sonar, covering frequencies from 30-130kHz with a relatively wide beam, varying from 40° at the highest frequency to 80° at the lower end. Two projectors, each covering around two octaves, used in conjunction are capable of emitting significant energy in a band ranging from 30-200kHz. The higher frequency unit has both peak response and -3dB band centre at around 100kHz and has proved most effective in cable discrimination. A set of six short duration bio-inspired wideband pulses comprising pairs of downchirps is used. Labelled DC1–DC6, only the rates of change of frequency differ between signals. As previously reported these allow us to mimic much of the spectral variation in recordings from dolphins performing target recognition tasks [2].

3. TARGET RESPONSES

In using wideband sonar we are interested in recognition from differences in spectral content between echoes from different targets. Classification follows from the analysis of discriminant spectral features, and we have found notch features provide the clearest solution in the relatively simple task of identification of strong 'broadside' target echoes. For low target strength targets or for targets insonified 'off-broadside' we need a new approach and a more adaptable system, capable of integrating information over a series of returns.

3.1. Spectral Features and Classification Models

In recent work we have derived various features from the echo data, including 'spectral texture' features based on Haralick's cooccurrence feature sets and low order cepstral coefficients [3,4]. These features vary reasonably slowly and predictably with changes in target aspect and are easier to track through a feature space than precise locations and
spacings of spectral notches. Moreover we can formulate the analysis to operate through a hidden Markov model (HMM) [5], which allows responses to be tracked through the feature space, providing a more robust classification.

The HMM requires the specification of state transition probabilities and emission probabilities. In our work, state usually relates to target and aspect relative to the sonar and the emission probabilities are the probabilities of generating a particular spectral feature value in each of the possible states. The data sequence is the set of consecutive returns at different target aspects.

In standard notation we denote the state transition probabilities as [6],

\[ a_{ij} = P(\omega_j(t+1)|\omega_i(t)) \]  

and the emission probabilities as,

\[ b_{jk} = P(\nu_k(t)|\omega_j(t)) \]  

where, \( \omega_j(t) \) indicates the system is in state \( j \) at time \( t \) and \( \nu_k(t) \) indicates the observation of symbol (feature value) \( k \) at time \( t \). The \( a_{ij} \) are determined by changes in target aspect relative to the sensor and may be related to platform motion. The \( b_{jk} \) are typically modelled as beta distributions using feature sets normalised to the range [0:1].

### 3.2. Finite Target Responses

Whether considering spectral notches or spectral 'texture' measures, the echo response is determined by a combination of interferences between scattering points on the target and the spectral content of the insonifying pulse.

#### 3.2.1. Cylinders

Cylinders provide a good reference target which can be modelled effectively. The broadside echo is modelled using different methods dependent on the impedance mismatch between the wall material and the surrounding water. High impedance thin shell cylinders are well described by models accounting for Lamb wave propagation through the shell which enters and is back diffracted at the critical angle determined by the material's acoustic properties. For low impedance materials a resonance model based on internal reflections and re-emanation of energy which passes into the interior of the cylinder work well. Off-broadside the situation is quite different and by far the major contributors to the back propagated echo come from the 'corners' defined by the cylinder cross-section, see Fig. 1.

Fig. 2 shows the temporal response for a 110mm diameter PVC pipe over a 360° rotation. The principal peaks around 0°, 180° and 360° correspond to 'broadside' returns. In between a figure-of-eight pattern is seen due to rotation of the main scattering points. The corresponding frequency responses for the target are given in Fig. 3, illustrating the strong variability in the off-broadside spectra. These spectra still contain strong notch features, but differentiation between different targets is rarely possible from single pings.
Tracking of the scattering points in the time domain is possible and can give good information on target sizes which may support target recognition coupled with a characteristic 'broadside' response. In the frequency domain, however, variations can be tracked using the spectral features described above. Whilst we still require multiple pings over a range of target aspects, it is no longer essential to find a 'broadside' return and in midwater trials we have demonstrated excellent discrimination for a set of manmade and natural objects integrating responses over a $25^\circ$ sector. In these tests likelihoods derived from the HMM have shown that the cooccurrence features have some potential to group like objects together. Further experimentation is ongoing to investigate this ability to generalise more fully. Since classification is achieved by integrating returns over a series of aspects, a useful by-product of the HMM is that it provides an estimate of the target aspect relative to the sensor platform.

1.1. Cable Responses

The cables of interest are relatively small diameter communications cables. Examples of their internal structures are given in Fig. 4. The cables are typically deployed without an armoured sheath and their diameters vary between 14.5mm and 32mm giving low target strengths compared to typical MCM targets.
Measured multiaspect cable responses are consistent to $\pm 15^\circ$. Beyond $\pm 10^\circ$ the higher frequencies are lost progressively due to the frequency-dependent nature of the transducer beamwidths. The wideband responses are once again characterised by spectral patterns derived from interferences between overlapping target echoes. For the simpler cable structures, notch positions can be predicted using a thin cylindrical shell model. Since the outer plastic sheath has relatively low impedance (close to water) some sound is reflected at this layer, but much of the energy is transmitted through to the higher impedance copper layer. Sound enters this layer at the critical angle $\theta_c$, propagates around the metallic shell and is back-diffracted at the same angle $\theta_c$, see Fig. 5.

These phenomena are the $S_0$, symmetric, and $A_0$, anti-symmetric, Lamb waves [7]. At biosonar frequencies the $A_0$ wave is subsonic and only the $S_0$ need be considered. The delay between specular and secondary echoes can be calculated given sound speeds in the water and the copper layer, eqn. (1),
\[ \Delta t_n = 2r \left( \frac{n \pi - \theta_c}{v_g} - \frac{1 - \cos \theta_c}{c} \right) \]  

(1)

where, \( n \) represents the number of turns around the cylinder, \( r \) is the radius of the copper layer and \( v_g \) is the group velocity in the copper. For the type C cable: \( \Delta t_1 = 21 \mu s \) and \( \Delta t_2 = 43 \mu s \). The empirically measured notch spacing is around 23kHz, equivalent to a time delay of 43.8\( \mu s \), corresponding well with the predicted \( \Delta t_2 \) value.

Fig. 6 gives clear water spectral responses for each of four cable types. Type D has a 17mm external diameter and is similar in construction to type A depicted in Fig. 2, but with a fibre optic core. Variations in response are due partly to the spectral shape of the pulse and partly to the echo responses themselves. For the type B cable, the ring of strengthening cables typically produces more oscillatory spectral responses. For cable type C, pulse DC5 provides a good candidate for discrimination from strong oscillations in the 50-80kHz band. For cable type D, consistency between pings is good, but this cable does have the lowest SNR, indicated by the higher noise levels towards 200kHz.

![Fig. 5: Lamb wave propagation path in cylindrical cross section](image)

Fig. 5: Lamb wave propagation path in cylindrical cross section

Fig. 6: Echo spectra for each of four cable types insonified with bio-inspired pulses DC1, DC3 and DC5.
Experiments with cables lying on fine and coarse sediments show that the characteristic patterns can still be achieved, though shallower grazing angles are required for detection and recognition on more highly reverberant surfaces. At a 45° grazing angle, on a coarse sand and grit mixture (particle sizes varying between 0.5mm and >5mm), the main lobe response for the type D cable is completely buried in reverberation noise. At a shallower grazing angle, 27°, the characteristic three-lobed response is still clearly visible in a sequence of 43 DC3 echoes recorded at 5cm intervals along a 2m cable segment, see Fig. 7. In addition to the sediment reverberation, these data are compromised by cable curvature, tank wall returns, disturbances in the sediment surface and ambient noise sources.

Fig. 7: Lamb wave propagation path in cylindrical cross section

2. IMPLICATIONS FOR SURVEY

Various strategies may be proposed to ensure gaining strongly characteristic responses from a min-like object (MLO) with a non-axially symmetric response. For maximum certainty a circular reacquisition strategy would ensure all characteristic responses were met, Fig. 8.
Fig. 8: Target acquisition strategies: a) circular pattern ensures all characteristic responses are found; b) the wide beamwidth of the sensor ensures a good range of multiaspect returns are available with a straightline trajectory past the target.

However, the HMM approach will also allow for rejection of many false alarms directly from single pass transits in a conventional ‘lawnmower’ survey pattern. In both of these target acquisition scenarios a vehicle speed of 2 knots and a pulse repetition frequency of 10Hz would give better than 0.5° angular variation between pings at a range of 20m, similar to the data gathered during the experimental trials.

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REFERENCES


Structured Session 10

Synthetic Aperture Sonar: State-of-the-art

Organizers: Roy Edgar Hansen & Peter Gough
DETECTION RATE STATISTICS IN SYNTHETIC APERTURE SONAR IMAGES

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Abstract: Synthetic aperture sonar (SAS) has proved to be successful for mine hunting and is now robust for generating high-resolution images over wide swath. The subsequent step in the processing is detection, discriminating between mine-like and non-mine-like objects, which is designed to minimise the number of missed mines so that the system can manage the detection rate. Statistical analysis using SAS has been limited, because operational use of the technology is at an early stage. The design of automated detection and classification systems depends however on these statistics, which for a SAS are environment dependent. NURC has collected a comprehensive data set off the coast of Latvia with the MUSCLE SAS, which comprises a wide range of seabeds, clutter and vehicle motion. The statistical analysis is based on 50 km² of SAS images at centimetre resolution.

Keywords: Synthetic aperture sonar, detection, mine hunting, statistics

1. INTRODUCTION

Fig.1: Colossus 2 trial off the coast of Latvia with predominant system MUSCLE.
Modern mine countermeasures (MCM) sonars have recently improved imaging resolution to such an extent that the traditional approach of post-processing is to be reassessed. The most successful methodology that has been achieving this resolution improvement is synthetic aperture sonar (SAS). The last decades this technology has been researched and is now well-understood in terms of requirements, robustness and sensitivity to environment and geometry. Operational use is limited however to a few countries.

This article focuses on a part of the signal processing chain that comes with SAS systems employed for MCM purposes. The signal processing chain roughly consists of imaging, followed by detection, classification and localisation. The part of interest in this article is the detection phase. In April 2008, NURC conducted a sequence of experiments off the coast of Latvia, which is illustrated in Fig.1 and resulted in a large dataset collected with the MUSCLE AUV equipped with a SAS operated around a frequency of 300kHz.

The outline of the article is as follows. Section 2 describes the dataset and its features. The detector is explained in Section 3, followed by the corresponding statistical analysis in Section 4. Section 5 discusses the impact on the MCM system and conclusions follow in Section 6.

2. DATASET OF SAS IMAGES

During the Colossus 2 trial mentioned in the previous section experimental data were gathered in order to support R&D on signal processing techniques and to establish the potential of AUVs with a high-resolution SAS. The signal processing techniques of interest are SAS imaging, automated detection and automatic target recognition. Owing to the excellent weather, the sea trial resulted in a total of 24 AUV missions in the areas B, C and D. The respective areas provided 12 (D), 21 (C) and 22 (B) km² of SAS images.

![Fig.2: SAS images with a cylinder on different seabeds. Image dimensions are 40 m along-track and 100 m in slant range starting at 50 m and colour range is 40 dB.](image)

Fig.2 gives an idea of the variety of images collected. From top to bottom the images are representative for area D, area C and area B, respectively. The images are normalised with a median filter so that range dependent echo strength due to the vertical beampattern and propagation loss is removed. In each area, AUV surveys were conducted and provided a
comprehensive MCM dataset that contains a large number of bottom types, false alarms (man-made and natural), and actual mines. In addition seven target shapes were deployed in each of the areas, and those were surveyed to obtain SAS images at different aspect angles and ranges. One target shape visible in the images in Fig.2 is an MP80 cylindrical mine shape. In area D often trawl marks were visible, and objects on the seabed are usually surrounded by a shadow and a patch of brightly scattering seabed, which indicates that the object is in a scour pit and has increased seabed roughness locally. Area C is a rather challenging area for MCM, especially when the targets are hidden within the ripples. In particular the detector phase is difficult when irregular ripples are present. Area B can be regarded most benign for MCM with SAS. The sonar signal correlation is always good, which results in constant good imaging performance. In addition to this, highlight and shadow are sharply defined on such seabed.

GPS positions of the targets were recorded during deployment and the targets were deployed far from other unknown objects for safety reasons. This enables ground truth on the targets: detection close to the recorded deployment can be concluded to be correct.

2.1. SAS issues affecting imaging and detection robustness

Fig.3: Three cases where detection of mines with a SAS is a challenge; from left to right: low scattering energy, sonar instability and high contact density. Dimensions are 50 by 100 m starting at 50m and colour range is 40 dB.

SAS performance degradation can have different causes, of which some were observed in this trial dataset. The most significant effect that all these causes have is an overall decrease in ping-to-ping correlation, leading primarily to difficulties in the sonar motion estimation and secondarily to integration of incoherent data. Both damage image quality. One of the causes is multi-path, especially second-order multi-path, which can have a rather destructive impact on the SAS image. In the dataset of this paper very few problems from multi-path were found owing to the relatively deep water of over 30 m and the narrow vertical beam design of the SAS. Another source that was observed was the influence of bottom type. In area D, consisting of mud, sometimes lack of signal occurred at long range, which causes low signal-to-noise ratio, in turn leading to low correlation, which can be seen in the left image of Fig.3. However, the most frequent effect in this dataset was sonar instability, which occurred at the beginning of a leg or when current was significant. When the vehicle is yawing within the SAS integration time some patches of sea floor are not insonified enough. Considering that the horizontal beamwidth is 6° and that, at times, vehicle yaw variation was more than 3°, it is no surprise that image quality then suffers. Nevertheless, it should be mentioned that none of these effects occurred very often and almost all images analysed were of good quality.

Regrettably, good SAS image quality is not the end of the story and no guarantee for target detection, which is also environment dependent. A part of area C with many pebbles and boulders is visible in the most right image of Fig.3. Even if the imaging characteristics are rather benign, the detector is facing many items that are typical for mines such as highlight and shadow.
3. DETECTOR

A detector of mines in SAS images aims to distinguish between mine-like objects and non-mine-like objects. Since this is not the end station in the processing chain, the design of the detector can be regarded as merely a data reduction. For an automated system, the data reduction requirement is given by the next phase, the classifier. As long as the classifier, which may include better discriminative techniques [1], can handle the detector output, the system is well-designed. For this reason, it is important to keep the detector threshold low so that enough real mines pass. The detector itself should also be relatively simple, because it should not have a very strong discriminative behaviour (rocks of mine size are required to pass) and the detection algorithm has to be applied to the full dataset (22 Gigabyte per hour in this case). It is regarded as a first rough sifting of the data.

3.1. Approach and results

The core part of the detector analysed in this article is based on convolution with a template. Not much a priori knowledge or assumptions about the target are inserted at this stage. The template consists of a block of highlight (+1) and a block of shadow (-1) as described by [2]. The size of the template highlight area was set to 1m (along-track) by 0.5m (range). The width of the shadow was set to 0.5m and the (range-dependent) shadow length was computed with the altitude $a$ and object height $h$. Before convolution with this template the SAS image is first normalised as described in [2]. After detection the scores also undergo a further normalisation step in order to make the statistical properties of the detection scores the same. When computing mean and variance of the detection scores, a range dependent trend appears. This trend is removed based on all the images, mean and variance are not range dependent after this step.

Fig.4: Detection results in area B with 12 detections (3 by 7 m) of the same cylindrical mine (left-hand side) and a detection map of the total area (right-hand side). Colour coding in the detection map is logarithmic with a dynamic range of 70 dB.

Fig.4 shows twelve detections around the position of the cylindrical mine. The image snippets of 3 by 7m are stored with location information and detection score. The detection map of area B after eight missions is shown in addition.
4. MEASURED DETECTION STATISTICS

The detector described in Section 3 was applied to the complete database of about $10^5$ measured SAS images of dimensions 50 by 110 meters. The detection threshold was kept relatively low in order not to miss actual targets and to analyse higher thresholds as well. It is impossible to truly analyse false alarm rate in this dataset, because the ground-truth is incomplete. Detected objects can be false alarms, but can also be actual mine-like objects. However, since the target-deployment positions are known, we focus on the detection performance in terms of detection density $\rho$ versus probability of missing a target $P_{\text{miss}}$. The dependence of these two parameters on bottom type and grazing angle is investigated.

The trial enabled detection assessment on bottom types different in nature as suggested by the top panel of Fig.5. These five types aptly covered the variability and are categorised as follows. Bottom type 1 is benign, flat and muddy, typically found in area B. Bottom type 2 consists of sand ripples, found in area C. Bottom type 3 is complex, with patches of sand ripples, pebbles & boulders, found in the northern part of area C. Bottom type 4 is benign, flat and sandy with little texture, found in area D. Bottom type 5 was found in areas C and D and is a mix of sand and small rocks/shells. The seabed was classified with unsupervised seabed segmentation, which resulted in bottom type estimate every 2 m$^2$ of SAS image as described in [4]. Here one estimate per SAS tile, the dominant one, is used.

![Fig.5: Typical examples of bottom types (top panel) and detection performance in the three trial areas (bottom left) and for the five different bottom types (bottom right). SAS image dimensions are 50 m by 50 m and logarithmic colour scale has a 40 dB range.](image)

The left bottom plot of Fig.5 shows the performance of the detector described in Section 3. The probability $P_{\text{miss}}$ is computed by varying the threshold $\tau$ and counting the number of targets with a detection score $\sigma(k)$ below the threshold

$$P_{\text{miss}} = \frac{\sum_{k} I_{\sigma(k) < \tau}}{N_t}$$

(1)
The total number of targets is given by $N_t$ and $k$ is the index for the detections. The contact density that corresponds to these values of $P_{\text{miss}}$ and $\tau$ is computed with

$$\rho = \frac{\sum_k 1_{\sigma(k) > \tau}}{A}$$

The total area is denoted by $A$. The other three curves, for area D, C and B, are computed in the same way but using the subset of detections from each area, which is possible because the targets were deployed in each of the three areas. The results visible with these curves indeed show a rather strong dependence on area. The performance in area C is much worse, which is no surprise after interpreting the examples is Figs. 3, 4 and 5. Whilst contact density (for very low $P_{\text{miss}}$) is below 200 per km$^2$ in areas C and B, it is almost 10 times this value in area D.

The right bottom plot provides a better insight on the detection dependence on bottom type. Overall the areas B, C and D were different, but the bottom-type classes were found to give a more sensible subdivision. However, compared to the left-hand plot, $P_{\text{miss}}$ had to be calculated differently, due to the fact that targets were only deployed on three out of five bottom types. No targets were deployed on bottom type 3 and 5. Instead all target detection scores were used for each of the curves and only the number of contacts corresponding to the bottom type was used. The right-hand side plot reveals an even stronger bottom type effect, with a low contact density ($<200$ per km$^2$) for bottom types 1 and 4 and a factor 30 to 40 worse for bottom types 2 and 3.

MCM detection and classification performance is known to depend on grazing angle $\gamma$. It is beneficial to image the target at shallow grazing angle so that it “sticks out”. At steeper grazing angles highlight-to-reverberation ratio is lower and shadows are smaller. However, performance at shallower grazing angle (long range) is more sensitive due to the following five reasons: (1) SAS resolution gain is required to be higher to maintain constant resolution, (2) SAS becomes more sensitive to motion, (3) bathymetric variations cause increased shadow zones, (4) coherence and energy loss due to longer propagation, (5) multi-path.

![Fig.3: Detection performance versus range where short range is 40-70m, medium range is 70-110m and long range is 110-150m.](image)

Fig. 3 shows that the suspected grazing angle dependence is indeed obvious in this data. During the trial the altitude of the vehicle was kept constant at $a = 13$ m, which enables to compute the grazing angles for these ranges $r$ with $\sin \gamma = a/r$. 

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This means that the grazing angle values are 5-6.8° for the long range set, 6.8-10.7° for the medium range set and 10.7-19° for the short range set. It is better to subdivide into grazing angle windows with equal width to highlight this dependence, but unfortunately here this led to insufficient statistical information in the short range set.

5. IMPACT ON MINE HUNTING AUV SYSTEM

The results presented in the previous section prove the strong influence of bottom type and geometry (grazing angle) on detection statistics and performance. The performance was analysed in terms of contact density, which is an important parameter in further analysis. The contact density can be related to contact frequency $f$ (contacts per hour) and to detector’s data reduction factor $d$. The contact frequency is $f = 2\rho \nu (r_{\text{max}} - r_{\text{min}})$.

The velocity of the AUV is given by $v = 1.5$ m/s, the maximum range $r_{\text{max}} = 150$ m and the minimum range $r_{\text{min}} = 40$ m. The data reduction factor $d$ is given by

$$d = \frac{1}{\rho l w}$$

In this analysis the dimensions of the snippet are set to $l = 7$ m and $w = 3$ m, in range and along-track, respectively. Contact frequency $f$ is an essential parameter when human operators are involved in the classification process; it reveals how much information the operator has to analyse. For computer automated classification the essential requirement is data flow. The classification algorithms are usually sophisticated and computationally expensive, and cannot be applied to the complete data set. The parameters for the different bottom types can be found in Table 1.

<table>
<thead>
<tr>
<th>Bottom type</th>
<th>$\rho$ [1/km²]</th>
<th>$f$ [1/hour]</th>
<th>$d$</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>200</td>
<td>238</td>
<td>238</td>
</tr>
<tr>
<td>2</td>
<td>3800</td>
<td>4514</td>
<td>13</td>
</tr>
<tr>
<td>3</td>
<td>2900</td>
<td>3445</td>
<td>16</td>
</tr>
<tr>
<td>4</td>
<td>100</td>
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<td>476</td>
</tr>
<tr>
<td>5</td>
<td>600</td>
<td>713</td>
<td>79</td>
</tr>
</tbody>
</table>

Table 1: Detection performance for the considered bottom types in terms of contact density $\rho$, contact frequency $f$ and data reduction factor $d$.

From Table 1 it can be concluded that the output data flow varies from 90 Megabytes/hour in the easiest case to 1.75 Gigabytes/hour in the hardest case. It has yet to be verified how much this can be enhanced with an improved detection scheme, which is for instance tuned to discard typical false alarms such as fish and ripples. On the other hand, the requirement for the detector needs to be established. Intuitively, it would seem that a data reduction of around 20 is not sufficient, but it may well be that the automatic classifier can cope with such a dataflow. Visual analysis on the detection snippets does suggest that there is a quick gain expected when further improving the detection algorithm. The detector can easily be tuned to discard typical false alarms such as fish and ripples.
6. CONCLUSION

In this paper detection of mines in SAS images is investigated. In April 2008 NURC collected a comprehensive dataset with variations in bottom type, target type and geometry. This dataset proved useful for analysis of a detector in a statistical manner. The detection algorithm was applied to $10^5$ SAS images, and after thresholding with a very low threshold resulted in a total number of detections of 60,000. The relationship between detected contacts and missed targets was established for the dataset and for subsets of certain bottom type. In addition, a similar analysis was performed to reveal the influence of grazing angle. Both bottom type and grazing angle showed to be important for the detector. These results were shown to have an importance for the design of an automated MCM detection and classification system.

7. ACKNOWLEDGEMENTS

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REFERENCES

IXSEA  Shadows, synthetic aperture sonar and forward looking gap-filler

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Abstract: SHADOWS is a new Sonar system developed by IXSEA SAS. It is composed of a synthetic aperture side-scan sonar and a forward looking sonar and uses a very precise INS. It is imaging at 300m or 10 times the altitude on each side with a 15cm constant resolution and is filling the gap at nadir with side scan like images with comparable resolution. It produces a real time georeferenced mosaic using the position provided by the INS. The working speed is 5 knots. The synthetic aperture sonar algorithm uses INS data combined with the Displaced Phase Center Algorithm (DPC). Differences between INS and DPC navigations can be used to compute a topography profile of the ground. The post processing and real-time modes differ on the weight given between INS and DPC, and on the approximations to be done. The real-time beam-forming algorithm used is the time-domain fast factorized back projection which can be pushed to an exact back projection in the post processing mode. The forward-looking sonar uses a patented sectorized emission architecture. The images are side-scan-like, i.e. we see echoes and shadows as on the sides. The geometry of the antennas is calculated to maximize the contrast on images and to minimize the noise and interferences. The real-time algorithm can be customized to make some incoherent integration on several pings increasing the contrast but slightly decreasing the resolution. A post processing algorithm can also provide an animation on a specific contact on the floor. All the images are presented on a georeferenced map and can be visualized in the NASA World Wind interface. The data can also be imported in the IXSEA web contact analyser which allows doing some detection, classification and post processing work a selected contact images. This system is particularly efficient in shallow water. In this paper, we show some results obtained during sea trials in La Ciotat Bay. We explored different configurations with several water depths. The results we obtained allows us to claim that even in very shallow water the swath can be greater than 10 times the altitude of the fish.
PULSE-COMPRESSION: COUPLING THE MATCHED FILTER AND THE STOCHASTIC MATCHED FILTER

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Abstract: Nowadays, Synthetic Aperture Sonar (SAS) images are of great interest especially for the detection and classification of objects lying on sea bed. Indeed, synthetic aperture methods greatly improve the azimuthal resolution and thus allow obtaining a far better description of the sea floor. In the scientific literature, one can find several algorithms to reconstruct SAS images (Time-domain correlation, Back projection, Fast correlation, Range-Doppler algorithm, Chirp-scaling algorithm ...). Most of them are based on the use of the matched filter theory for the pulse-compression step. This corresponds to a correlation between the echo signal with the emitted pulse assuming therefore a white disturbing noise and a deterministic useful signal. But as the noise is colored and the signal a realization of a random process, we propose in this paper to perform the pulse-compression step using the stochastic extension of the matched filter notion in the echo signal's time-frequency transform domain. Results obtained on real data are proposed and discussed.

Keywords: SAS image, Pulse-compression, Matched filter, Stochastic matched filter, Time-frequency domain
1. INTRODUCTION

Synthetic Aperture Imaging Systems are mature and offer a vast choice of reconstruction algorithms. Most of them use pulse-compression. This step is usually performed as a matched-filtering with the band-pass transmitted signal. Taking into account the main assumptions of the matched filter theory, the use of the band-pass transmitted pulse as matched filter’s impulse response is only available if the useful signal is perfectly known and if the noise is white, which is not the case in practice. For this reason, it has been proposed in [1] to substitute the classical matched filter used for the pulse-compression by its extension to random signals: the stochastic matched filter. Results obtained on real data have revealed the well-founded of such an approach with a signal to noise ratio’s averaging improvement of 2.26 dB for a 0.332 ms duration chirp signal. But due to the stationary assumptions, an important drawback of this method is a slightly range resolution deterioration. In this paper, we propose to improve this technique using the stochastic matched filter in the echo signal’s time-frequency transform domain. First, in section 2, we describe the context and the materials used during our experiments. Next, in section 3, we recall and discuss the classical pulse-compression step used in SAS image formation. In section IV, we present the stochastic matched filter. We finish this article with the presentation of the proposed method and with some results obtained on real data.

2. THE MITOHAMA SEA EXPERIMENT

Unknown underwater areas are still vast and each step forward, teach us how few we know about it. In an era where natural resources are key to a proper evolution, the need to discover new resources is increasing. In this way, the development of a new 3-D SAS system covering large areas while conserving a high resolution is needed. In order to achieve a resolution of a few centimeters, we have developed a new L-Array interferometer. This one presents five hydrophones (see figure 1.a), and thus offers three dimensional restitution. Furthermore, for this experiment, we have used a Reson Multibeam projector and an ADCP (TRDI, WHS-600kHz) (see figure 1.b). Concerning the positioning system, we used a Real Time Kinetics GPS, and a PHINS.

![Fig. 1: 3-D SAS system](image-url)
In addition, we will jointly use the synthetic aperture method. Actually, synthetic aperture method should greatly improve the azimuthal resolution. Yet, using such a method brings a strong speckle noise. Therefore, we will also aim at reducing speckle’s hindrance. However, we wish to fine down each channel before the SAS image formation, so as to reduce the usual post-processing de-noising step.

Our first experiment took place around the city of Mitohama, which is located in the prefecture of Shizuoka (Japan). We worked on a barge (see figure 2) in order to avoid any fluctuation that would arise while using vehicles such as boat, fish, AUV... The barge is situated in an area of 30m depth, whose sea bottom is mainly composed of sand. To have the device moving, we have used a 30m long rail (see figure 2) along one side of the barge, offering a smooth movement of 0.1 m.s⁻¹.

![Fig. 2: Mitohama sea experiment context](image)

Various pulses have been tried. Indeed, continuous waves (CW) of 0.332ms and 0.5ms, respectively, centered at \( f_0 = 110 \text{ kHz} \) and linear FM chirp signals of 0.332ms, 0.5ms and 1ms, centered at \( f_0 = 110 \text{ kHz} \), with a bandwidth \( B = 15 \text{ kHz} \) were used. Those were sent at a frequency of 5Hz with a power of 214 dB and a horizontal beamwidth of 24°.

3. SAS IMAGE FORMATION: PULSE-COMPRESSION

Let \( p(t) \) be the waveform used in the echo imaging system. This band-pass signal can be expressed as follows:

\[
p(t) = \Re\{p_e(t) \exp[j2\pi f_0 t]\}
\]

where \( p_e(t) \) is a low-pass complex envelope signal (either a continuous wave (CW) signal or a chirp signal for our experiments) and where \( f_0 \) designates the carrier frequency (110 kHz for our experiments).

After its propagation, to improve the resolution of the system, the received echo signal is pulse-compressed [2]. This is usually performed as a matched-filtering with the band-pass
transmitted signal. Such an operation strongly improves the signal to noise ratio of the data and gives a final resolution that is proportional to the signal bandwidth. Let \( s(t, y) \) be the pulse-compressed echo signal, we have:

\[
s(t, y) = r(t, y) * p_c(t)
\]

(2)

where \( r(t, y) \) is the raw echo signal and * denotes the correlation in time axis.

As it takes part in most of the SAS image reconstruction techniques (such as Time-domain Correlation \[3\], Back Projection \[4\] …), one should make sure of the reliability of this pulse-compression step, otherwise the range resolution could be deteriorated. As previously explained, the impulse response of the matched filter corresponds to the band-pass transmitted signal itself. This is valid in theory only if, on the one hand, the useful signal is well-known (i.e. deterministic) and, on the other hand, the disturbing signal corresponds to a white noise. But in most cases, the noise can not be considered as white even in the signal bandwidth. Furthermore, as it is necessary, on the one hand, to take into account the Doppler effect and, on the other hand, to consider a random phase due to the acoustic propagation in the transmission channel, the received band-pass signal may be written as:

\[
p_r(t) = \Re\{p_r(t)\exp[j2\pi v_0 t + \phi]\}
\]

(3)

where \( v_0 \) and \( \phi \) are random variables. We can assume that \( v_0 \) is normally distributed with \( f_0 \) as mean value and \( \phi \) uniformly distributed in \([-\pi; \pi]\). Thereby, the useful signal is not perfectly known. For all these reasons, using the band-pass transmitted signal as impulse response for the matched filter is under-optimal. Obviously, it is possible to use a whitening step and some phase and frequency estimators to ensure the matched filter assumptions, but this would involved an increase of the computational complexity, which is not compatible with SAS systems aiming at a reconstruction with a low computational burden. To solve this problem, it has been proposed in \[1\] to replace the matched filter approach by its stochastic extension: the Stochastic Matched Filter (SMF).

4. THE STOCHASTIC MATCHED FILTER

The stochastic matched filter has been first described by Cavassilas \[5\] in a context of detection and since was extended to the problem arising from the restoration of data in disturbed environment. The principle of this filtering method is to expand the noise-corrupted signal into series of functions with uncorrelated random variables for decomposition coefficients and then to reconstruct an approximation of the signal of interest by considering only a few part of these random variables.

Let the observation \( Z(t) \), defined over a real set \( D \), be the additive superposition of two random signals, a signal of interest \( S(t) \) with a disturbing signal \( N(t) \):

\[
Z(t) = \sigma_S S_0(t) + \sigma_N N_0(t)
\]

(4)

where \( E\{|S_0(t)|^2\} = E\{|N_0(t)|^2\} = 1 \) and where \( \sigma_S \) and \( \sigma_N \) represent signal and noise standard deviations. By assumptions, both signal and noise are supposed stationary, independent and at least one of them with zero-mean.
Noise corrupted signal \( Z(t) \) is expanded onto a series of deterministic functions \( \{ \Psi_n(t) \} \):

\[
Z(t) = \sum_{n=1}^{\infty} z_n \Psi_n(t)
\]  

(5)

where \( \{z_n\} \) is an infinite sequence of zero-mean uncorrelated random variables. These variables are determined using the projection of the noisy data onto a basis of deterministic functions, i.e.:

\[
z_n = \int_{D} Z(t) \Phi_n(t) dt
\]  

(6)

The basis is chosen such as the uncorrelation of the random variables is insured. Using this criterion, we can show the set of functions \( \Phi_n(t) \) corresponds to the solution of the following integral equation:

\[
\int_{D} \Gamma_{S,S}(t_1-t_2) \Phi_n(t_2) dt_2 = \lambda_n \int_{D} \Gamma_{N,N}(t_1-t_2) \Phi_n(t_2) dt_2
\]  

(7)

where \( \Gamma_{S,S}(t_1-t_2) \) and \( \Gamma_{N,N}(t_1-t_2) \) are signal and noise reduced covariances. One can show solving this integral equation is equivalent to maximize the observation signal to noise ratio described as a Rayleigh quotient [5]. Next, basis \( \{ \Psi_n(t) \} \) is given by:

\[
\Psi_n(t) = \int_{D} \Gamma_{N,N}(t_1-t_2) \Phi_n(t_2) dt_2
\]  

(8)

In this context, quadratic moment of the \( n^{th} \) coefficient of the observation expansion \( z_n \) (i.e. \( E[z_n^2] \)) is equal to \( \sigma^2 \lambda_n + \sigma^2_N \). Thus, the signal to noise ratio of component \( z_n \) corresponds to the native signal to noise ratio times eigenvalue \( \lambda_n \) (i.e.: \( \sigma^2 \lambda_n / \sigma^2_N \)). With these considerations, a filtered observation can be built by keeping only components associated to eigenvalues greater than a certain level, anyhow greater than one. Finally, let us remark that although the theory has been given for continuous signals, it is easily possible to transcribe it for discrete signals considering for example the continuous signal has been sampled over its definition set \( D \), replacing integrals over \( D \) by sums along all the samples and considering correlation matrix instead of autocorrelation function.

In [1], it has been proposed to substitute the classical matched filter by the SMF. In this context, the purpose is to obtain a strong signal to noise ratio enhancement. The only way to achieve this goal using the SMF theory is to use as impulse response, the eigenfunction \( \Phi_i(t) \) associated to the greatest eigenvalue \( \lambda_1 \). With such an approach and in a case of a 0.332 ms duration linear FM chirp signal, results have revealed the efficiency of the method allowing a signal to noise ratio’s averaging improvement of 2.26 dB compared to the classical matched filter technique. Nevertheless, a main assumption of the SMF theory is the stationary of the useful signal; it means that this signal should present the same frequencies at each moment, which is not the case by definition for a chirp signal. This induces a slightly resolution deterioration (the pulse is less compressed using the SMF than the classical
approach). Indeed, due to the stationary assumptions, the impulse response $\Phi_1(t)$ is stationary, so that the response to a given impulse (at a frequency located in the $\Phi_1(t)$ frequency range) has a duration equal to the one of $\Phi_1(t)$: the pulse energy is spread on impulse response's duration. To solve this problem, we propose to use a time-frequency approach.

5. COUPLING THE MATCHED FILTER AND THE SMF IN A TIME-FREQUENCY APPROACH

5.1. Principle

As the useful signal can not be considered as a realization of a stationary process, we are going to use the SMF considering the time-frequency plane of the noisy data, obtained applying the Gabor transform. For each raw $y$, the Gabor transform of the echo signal is:

$$R_y^{\text{Gabor}}(b, \nu) = \int r(t, y) g(t-b) e^{-2j\pi \nu t} dt$$

(9)

where $b$ is the translation parameter and $g(t) = \pi^{-1/4} \exp\left[-t^2/2\right]$. Next, for each value of $b$, the SMF is applied in the frequency domain and the filtered echo signal $\tilde{r}(t, y)$ is reconstructed by inverse Gabor transform:

$$\tilde{r}(t, y) = \frac{1}{\|g(t)\|^2} \int g(t-b) \int R_y^{\text{Gabor}}(b, \nu) \Phi_1(\nu) e^{2j\pi \nu t} d\nu db$$

(10)

where $\Phi_1(\nu)$ represents the $\Phi_1(t)$ spectrum and $\|..\|^2$ the Euclidean norm. Finally, the pulse-compression step is performed making the cross-correlation between the filtered echo signal and the filtered band-pass transmitted signal $\tilde{p}_e(t)$ (this one is obtained using the same approach than for the filtered echo signal):

$$s(t, y) = \tilde{r}(t, y) * \tilde{p}_e(t)$$

(11)

This last operation corresponds to a matched filtering step with as impulse response the filtered band-pass transmitted signal, the pulse shape being modified by the echo signal’s filtering operation.

5.2. Experimentation

To evaluate the efficiency of the proposed method, we have applied it on a set of data obtained during the Mitohama campaign. For these data, the emitted signal is a 1 ms duration linear FM chirp pulse. The image formed of pulse-compressed data obtained applying the classical matched filter approach is presented on figure 3.a in dB magnitude. Using the
The proposed method gives the result presented on figure 3.b. An analysis of these results reveals that they seem to be quite similar. Nevertheless, if we pay attention to the upper part of each image (for a slant-range taking values between 0m and 35m), we can notice that the use of the proposed method enhances some patterns corresponding probably to bubble signatures and to reverberated echoes on individuals from the marine fauna. Such differences between the two methods reveal that the proposed method improves the matching power to the useful signal. Furthermore, in order to quantify the signal to noise ratio improvement, we have computed the variance in some homogeneous areas of the pulse-compressed data. As the seafloor is flat and mainly composed of sand, we can assume that these areas correspond only to noise, so that lower is the computed variance better is the signal to noise ratio improvement. For these images, we have found that the use of the proposed method allows a signal to noise ratio’s averaging improvement equal to 1.9 dB. So, compared to the approach proposed in [1], the signal to noise ratio’s gain is lower but contrarily to this approach there is no more range resolution deterioration.

**Fig. 3: Pulse-compressed data**

6. **CONCLUSION**
In this article, we have presented a new way to achieve pulse-compression in order to obtained pulse-compressed data less noise-corrupted than using the classical matched filter technique. The goal is to obtain after reconstruction (using for example a back-projection algorithm) a SAS image less speckle noise corrupted. The new pulse-compression method is based on the use of the stochastic matched filter in the echo signal’s time-frequency domain. This way, we bypass the stationary assumptions of the stochastic matched filter theory allowing obtaining a strong signal to noise ratio improvement while keeping the same range resolution than using the classical matched filter.

REFERENCES

CIRCULAR SYNTHETIC APERTURE SONAR WITHOUT A BEACON

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Abstract: Collection of synthetic aperture sonar (SAS) data along a circular track and forming a circular SAS (CSAS) image has several benefits over traditional stripmap SAS: the area of interest is observed from all aspect angles giving a better perception; the resolution in the image increases and shadow zones are avoided. Navigation requirements however, become even more stringent than for rectilinear SAS. Previously, CSAS images have been formed through use of an underwater beacon to alleviate the navigation problem. We present a scheme whereby CSAS may be achieved without use of a beacon.

In this paper, we calculate the required accuracy in navigation, bathymetry and sound velocity for successful circular SAS. We present a new processing chain for CSAS with modifications to micronavigation and autofocus. Modifications to micronavigation and autofocus have not previously been discussed in the literature. Finally, we show the results from the new processing chain with circular SAS images of small targets. These data have been collected over an arbitrary seafloor by the HUGIN autonomous underwater vehicle carrying the Kongsberg HISAS 1030.

As a result of these investigations we conclude that CSAS without beacon use is feasible for benign imaging geometries. Image interpretation however becomes more challenging. In addition we found that relative height was of considerable importance, with challenging topographic features causing degradation. To combat this we propose a hybrid autofocus / interferometry system to treat the height estimation problem in an iterative framework.

Keywords: synthetic aperture sonar, circular SAS, bathymetry, height-maps
1. INTRODUCTION

Circular synthetic aperture sonar (CSAS) can be useful for identification of underwater objects as scattering features are collected a wide range of view-angles. In addition, there is potential for a large improvement in image resolution. The combination of high-resolution imagery and the ability to measure variations in the scattering strength as a function of view-angle increases the information content. For example, many features on a target have strong angular dependence—smooth surfaces gives high backscatter in the specular direction and little echo in other directions. Information of this type offers the possibility of measuring surface orientation, roughness and possibly penetration into object surfaces.

CSAS however, is very difficult to implement in practice because SAS motion and environmental constraints increase dramatically (see section 2, [1, Chapter 7]). Despite this, the possible benefits to target-identification provide incentive enough to attempt CSAS. Examples of CSAS from the literature include [2] and [3]. Whilst impressive, the results have been obtained through use of navigation beacons placed inside or near the imaging scene [3] or stationary experimental setups [2]. Use of beacons in the scene of interest solves many problems but limits the usefulness of the method. There are no published results to the authors’ knowledge that implement successful CSAS without use of additional navigation beacons.

We present a SAS processing chain intended to provide CSAS imagery without use of a beacon. In section 2 we give background to the CSAS imaging problem and present navigation, sound-speed and topographical accuracy constraints. We present modified micronavigation and autofocus methods in sections 3.1, 3.2 and 3.3. Results and an analysis of the on-going work follow in sections 4, through 6.
2. CIRCULAR SAS CONSTRAINTS

Circular SAS has much tighter motion and environmental constraints than traditional synthetic aperture imagery. This is due to the violation of along-track cylinder-symmetry assumptions that are allowable in stripmap SAS [4]. Thus to prevent loss of phase-coherence, coherent imaging must be accurate to $\lambda/8$ over the entire circular aperture. For a 100 kHz SAS, for example, even $\lambda/4$ requires better than 3 mm position accuracy and 0.06 m/s sound-speed accuracy. Because of the non-straight aperture path, relative bathymetry is also required. In the previous example, for a target at 70 m range and 10 m below the sonar requires a height accuracy of about 5 cm.

To demonstrate the effect of similar errors in imaging geometry we have simulated a small scene containing a single point scatterer on the sea-floor at 28 m with a 10 m relative depth from a model HISAS-1030 SAS. Results are presented in fig. 2. The first image (2a) is of the scene without induced error. Image resolution is at pixel resolution and Airy pattern sidelobes taper rapidly to below -50 dB. In the second and third images (2b-2c) we have deliberately used inaccurate imaging parameters, in this case a height error of 10 cm and a sound-speed error of 1.5 m/s respectively. The result of using incorrect parameters is to limit real image resolution to the size of the 2-way slant-plane range error. In both cases the resolution is severely degraded.

Slowly-varying navigation errors will cause a geometrical degradation effect similar to that shown above. Incorrect imaging parameters cause significant resolution degradation in addition to that suffered though inaccurate navigation. However, as noted in Carrara [5, Chapter 5], rapidly varying navigation-errors will cause increasingly severe degradation.

3. METHOD

As shown in section 2, slant-range to the scene needs to be accurately known. Beacon measurement along-with accurate bathymetry is able to provide slant-range to sub-millimetre accuracy. Our goal is to duplicate this accuracy using the sonar data itself.
3.1. Micronavigation

Using the definitions in [6] we regard micronavigation as operating on redundant elements and autofocus as operating on scene redundancy. Most SAS micronavigation schemes assume a quasi-linear ping-to-ping movement with little angular change. For CSAS these assumptions are invalid and must be altered. We have implemented a variant of the displace ping imaging autofocus (DPIA) algorithm [7] whereby all ping-to-ping images are beamformed onto a common image grid fixed in global position. As the inertial navigation system (INS) provides accurate measures of ping-to-ping displacement this information is used in the beamforming steps to remove geometrical inaccuracies.

We first form an interferogram from images \( f(p, x) \) and \( f(p+1, x) \) from redundant elements in adjacent pings and noted that its phase \( \Phi(x, x_s) \) is related to position error via

\[
\Phi(x, x_s) = \angle \left[ f(p, x) f^*(p+1, x) \right] = 2k_0 \Delta r(x, x_s, \Delta x_s)
\]

with

\[
\Delta r(x, x_s, \Delta x_s) = \sqrt{(x - x_s) \cdot (x - x_s)} - \sqrt{(x - x_s - \Delta x_s) \cdot (x - x_s - \Delta x_s)}
\]

where \( x = [x, y, z] \) is the image coordinate, \( k_0 = 2\pi f_c/c \), where \( f_c \) is the center-of-mass frequency of the received echo and, \( c \), sound-speed and \( x_s = [x_s, y_s, z_s] \) is the average position of the overlapping elements for ping \( p \) and, \( \Delta x_s \), is the unknown displacement from ping \( p \) to ping \( p+1 \). We solve for the unknown \( \Delta x_s \) using an iterative search method where the difference between predicted and measured interferograms

\[
\left| \Phi(x, x_s, \Delta x_s) - \Phi(x)_{\text{measured}} \right|^2
\]

is minimized with respect to \( \Delta x_s \). For every pair of pings a new interferogram is created and the search run again. The pair-wise estimates of \( \Delta x_s \) are then integrated

\[
\hat{x}_s(p) = \sum_{p=0}^{\text{max}} \Delta x_s(p)
\]

to give a navigation error estimate \( \hat{x}_s(p) \) which is used to correct the INS-provided navigation solution.

3.2. Image correlation

Even after the micronavigation has been used, the navigation system suffers drift. To aid navigation and improve total image focus we have chosen to employ techniques based in autofocus. Due to the lack of common autofocus methods for circular geometries we chose to use non-coherent image correlation to generate a navigation solution. The method is related
to multi-aperture map-drift [5, Chapter 6.2] and operates by aligning images from sub-apertures.

The intensity image from each individual sub-aperture $i$, $f(i, x)$, is 2D cross-correlated with all other images $f(j, x)$ giving

$$Y(i, j, \tau) = f(i, x) \otimes f(j, x)$$

(5)

where the peak-location in $Y(i, j, \tau)$ as a function of 2D lag $\tau$ gives the image shift $\Delta \hat{x}(i, j)$ and $\otimes$ represented cross-correlation. In the current implementation, we weight individual correlations by a modified signal-to-noise ratio $w$ and multiply by the curvature of the cross-correlation function at its peak. Thus from each image-pair, a single estimate of movement in $x$ and $y$ is obtained, along with a weighting in each direction $w$.

To estimate the relative position of each image, we use a weighted least-squares solution for position. This is accomplished by forming difference equations in matrix form [5, page. 258], $A$ for each image pair and using a pseudo-inverse via

$$x_{est} = y(A^T w A)^{-1} w A^T$$

(6)

with $y = \Delta \hat{x}(i, j)$ and weights $w$.

Traditionally in map-drift the movement between images is assigned to a linear-phase error (which causes along-track image displacement). This is necessary for generating coherent CSAS imagery. Instead we use the $x_{est}$ estimates to shift sub-aperture images and combine them incoherently. This is a clear weakness in the method and we aim to provide navigation estimates in future work.

### 3.3. Height-map correlation

As the HISAS-1030 is an interferometric SAS, we also have the possibility of running sub-aperture height-map correlation. Use of height maps for the sub-aperture-image alignment provides a redundant source of information. Whilst operating in similar fashion to image correlation, height-map correlation has an important advantage: height-maps look the same from all directions and are obtainable in scenes containing only speckle (within speckle induced height-map noise). Normally these types of scenes cause de-correlation in map-drift and provide no useful information. In addition, height-map comparison allows straightforward measurement of vertical displacements, something very challenging in image-based correlation.

We generate the height-map correlation function $Y(i, j, \tau)$ via the ad-hoc

$$Y(i, j, \tau) = \frac{b(i, x) \otimes b(j, x)}{|b(i, x)| \otimes |b(j, x)|}$$

(7)

where $b(i, x)$ is the complex interferogram created from sub-aperture $i$. The rest of the method proceeds as for image correlation.
4. RESULTS

We have implemented a time-domain beamformer to generate imagery on the graphics processor unit (GPU). The GPU-based beamformer operates between 50-70 times faster than its C language equivalent on a Compaq 8510W laptop with an NVIDIA quadro FX 570M. This allows straight-forward (and fast) beamforming of acoustic data onto an arbitrary imaging grid.

For the results presented here, we used a data-set collected at 180 m depth in the area outside of Horten, Norway with the HISAS-1030 sonar. The data collection was obtained on January 17, 2008 with the Hugin 1000-MR [4] as the carrier platform. The scene of interest contains 2115 pings of 32-element multiple-receiver SAS data in a circle around an unidentified object. The data spans 540 degrees of circle with a radius of approximately 70 m. Due to the difficulty of keeping a stable long-term track for the 8 minute circle-duration the overlapping circle region is offset by 1-2 metres. A schematic of the scene is shown in fig. 1.

We assumed that the navigation is accurate enough on short time scales to obtain the HISAS-1030’s theoretical strip-map resolution of 2 cm by 2 cm. We then made 30 images, each containing 71 pings spaced uniformly over 540 degrees—18 degrees of angular coverage for each image. Each sub-aperture image has a theoretical resolution of approximately 2 cm by 2 cm and at 1024 by 1024 pixels covers just over 20 m by 20 m. All images are processed onto a flat ground-plane at the average sea-floor depth 184.5 m.

![Fig. 3: Incoherent CSAS images, log intensity with 35 dB dynamic range from peak: (a) original INS navigation, (b) after micronavigation, (c) after micronavigation and image correlation, (d) after micronavigation and height-map correlation.](image-url)
All 30 images were then added incoherently to give a “semi-coherent” image for comparison imagery. The object of interest is not covered in the entire circular aperture. Lacking a full circular aperture coverage, we presume only incoherent imaging will work without causing excessive side-lobes.

We made a reference image using the navigation solution provided by the INS system on-board the collection AUV. This image is shown in fig. 3a. As opposed to the images shown in fig. 2 we see replicas of the object. This is due to the uneven scene visibility noted above. Five to ten of the images cover only small parts of the scene thus instead of getting donut-like point-spread-functions one gets scene replicas. These replicas are widely displaced in position indicating large navigation-errors.

Using the micronavigation technique outlined in section 3.1 we generated a ping-by-ping navigation error estimate (see fig. 4a) and used this to generate a new image shown in fig. 3b.

We then used the image-correlation correlation routines of section 3.2 on the image obtained in fig. 3b the image shown in fig 3c is formed. The equivalent image when using height-map correlation is shown in fig. 3d. Image position error estimates from the methods are shown in figures 4b and 4c as a function of image-number.

5. ANALYSIS

Increasingly accurate incoherent images over those from INS navigation are available from micronavigation and image-correlation respectively. Height-map correlation also improves the final result, although it seems to not have the accuracy of image-correlation for this data-set. This trait is perhaps to be expected. Scattering strength in particular is strongly dependent on local slope of the scene. Imagery thus has a large signal at points of changing bathymetry and will as a consequence have better cross-correlation properties. That said, a better method for height-map correlation may provide a better navigation-error estimate.

We estimate the total resolution of the best result (in fig. 3c) to be on the order of 10 cm. Given that all of the imagery is generated on a flat seafloor at 184.5 m and the object is estimated from height-maps to be >0.4 m high, 10 cm accuracy corresponds to that expected from the imaging height inaccuracy predicted in section 2. As we have constrained the navigation to only cases where bathymetry is well known, our current technique can not get better accuracy.

To solve this problem, one needs to form the images onto an accurate digital terrain map (DTM). Typically, DTMs are not available in 2 cm by 2 cm resolution with 5 cm height-
accuracy. In order to generate such a DTM, one needs to use SAS-generated height-maps. In addition, each image needs to be registered accurately to the height-map for this to work, also a non-trivial undertaking. Overcoming these limitations will probably require a combined height/navigation/sound-speed estimation procedure.

6. SUMMARY

Using micronavigation and autofocus we have produced a useful incoherent CSAS image without a beacon over 540 degrees of motion and for 20 degrees coherent apertures. The worst-case resolution of the image is approximately 10 cm. Obtained resolution is not constant throughout the image and is highly dependent on scene differences from the assumed height-map.

We have shown that it is possible to use height-map correlation as the basis for an independent navigation data-source although on the dataset investigated; the accuracy of image displacement measures was poorer than with those generated from image amplitude.

Future methods should concentrate on improving micronavigation accuracy to provide a better starting point. We recommend combining sound-speed, relative-height and relative-position estimation techniques to solve the tight coupling between errors in each domain in CSAS. A method of linking sub-aperture image displacements and navigation errors is also strongly desirable in order to compare image correlation results with micronavigation and allow image correlation to be used for improving overall vehicle navigation.

REFERENCES

EXPERIENCES FROM THE SWEDISH ARMED FORCES USING A SAS-AUV

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Abstract: The AUV 62-MR is a system for Mine Reconnaissance, based upon the AUV 62 family of 21”-diameter vehicles, which is today in operation with the Swedish Armed Forces. Each MR-vehicle is equipped with a pair of Flank Array Sonars, suitable for high-resolution bottom imaging using Synthetic Aperture Processing. Including dual receiver arrays, each of the sonars is also able to produce interferometric data, which enables simultaneous bathymetry. This has shown to be very valuable in areas with large water depth variations, in order to project the image correctly at the true height.

After extensive evaluation together with the Swedish Defence Materiel Administration, a large number of different operations have been performed by the Mine Clearance Units as well as the Amphibious and Submarine Units of the Royal Swedish Navy. During these operations, a lot of high-resolution data have been delivered to the Mine Warfare Data Center (MWDC) for further evaluation. In parallel, the basic capabilities of the AUV 62-MR have been demonstrated during an intensive exercise on the Australian East-Coast in May 2007.

During 2008, a lot of valuable experiences have been achieved – including finding new underwater objects in harsh areas, where conventional ship-based missions have failed. The high-resolution sonar of AUV 62-MR has also been found very effective in surveying dump sites, where objects are spread over large areas. The low operating frequency, as compared to conventional Side Scan Sonars, also enables semi-buried objects to be easily detected in soft bottom areas. This has also been proven effective during searches for drowned persons in soft bottom lakes, commissioned by the Swedish Police Service and the Swedish Coast Guard, where divers experience great problems with bad visual conditions.

Keywords: AUV, Side Scan Sonar, Synthetic Aperture Processing
1. INTRODUCTION

AUV 62 is a family of modular 21” vehicles, which can be configured for a number of missions, where Mine Reconnaissance is one of the most demanding. The vehicle in the AUV 62 MR-configuration is shown in Fig. 1.

![Fig.1: AUV 62 in the Mine Reconnaissance configuration.](image)

The main sensor is a dual Flank Array Sonar (one at each side), which together with on-board Synthetic Aperture Sonar (SAS) Processing allows high-resolution images of the seabed. Using SAS enables a lower frequency of operation than for a conventional Side Scan Sonar (SSS) to achieve the same resolution. The lower frequency gives the ability to penetrate soft bottom sediments slightly. This will, in some conditions, enable detection of buried objects, which would not be possible to detect using a high frequency SSS.

2. THE SYSTEM

2.1. The AUV 62 System

The complete AUV 62 System is modular. This is established by an open architecture, as shown in Fig. 2. The major parts are the AUV 62 Vehicle and the corresponding Support System, which includes the systems ashore and onboard the launching platform.

![Fig.2: The AUV 62 System Architecture.](image)
2.2. The Sonar System

To enable MR-operation, the Payload Module shown in Fig. 2 is adapted as follows:

- The Sensor Analysis System is equipped with enough processing power to enable real-time SAS-processing.
- The Sonar Arrays are chosen to enable enough Spatial Resolution and Area Coverage Rate for the purpose.
- The Payload Control System is dedicated for sonar operation and massive data storage, since raw-data is always stored together with the on-line produced SAS-images.

In addition to that, the shipboard AUV Operators Console (AOC in Fig. 2) is furnished with additional software, to enable efficient presentation of high-resolution images, and to perform any necessary post-recovery analysis of the sonar data. Table 1 shows the major system characteristics for a typical MR-configuration.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Spatial Resolution</td>
<td>≤ 5x5 cm</td>
<td>Governed by the Sonar Element Size and Signal Bandwidth used</td>
</tr>
<tr>
<td>Area Coverage Rate</td>
<td>4-5 km²/h</td>
<td>Given by the Sonar Array Length</td>
</tr>
<tr>
<td>Operating Frequency</td>
<td>100 kHz</td>
<td>Chosen by design</td>
</tr>
</tbody>
</table>

*Table 1: Major characteristics for a typical MR-configuration.*

Fig. 3 shows parts of the results from a search operation, resulting in a high-resolution image of a “Rockan” mine. The distance from the sonar to the mine in the right-hand image is approximately 75 m, with the mine lying proud upon a smooth seabed.

*Fig. 3: Left – detected objects high-lighted. Right – SAS-image of a “Rockan” mine.*

The left-hand image of Fig. 3 shows the track-record of the vehicle, with the detected objects high-lighted. This detection is also performed in real-time, onboard the vehicle, and can thus be presented to the operator immediately after surfacing the vehicle.

The operator can then make a more thorough, manual, classification of the objects that the onboard CAD/CAC systems have indicated.
3. FIELD OPERATIONS

3.1. Short summary

The AUV 62 MR has been extensively used together with the Swedish Armed Forces for a number of different operations – with Mine-hunting as common denominator. Because of the size and shape of the vehicle, it can easily be launched and recovered using the crane of smaller ships, such as the 30 m long HMS Fårösund (Fig. 4).

![Fig.4: HMS Fårösund – minelayer with crane for launch and recovery.](image)

During 2008, more than 50 different missions have been performed only using the Sapphires-vehicle [1] shown in Fig. 4. Among these, there have also been some specific operations, worth to mention in somewhat more detail [2]:

- Finding new, undiscovered sea-mines in a Swedish WW2 mine-field.
- Discovering old ammunition in an early dump-site.
- Finding a drowned person, lying on a muddy seabed.

3.2. Interesting examples

A Swedish mine-field from 1942 has been cleared with conventional mine-hunting sonars at several occasions, and was considered to be fully covered. Since the seabed is a mixture of sand, rocks and stones of different sizes, there could still be some mines not found. The area was covered by the AUV in two separate turns, with manual image analysis in between. Fig. 5 shows an example of a mine found in the area, together with a picture of the particular type of mine.

After sending down a small ROV for identification, it was confirmed that the mine was intact and that it had obviously been missed during the earlier mine-sweep operations.
During another mission in a Swedish lake, the size of a dump-site for WW2-ammunition was to be established. During a total of 8 separate runs, with a total time of 20 hours, about 12 km² was covered with 4-5 cm resolution. The result was about 700 piles of ammunition-cases, grenades, mines, and likewise, more or less intact, of which about 60 was indentified using ROV. Some of these can be seen in Fig. 6, together with an ROV-image of a case.

The last example is taken from an operation in cooperation with the Swedish Coast Guard. The reason was an accident in a Swedish lake, where four persons fell into the cold autumn water from a small boat. Two of them managed to swim ashore, while the other two drowned. One of the two victims was found after two weeks of search using conventional sonar, while the other was still missing.

The early search area of 500x500 m was first extended to 1050x700 m, to assure enough coverage. The visibility in the area was around one decimetre, which made searching with divers totally out of question. The result is shown in Fig. 7.
The image in Fig. 7 is acquired from a distance of about 70 m. A conventional SSS would have to be passing by at a distance of less than 10 m, in order to achieve resolution enough for proper classification. The ability to zoom in the SAS-image without loss of resolution (using the AOC-software) made classification possible without any doubt only minutes after the recovery of the vehicle.

4. SEAMLESS ZOOMING

The ability to zoom in the high-resolution image without loss of resolution is another important feature of the evaluation software of the AUV 62 MR System. The images shown in Fig. 8 can serve as an example. The upper image shows a sketch of the wheel-steamer Erik Nordevall that sunk in lake Vättern, near Motala, some 150 years ago. Since the water in this lake is extremely clear and clean, the wooden ship is almost intact – possible for divers to examine in detail. The lower image is part of a high-resolution overview SAS-image of the ship, which is captured on the screen and enlarged in the conventional image-editor.

In Fig. 9, the captured images are enlarged further, in order to discover some interesting details within the areas which are marked with red frames in Fig. 8. The objects lying on the ship’s deck are visible as blurry highlights, but not so much more.

In Fig. 10, finally, the zooming was instead made using the software tools in the AOC. Now is not only the anchor that is clearly visible, but also strip of wood lying on the deck. This means that every interesting object found in the overview image, which is presented immediately after recovery, can be seamlessly zoomed-in for further inspection. This is very time-saving, especially in situations where the number of detected objects to explore is large.
Fig 8: Medium-resolution image of wheel-steamer Erik Nordevall.

Fig 9: Enlarged, medium-resolution image of details.
5. CONCLUSIONS

The AUV 62 MR, equipped with Flank Array Sonars and real-time SAS-processing is an efficient tool for high-resolution seabed imaging. Even if the major purpose is Mine Reconnaissance, which has been proven extensively in cooperation with the Swedish Armed Forces, there has also been shown that there are other useful missions possible. These include the detailed examination of dump-sites, as well as the search for drowned people.

REFERENCES

FILTERING OF HIGH RESOLUTION INTERFEROMETRIC SYNTHETIC APERTURE SONAR DATA

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Abstract: Interferometric processing of synthetic aperture sonar (SAS) data can produce bathymetric estimates at very high resolution. Due to the speckle in SAS-images, the interferometric phase will have a large variability. In order to increase the accuracy of the phase estimate a window of pixels can be applied, leading to a compromise between horizontal resolution and vertical accuracy. However, even a maximum likelihood estimator will significantly degrade the resolution of the depth estimates. In this paper we consider three different smoothing filters, which we apply directly on the bathymetric maps. A smaller window of pixels can then be used to estimate the interferometric phase difference. Both a weighted bilateral filter and a weighted median filter are suited for this purpose. We use the signal-to-noise ratio, also estimated from the interferometer, as weights. We present results from real data collected with a HISAS-1030 mounted on a HUGIN autonomous underwater vehicle. We have tested several different types of bathymetries and assessed the performance of our methods for each case. In all cases, applying a weighted filter on the bathymetric map can enable a bathymetry with less noise while preserving the edges of objects.

Keywords: Synthetic aperture sonar, interferometry, bathymetry, height estimation, weighted bilateral filter, weighted median filter, smoothing filter, coherence
1. INTRODUCTION

Interferometric synthetic aperture methods benefit from the coherent properties of speckle. However, low intensity pixels inherent in speckle suffer from low signal-to-noise ratio (SNR), rendering full-resolution phase-differencing unpractical. A common solution is to use a maximum likelihood phase (MLP) estimator on a few neighbouring pixels [1], compromising between phase accuracy and horizontal resolution [2].

Within the MLP-filter, the phase-differences are assumed to be homogenous, which sets an upper limit to the size of the filter. Exceeding this size causes a drop in coherence, which again decreases the accuracy of the phase estimates. This means that the MLP-filter is unable to decrease the variance of the phase-difference estimates to a desired level.

In this paper we describe and demonstrate how the interferometric depth estimates can be filtered in order to decrease the variances. Instead of applying a filter on the complex interferometric images we filter the depth estimates in post-processing. This removes the need for homogeneous phase-differences within the filter window.

We have compared a weighted smoothing (WS) filter [3], a weighted median (WM) filter [3] and a weighted version of the bilateral (WB) filter [4] and found that the WB-filter performs best. The WS-filter degrades the sharp transitions to an unacceptable degree while the WM-filter fails to smooth flat areas sufficiently. These findings are presented on interferometric data from the HISAS-1030 synthetic aperture sonar [5], [6].

2. SYNTHETIC APERTURE SONAR INTERFEROMETRY

A synthetic aperture sonar (SAS) image is generated by coherently combining pings along the direction of travel. The result is an image with range and frequency independent resolution [7]. In interferometric SAS, two images are formed from different vertical positions, and the seafloor depth determined from the range differences [8] (see Fig. 1).

![Fig.1: Basic vertical geometry in interferometric sonar.](image)

The relative depth, \( z \), is related to the phase-differences, \( \phi \), by the following relation

\[
z \approx \frac{\phi \lambda}{2\pi \delta z} - \frac{\delta z}{2},
\]

where \( \lambda \) is the wavelength of the transmitted signal, \( \delta z \) the baseline between the two receivers and \( r_2 \) the range from receiver #2 to the seafloor.
3. FILTERING TECHNIQUES

A very useful quantity of the interferometric processing is the coherence [1] of the estimated bathymetry. The coherence, $w_1$, is defined as the magnitude of the complex interferogram

$$w_1 = \left| \frac{\sum_i m_i s_i^*}{\sqrt{\sum_i |m_i|^2 \sum_i |s_i|^2}} \right|,$$  

(2)

where $m_i$ is the $i$'th pixel of the master image and $s_i$ is the corresponding pixel in the slave image. The coherence can be converted to an equivalent signal-to-noise ratio from the following relation [1]

$$w_2 = \frac{w_1}{1 - w_1}.$$

(3)

Both of these quantities can be employed as input weights to the filters we present. We have found that $w_2$ is best suited since it better separates weights at high SNRs. In addition, we remove low SNR estimates completely by thresholding the correlation coefficient at 2/3. We therefore define a set of filtering weights, $w$, as

$$w = w_2, \quad w_1 \geq 2/3$$

$$w = 0, \quad w_1 < 2/3.$$  

(4)

These are the weights we use in the results presented in this paper. In addition we have tested the square root of SNR and the logarithm of SNR, but they both performed poorer than SNR itself.

3.1. Maximum likelihood phase (MLP) filter

As described in Section 1, we can reduce the variances in the phase-difference estimates by increasing the size of the MLP filter. The phase-differences are estimated by

$$\phi = \angle \left\{ \frac{\sum_i m_i s_i^*}{\sqrt{\sum_i |m_i|^2 \sum_i |s_i|^2}} \right\},$$

(5)

and then the relative depths are found using Eq. 1. This filter is always used in our interferometric processing, independent of any subsequent filtering. The advantage with this filter is that it is a maximum likelihood filter, but as described in Section 1 it assumes homogeneity within the filter size.
3.2. **Weighted smoothing (WS) filter**

The WS filter is a weighted boxcar average filter applied on the relative depth estimates, \( z \)
\[
\hat{z}_i = \frac{\sum_i w_i z_i}{\sum_i w_i},
\]  
(6)

where \( w \) are the weights defined in Eq. 4. The WS-filter performs well on slowly varying seafloors but it degrades edges and small objects.

3.3. **Weighted median (WM) filter**

Contrary to traditional WM-filters, our version is implemented with adaptive, non-integer weights. First the window depth values are sorted in ascending order. Using this sample order, the smallest index is found whose cumulative weight is equal or greater than half the total weight sum. The corresponding depth is output as the WM value.
\[
\hat{z}_2 = z_k, \quad \text{where} \quad \min_k \left\{ \sum_{i=1}^k w_i \geq \frac{1}{2} \sum_{i=1}^n w_i \right\},
\]  
(7)

where \( n \) is the number of window pixels, \( z \) are depths and \( w \) are weights sorted in ascending depth. The advantage of the WM-filter is that it removes wild points while preserving edges. The disadvantage is that it has an unpredictable statistical behaviour and it does not smooth slowly varying regions sufficiently.

3.4. **Weighted bilateral (WB) filter**

The WB-filter is the most complex filter. It consists of a Gaussian smoothing in both horizontal and vertical direction. Thus each sample is weighted according to three criteria: The horizontal distance from the centre sample, the difference in value (vertical distance) from the centre sample, and the weight defined in Eq. 4
\[
\hat{z}_3 = \frac{\sum_i G(d_i) G(v_i) w_i z_i}{\sum_i G(d_i) G(v_i) w_i},
\]  
(8)

where \( d \) are the horizontal distance, \( v \) the difference in value and \( G \) the Gauss-function:
\[
G(x) = \frac{1}{2\pi\sigma^2} \exp\left( -\frac{x^2}{2\sigma^2} \right). \]

The WB-filter smooths slowly varying regions while preserving edges and therefore provides high resolution on sharp objects and less on a flat seafloor. These properties are both desirable for visualisation of 3D surfaces.
4. EXPERIMENTAL RESULTS

Fig. 2: HUGIN 1000-MR AUV launch from a Royal Norwegian Navy minehunter. One of the receive arrays of the HISAS-1030 system can be seen on the midsection of the AUV.

We have tested the different filters on a selection of experimental data collected with the HISAS-1030 mounted on the HUGIN 1000-MR (see Fig. 2). The original bathymetry often shows an unnatural large variability on flat regions of the seafloor, caused by a marginal SNR in the interferometric processing. However, on rocks or man-made objects, the reflectivity and thus the SNR are usually higher. Therefore the accuracy of the estimated shape of objects can be much higher.

The ideal bathymetric filter should smooth slowly varying parts of the seafloor and preserve edges, while using the coherence (or SNR) as weights such that the bathymetric error is smallest possible. One approach to obtain such properties is to change the size of the MLP filter adaptively. This has been studied in [2] where we found that this method works well up to a size where the MLP filter starts to decorrelate.

Fig. 3 shows an example of the estimated bathymetry on a small part of a 30 metres long wreck. Notice two different sections: The flat seafloor at the left and the small “bridge” at the upper right. In the original bathymetry the bridge is quite sharp, but the seafloor is contaminated by noise. After applying the SM-filter, the flat seafloor is smoother, but the bridge is rounded at the edges. The WM-filter does a better job of preserving the edges of the bridge, but it does not smooth the flat seafloor sufficiently. By applying the WB-filter, we clearly see that the seafloor is smoothed while the edges of the bridge are equally distinct as in the original bathymetry.

In Fig. 4 we show a patch of seafloor with small rocks. The WS-filter (upper panel) actually removes some of the small objects. These objects are quite easy to detect in the bathymetry after WB-filtering. WM-filtering on this scene does not contribute to a smoother surface and is omitted from the figure. On all other scenes we have tested we have found similar results: The WB-filter gives the best trade-off between smoothness and edge preservation.
Fig.3: Upper left: SAS image of a 30 m long wreck at 340 metres water depth. Upper right: Corresponding coherence. The four lower panels show a cut-out of the estimated...
bathymetry. Upper left: No filter; Upper right: Smoothing filter; Lower left: Median filter; Lower right: Bilateral filter. Courtesy of the Royal Norwegian Navy.

Fig. 4: Estimated bathymetry of a 20 by 20 metres rocky seafloor at approximately 30 meters water depth. Upper panel: Unfiltered bathymetry. Centre panel: Smoothed bathymetry. Lower panel: WB filtered bathymetry. Note that the original bathymetry has an unnatural large variability caused by marginal SNR. A WS filter reduces this effect, but also smoothes real objects like the two rocks in the lower part of the scene. The bilateral filter gives a favourable trade-off between a smooth seafloor and distinct objects.
5. CONCLUSIONS

Synthetic aperture sonar interferometry has the potential to map the seafloor with extreme resolution. Due to the speckle in the SAS-images, interferometric SAS estimates will have large variability. Increasing the size of the maximum likelihood filter used in the interferometric phase difference estimates will reduce the variability at the cost of reduced horizontal resolution. This assumes, however, that the interferometric data are homogeneous inside the filter size. We have presented three different filters which can be applied in post-processing directly on the bathymetry. The filters do not require homogeneity within the filter window. We have applied the different filters on experimental data collected by HUGIN 1000-MR carrying a HISAS 1030 interferometric SAS on different scenarios. We found that the weighted bilateral filter achieves the best performance in smoothing slowly varying areas while preserving edges of distinct objects.

6. ACKNOWLEDGEMENTS

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REFERENCES


SYNTHETIC APERTURE SONAR IN CHALLENGING ENVIRONMENTS: RESULTS FROM THE HISAS 1030

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Abstract: Successful synthetic aperture sonar (SAS) imaging is dependent of several challenges to be overcome. The sonar has to be positioned with accuracy better than a fraction of a wavelength along the entire synthetic aperture. At 100 kHz this equals an accuracy requirement around 1 millimetre along tens of metres of travelled distance. The ocean environment, and particularly the sound velocity, has to be accurately estimated for successful focusing of SAS images. This is due to the fact that SAS uses near field imaging. For non-straight synthetic apertures, the bathymetry of the scene to be imaged has to be known. This less known fact is critical for robust autonomous underwater vehicle (AUV) based SAS in areas with rough terrain. The first HUGIN 1000-MR AUV was delivered to the Royal Norwegian Navy in mid-2008. The main sensor on the vehicle is the HISAS 1030 interferometric SAS. In this paper we present the system and the signal processing scheme we apply in SAS processing of data collected from challenging environments. We derive a simple rule-of-thumb for accuracy requirement in non-straight vehicle tracks and inaccurate seafloor maps. We show the effect of inaccurate mapping in combination with non-straight vehicle tracks in SAS imagery on real data collected by HUGIN 1000-MR from an area with severely rough topography.

Keywords: synthetic aperture sonar, sonar imaging, bathymetry
1. INTRODUCTION

Fig. 1: Left: The HUGIN 1000-MR onboard the RNoN MCMV Hinnøy during a mission in August 2008. Right: closeup of the HISAS 1030 interferometric SAS.

The principle of synthetic aperture sonar (SAS) is coherent combination of successive pings such that a large sonar antenna along-track is formed. By applying near-field beamforming to the synthetic antenna with sufficient motion estimation and correction, SAS images of the seafloor can be produced. The main improvement over traditional sidescan sonar (SSS) is the resolution gain along-track (the SAS to SSS gain), which can be a factor 10 - 100. The principle of SAS is not new [1], but it is only in the recent years several commercial systems have become available.

Synthetic aperture radar and sonar images possess the rather unique feature that along-track resolution is fixed, independent of range and frequency, only determined by the transducer element size [1][2]. This opens for new sonar design possibilities, not available in traditional SSS, where along-track resolution is inevitably linked to sonar frequency. A high resolution SAS can have a much lower center frequency than a high resolution SSS.

The first HUGIN 1000-MR autonomous underwater vehicle (AUV), as shown in Fig. 1, was delivered to the Royal Norwegian Navy (RNoN) in mid-2008. The primary payload sensor is the HISAS 1030 interferometric SAS [3]. The sonar has two full length receiver arrays, and a vertical phased array transmitter (that also can receive). The transducers are wideband, with a usable frequency range between 50 – 120 kHz. The standard frequency selection is, however, 30 kHz bandwidth around 100 kHz center frequency. The theoretical resolution is better than 5x5 cm, and the area coverage rate exceeds 2 km² per hour. The interferometric capability gives full-swath, co-registered, high resolution bathymetry. HISAS 1030 is an extremely capable sensor designed for various military applications [4], and civilian applications such as underwater archaeology, detailed mapping of the seafloor, search, documentation of offshore oil and gas installations, documentation of dump sites and mapping of coral reefs.
2. ROBUST SYNTHETIC APERTURE SONAR

The success of synthetic aperture sonar (SAS) technology is highly dependent of the ability to produce well focused SAS images in all types of environments. Fig. 2 shows our basic processing scheme for robust SAS processing with adaptation to the environment. In this section we list the fundamental challenges in SAS and how we approach them.

Each sonar-element has to be positioned with accuracy better than a fraction of a wavelength along the entire synthetic aperture. For HISAS 1030 (100 kHz center frequency), this equals an accuracy requirement around 1 millimetre along tens of metres of travelled distance. The HUGIN AUV is equipped with advanced aided inertial navigation [5]. However, the requirement for relative position in SAS is generally not met. To achieve this goal we use sonar micronavigation [6] in optimal combination with aided inertial navigation [7].

The ocean environment, and particularly the sound velocity, has to be accurately estimated for successful focusing of SAS images [8][9]. This is due to the fact that SAS is near-field imaging. The mapping between time and space conducted by beamforming requires accurate knowledge about the integrated sound velocity along the acoustic ray paths. We approach this problem in several ways. We estimate the sound velocity profile from the onboard vehicle Conductivity, Temperature, Depth (CTD) sensor. If there still is a residual error in the sound velocity causing defocusing in the SAS imagery, we can apply a blind technique that both corrects the image and estimates the error in the average sound velocity [9]. This technique is strongly related to autofocusing [10].

For non-straight synthetic apertures, the imaging plane and thereby the bathymetry of the scene to be imaged has to be known [10]. This is very important for robust AUV-based SAS in areas with rough terrain – such as much of the Norwegian littorals. In order to address this problem, the HISAS 1030 is designed as an interferometric system [11][4], where a coarse bathymetry from the sidescan interferometer is calculated. The accuracy requirement in bathymetry for non-straight vehicle paths is the topic of the next sections.

Sonar micronavigation aims at correcting for any error in the measurement geometry, either due to incorrect navigation or incorrect projection plane. This means that, even though an out-of-plane motion caused projection error exists, micronavigation can, although not optimally, partially correct for this.
3. SENSITIVITY TO TOPOGRAPHIC ERRORS

Assume a vehicle track that deviates from a straight line normal to the imaging plane, as illustrated in Fig 3. When proper out-of-plane motion correction is performed, all objects in the imaging plane will be well focused. For an object outside the imaging plane, the distance becomes incorrect and thereby defocused, even for error free navigation [10]. This defocusing is proportional to the out-of-plane motion deviation. The difference in distance $\delta R$ due to out-of-plane motion $\Delta z$ to an elevated target is (see Fig. 3)

$$\delta R = 2(R_0 - R_1) \approx \Delta z \sin \phi \approx \Delta z \Delta h / R_0$$

(1)

The difference in distance, or error in imaging geometry, leads to a quadratic phase error in the imaging and hence defocusing. We have found that a phase error less than $\pm \pi / 2$ gives acceptable focused imagery. This equals $\delta R \leq \lambda / 8$ (taking into account two-way travel) which gives the following requirement

$$\Delta z \Delta h \leq \lambda R_0 / 8$$

(2)

Note that this result differs from the similar requirement in [10], which we find too strong. Assuming a sound velocity of 1500 m/s, we get the simple rule-of-thumb

$$\Delta z \Delta h \leq R_0 / (5f), \quad f \text{ [kHz]}$$

(3)

As an example, consider one meter deviation $\Delta z = 1 \text{m}$. At 100 m range, the required seafloor height accuracy then becomes $\Delta h \leq 0.2 \text{m}$ at 100 kHz. This is a surprisingly strong requirement, but for normal conditions for the sidescan bathymetry on HISAS 1030, this is actually met.

The required height accuracy relaxes with increasing range. The provided accuracy in sidescan bathymetry does, however, decrease with increasing range [12]. The length of the synthetic aperture, and thereby the average deviation from straight-line, also increases with increasing range. This means that the requirement (2) is in general most difficult to obey at maximum range. This is in agreement with our observations. In rough topography, there is often a maximum range for which the images are well focused. Note, however, that errors due to incorrect navigation and sound velocity also increase with range.
4. ANALYSIS OF REAL DATA COLLECTED BY HUGIN AUV

Fig. 4: Vehicle depth and seafloor depth for the HUGIN mission line 1502 from June 17, 2008. The red dashed vertical lines indicate the collection window for the SAS image shown in Fig. 8. The purple dash-dotted vertical lines indicate the collection window for the SAS image shown in Fig. 6.

In this section, we analyse a particular set of SAS data that was collected with HISAS 1030 on the HUGIN 1000-MR, June 17, 2008, outside Horten, Norway. Fig. 4 shows the vehicle depth (green solid line) and the seafloor depth (dashed blue line). The track is non-straight with large depth variations. Fig. 5 shows the seafloor bathymetry estimated with sidescan interferometry. The solid line is the vehicle track. We see large variations in the bathymetry. The seafloor depth varies from 30 m to 85 m during this particular mission line. This leads either to a non-straight AUV-path in bottom following mode (as this mission actually was), or a severely sub-optimum altitude during parts of the mission line in constant depth mode. Neither is optimal for SAS imagery.

Fig. 5: Sidescan bathymetry with the vehicle track shown as solid white line. The map is color coded in depth in meters. The depth has been exaggerated by a factor 2.
Fig. 6: SAS image from a HUGIN 1000-MR mission in June 2008. The image size is 140 m along-track (x-axis) times 100 m cross-track (y-axis). The theoretical resolution in the image is 4.5 cm times 3.2 cm. The image shows an area with rough topography.

Fig. 6 shows a SAS image from the data collected in the time interval 1260 s to 1350 s (see the purple dash-dotted lines in Fig. 4). The depth variation during this collection was approximately 20 m. The image scene contains various topography and small objects. Fig. 7 shows four zoomed snippets with corresponding coherence maps based on the SAS coherence [13][11]. In the areas of the images that are well focused, the coherence is in general high. In the areas of low coherence, the image might be defocused. This is caused by the non-straight vehicle path in combination with reduced quality in the seafloor maps.

Fig. 7: Upper row: 12 x 12 m areas around selected targets in Fig. 6. The images are shown with 50 dB dynamic range. Lower row: corresponding interferometric SAS coherence. Red equals high coherence, and blue equals low coherence.
Fig. 8: SAS image from a HUGIN 1000-MR mission in June 2008. The image size is 90 m along-track (x-axis) times 60 m cross-track. The theoretical resolution in the image is 4.5 cm times 3.2 cm. The image shows the wreck of the Norwegian tanker Holmengraa.

Fig. 8 shows a SAS image from the data collected in the time interval 1070 s to 1130 s (see the red dashed lines in Fig. 4). The image shows the wreck of the Norwegian tanker Holmengraa that was sunk during World War II in 1944. The AUV track has severe out-of-plane deviations. There is local defocusing, particularly visible in the bow region of the wreck. In the areas on the seafloor outside the wreck, small objects are well focused. This indicates that the integrated navigation solution is sufficiently accurate, and the defocusing in the image is caused by the out-of-plane deviations. Note that the pollution in the image at range larger than the maximum range for the wreck (155 m) cannot be caused by defocusing. This is more likely multiple reflections close to the bow-region of the wreck.

As discussed in the previous section, there are generally two solutions to the problem of SAS imaging in areas with rough bathymetry: 1) force the vehicle track to be a straight line; 2) map the seafloor with sufficient accuracy. Running on a straight line is not trivial in this case. The mapping accuracy of the seafloor is therefore critical. In the default processing of SAS data, we use sidescan bathymetry as the height estimate to the SAS imaging (from the same mission line). Large elevated targets (such as a large wreck) can potentially cause layover in the interferometric processing and thereby loss of coherence and inaccurate mapping [13]. This can be overcome by using the multibeam echosounder (MBE) on the HUGIN 1000-MR in combination with the sidescan interferometer to provide more accurate maps. This requires, however, more mission lines, as the swath width of the MBE is much more narrow than that of the HISAS sidescan bathymetry.

5. CONCLUSIONS

The success of synthetic aperture sonar (SAS) technology is highly dependent of the ability to produce well focused SAS images in all types of environments. There are three fundamental challenges in SAS: 1) The sonar has to be positioned within a fraction of a
wavelength along the synthetic aperture. 2) The ocean environment, and in particular the sound velocity, has to be accurately estimated in all depths for the acoustic wave travel. 3) For non-straight vehicle paths along the synthetic aperture, the full imaging geometry, including the seafloor bathymetry has to be known within certain bounds. To approach these challenges, we have developed a scheme for robust adaptive processing of the SAS data. The HISAS 1030 system is also specifically designed to be robust, applying key technologies such as interferometry, large bandwidth, and a large number of sonar channels.

REFERENCES

Structured Session 11

Threat Detection, Tracking and Deterrence for Port Protection

Organizers: Georgios Haralabus & Ronald Kessel
A low-frequency acoustic swimmer deterrence concept for port protection

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Abstract: The protection of high-value military and commercial assets while in port presents many challenges. From the water-side, the threat posed by a covert swimmer is of particular interest. At the system level, integrated defense systems are being developed to address the threat from detection through neutralization. In this work, the concept of an emplaced low-frequency acoustic system for swimmer deterrence is presented. The concept is based on eliciting a response from a submerged swimmer to low-frequency acoustic energy. Depending on the defensive posture at the time of engagement, a range of responses is sought which include hail-reply, evacuation of area, or complete incapacitation. The range of anticipated responses is based on what is known about physiological responses of the human body exposed to low-frequency energy having varying combinations of frequency and intensity. Implementation requires source configurations and control capabilities for delivering acoustic energy commensurate with the extant defense requirements. During the summer of 2008, a prototype system was demonstrated for adapting the low-frequency acoustic field emitted by an array of sources to track the presumed track of a covert swimmer. By controlling amplitude and phase of the individual sources, sound pressure levels were recorded at the presumed swimmer locations in excess of those thought to elicit a response. In this talk the design and performance characteristics of the prototype system will be discussed.
SYSTEM PERFORMANCE TRADE-OFF IN UNDERWATER HARBOUR PROTECTION

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Abstract:
This paper reports a simulative parametric study to evaluate in a quantitative manner several performance trade-off in underwater harbour protection systems. The basic system components considered are: a fixed active acoustic sensor for threat detection and tracking, co-located with the asset to be defended; a patrol boat, activated and guided toward the threat by the fixed sensor, for threat re-acquisition, classification and deterrence; an underwater threat. System performance is evaluated through Monte-Carlo simulation, by exploiting the possibilities of a harbour protection system simulator recently developed by the Authors.

Keywords: Underwater surveillance systems, Harbour protection, Monte-Carlo simulations
1. INTRODUCTION

The design of an underwater anti-intrusion system meeting some specified requirements is not an easy task. One of the difficulties lies in the fact that the metric used to define the anti-intrusion performance is not directly linked to the detection/classification instrument specifications. In fact, as it is well known [1], performance of underwater acoustic systems does not depend solely on the system characteristics (frequency, bandwidth, source level, beam pattern) but also on the environmental conditions (sound speed, bathymetry, ambient noise, etc.). So same system may have better or worse performance depending on the specific environmental situation in which it is operating. Whenever a complex underwater anti-intrusion system is deployed, it is necessary to determine the performance of the overall system taking into account the interaction of the various subsystems and the environmental conditions. In civilian harbour protection system design, one has the goal of reaching the required protection level at the minimum cost, hence it is crucial to obtain an operative assessment of the system performance at the design stage.

In order to bridge the gap between acoustic and magnetic systems data-sheet specifications and operational system performance, the authors have developed in the last years an underwater anti-intrusion system simulator that couples acoustic and magnetic prediction models, environmental information, static and dynamic event-driven simulation [2, 3]. In this contribution a parametric study is conducted in a simplified situation to show some of the design trade-off faced by an anti-intrusion system. In particular, it is shown how the decrease in detection range due to a thermocline in shallow water requires an increase in the speed and performance of the reacquisition, classification and deterrence system in order to maintain the same level of protection, while increase of the source level of the detection sonar has negligible effect on the overall performance.

2. HARBOUR INTRUSION SIMULATOR SYSTEM

The harbour intrusion simulator has been developed integrating acoustic and magnetic sensor models and a dynamic simulator into a GIS architecture. The tool allows creating a complete underwater surveillance system scenario, including fixed sensors, re-acquisition vehicles, underwater threats, and a detailed geographical characterization of the harbour. Each fixed sensor is characterized through a Probability of Detection (PoD) as a function of range, azimuth and depth from the sensor itself; such PoD is obtained by simulation of the acoustic and magnetic sensors performance, giving as input the specific harbour environment characteristics (coastline, bathymetry, weather conditions, etc), standard characteristics of acoustic and magnetic systems (as frequency, bandwidth, beam-pattern, source level, etc.), and intruder trajectories and features (acoustic and magnetic target strength, speed, etc.). The performance of each sensor is computed first in terms of propagation loss, which is then converted in PoD as a function of range/azimuth through the sonar equation or the magnetic equivalent of the sonar equation. In particular, the RAM model [4] has been used to predict the performance of passive acoustic systems, while a custom modified version of the Espresso code [5] has been used to model active acoustic sensors. Custom-built simplified magnetic models have also been implemented, but they have not been used in this paper and hence are not described here. When the sensor deployment has been defined the simulator computes the overall detection probability in the harbour, composing the PoD functions of each sensor. Patrol boats represent the second element of the anti-intrusion system and are
equipped with on board sonars (forward or side looking) to re-acquire, classify and deter the intruder once it has been detected from the fixed sensors.

After having completed the scenario configuration, the anti-intrusion system is ready to be tested through the dynamic simulator. The simulation can be carried out in Monte-Carlo fashion and begins with an intruder trying to intrude while the fixed sonars are continuously scanning the harbour area. When any of the fixed sensors have detected a possible threat, a patrol boat (the closest one) is alerted and it moves towards the alarmed area to start the re-acquisition operations. If the vehicle detects the intruder, it passes to the classification phase and, if necessary, it attempts to stop the threat. At the end of each dynamic simulation the tool returns the probability to stop the intruder, computed as relative frequency of successful cases in which intruders have been stopped over the entire number of simulations; the time to stop the intruder; the halt distance, computed as the distance from the end of the trajectory at which the intruder is stopped.

3. SIMULATION RESULTS

In this paragraph, results from a parametric study are reported in order to evaluate in a quantitative manner several performance trade-off in underwater harbour protection systems. In the following the basic system components considered are:

- An asset equipped with a sonar for self-protection.
- A fixed active acoustic sonar (DDS) for threat detection and tracking. The sonar is co-located with the asset to be protected. It is placed at 4m depth, with 360° coverage; it operates at 100kHz with a pulse length of 0.1ms; two transmitter source levels (SL) are considered, one at 220dB/μPa@1m and one at 235dB/μPa@1m (as a case limit). The transmitter vertical beam width is 15°. Finally the receiver has a horizontal beamwidth of 2.5°, a bandwidth of 10kHz, and a directivity index of 30dB.
- A patrol boat (PB) activated and guided towards the threat by the fixed sensor, for threat re-acquisition, classification and deterrence. The vehicle is patrolling along a circular path surrounding the ship. Anyway, when the simulation starts, its position is forced to be in its worst location with respect to the intruder arriving path: the PB initial point is extracted with uniform probability from the straight line placed on the opposite side of the ship (marked in green in Fig. 1).
- A forward looking sonar (FLS), as the sensor aboard the re-acquisition vehicle, with an horizontal aperture of 120° and characterized by a constant detection probability (PoD) of 90% up to a range of 40m, then linearly decreasing to 0% at a distance of 50m, for any depth from 0 to 15m; constant classification probability 70% up to a range of 40m, then linearly decreasing to 0% at a distance of 50m, for any depth from 0 to 15m. Detection/Classification time $t_{DC} = 5s$, as the time required by the vehicle to perform detection and classification operations. FLS performance are decreased in simulations until having a $PoD=70\%$, a $PoC=60\%$, and a $t_{DC} = 10s$.
- An underwater threat which moves, in all simulations, along a straight course towards the target, at 10m depth, and with a target strength of -20dB.

The harbour is characterized avoiding additional complications due to bathymetry and coastline. The sea bottom is flat with a constant depth of 15m. Fig. 1 shows the scenario.
Two environmental cases are simulated, one winter and one summer. In both seasons good weather conditions are considered (no wind and no rain) while they differ for the sound velocity profile used: an isovelocity profile is considered in winter; a profile with a main thermocline starting at 10m is used in summer. Finally, the seabed is composed by medium sand in the whole harbour.

All the simulation parameters are computed as functions of the intruder speed $v_0$ as reported in the following:

- **Patrol boat approaching speed** to the alarm point: $v_a = \frac{v_a}{v_0} = k_1v_0$
- **Patrol boat searching speed**, for re-acquisition operations: $v_s = \frac{v_s}{v_0} = k_2v_0$
- **Time to Transit**: $t_T = \frac{R}{v_0}$, as the time required by the intruder to get to its target from a distance $R$ that is the maximum range of the fixed sonar at which its detection probability is greater than 10%.

Fig. 2 reports the behaviour of $k_1$ (left), $k_2$ (middle), and $t_T$ (right), as a function of the intruder speed $v_0$. As the sonar range $R$ decreases so does $t_T$: the less distance has to be covered by the intruder to get to the asset, the less time the anti-intrusion system has to detect, classify and stop the threat.
System performance are evaluated through pseudo-Monte-Carlo simulations (30 runs per scenario) and compared as functions of the ratio between PB approaching speed and intruder speed, onboard forward looking sonar performance, fixed sonar transmitter source level (SL), and changing environmental conditions. The anti-intrusion capability to stop the threat is shown through the residual time to transit, \( rt_T \), defined as:

\[
rt_T = \frac{R}{v_0} - (t_R + t_D + \Delta t_{DC})
\]  

(1)

Where \( t_R \) is the anti-intrusion system time to react which adds up all the time delays in PB operation, as track formation time with any sensor, time for PB to unlock, to start its engine and to go full speed, time to slow down and activate its sonar; \( t_D \) is the PB deterrence time that is the time required to deter the threat; \( \Delta t_{DC} \) adds up the time required by the fixed sonar to detect a possible menace and the time necessary for the vehicle to detect and classify the intruder (\( t_{DC} \)). In all simulations \( t_R=20s, t_D=60s \).

The first set of simulations is executed in winter. The anti-intrusion system residual time to transit, \( rt_T \), is shown in Fig. 3 for \( SL=220dB \) (left) and \( SL=235dB \) (right). In winter the system is always able to stop the menace, and, as it can be expected, the anti-intrusion performance decreases with the PB sonar one, showing an asymptotical convergence towards the intruder time to transit, \( t_T \). In summer, the scenario changes completely since the probability of detection of the fixed sonar degrades dramatically. In this latter case, the sonar maximum range is about \( R=240m \) for both transmission source levels and it does not allow the system with the described characteristics to always protect the asset. Fig. 3 depicts the \( rt_T \) obtained in summer with \( SL=220dB \) (left, bottom diagram) and \( SL=235dB \) (right, bottom diagram). Fig. 4 shows the percentage of intruder victories as a function of velocity ratio between PB approaching speed and intruder speed, and vehicle onboard sonar performance. Specifically, supposing that the vehicle is fast enough with respect to the intruder, a smaller \( t_{DC} \) (very quick operator) increases the average halt distance.

![Fig. 3: Residual time to transit as a function of PB approaching speed. In summer the system performance degrades dramatically and a smaller \( t_D \) increases the \( rt_T \).](image-url)
4. CONCLUSIONS

The simulation study reported shows how the differences in environmental conditions may deteriorate the performance of an anti-intrusion system. Moreover, it makes it clear that the overall performance is related to the interplay between the difference subsystems. In the simplified case considered in the paper, it appears that the source level of the fixed detection sonar has little effect on counteracting the adverse environmental conditions, while it is crucial to have a fast approaching speed of the Patrol Boat for reacquisition and classification, with an efficient on-board detection/classification system.

REFERENCES

SONAR PERFORMANCE: TREATING OBSERVATIONS OF DETECTION RANGE AS MEASUREMENTS

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Abstract: The range of first detection of an approaching target is among the most important parameters defining sensors and defensive surveillance systems. A demonstration of capability therefore typically includes a demonstration of a sensor’s detection range, to show that the range of first detection occurs at notably greater range than competing systems, or to show that it exceeds minimum performance requirements. Such demonstrations are often carried out for the commercial sonars designed for underwater surveillance in port protection. Like any measurement, the observation of detection range is subject to unpredictable random variability, and this variability is important whenever practical conclusions are to be drawn from observed detection ranges, such as comparing the performance of one sonar against another or against minimum performance requirements. The variability also has implications for operations because the response measures taken by security forces against underwater intruders must be robust against the degree of variability that can be expected in their detection range. Here we report a sequence of detection ranges observed for repeated diver approaches for a commercially available sonar under realistic conditions during military experimentation for port protection. A probabilistic model is fit to the observations to explain much of their variance, and to extend the implications of such variance to performance evaluations and waterside security operations more generally.

Keywords: Detection range, underwater intruder, sonar performance.

1. INTRODUCTION

An essential capability in port protection is the enforcement of an underwater security exclusion zone. The threats envisioned are attack by divers, possibly with propulsion vehicles, surface swimmers, and autonomous underwater vehicles. The assets protected may
be military and civilian. Underwater surveillance is required when barriers are unavailable or provide insufficient coverage. A number of commercial diver detection sonars exist for the purpose, the leading class so far being monostatic active sonar with signals on the order of 80 to 120 kHz. Many analysts have been evaluating diver detection sonars to assess state of the art technology and often to buy the “best” sonar.

Performance assessments typically involve staged “intrusions” toward the sonar. Analysts arrange for divers to swim toward the sonar from some distance away in order to demonstrate the sonar functioning under realistic conditions, assess false alarm rates, see automatic detection at work, and above all to see the range of first detection. Inferences are then drawn about the expected detection range during real operations, and about the performance of one sonar relative to another or to minimum requirements. The observations will be inconclusive and open to biased interpretation if random measurement uncertainties are not taken into account. At a minimum, the analyst must plan to conduct enough repeated diver approaches for a significant sample size enabling unbiased inference. For security providers, on the other hand, the variability in detection range variability in part determines the waterside area in which their response measures against underwater intruders could be called into action.

The sources of environmentally induced variability are well known in ocean acoustics. Target strength is likewise a longstanding problem in ocean acoustics. These are usually minimized as much as possible during sonar evaluations by conducting repeated diver approaches for a short time, in the same area, with the same divers. A third, more intrinsic variability in detection range arises due to the probabilistic nature of detection, especially when the sonar system must work to keep false alarms low. This variability has apparently passed largely unnoticed. It is the subject here.

2. EXPERIMENT

While conducting experiments for unmanned response in port protection (Eckernforde, DEU, Nov 2009), NURC\(^1\) and WTD-71 (DEU) had occasion to stage repeated diver approaches toward a commercial sonar, not to assess its limits of detection range performance but to exercise the sonar in an unmanned intruder response system. The sonar was used to vector an unmanned surface vehicle into position to classify and closely observe intruders while issuing warnings to prove hostile intent. In the process a number of diver runs were staged along the same path over the course of a few days. The paths started from the same range and much the same bearing, in this way keeping environmental variation to a minimum. Divers always wore the same re-breather equipment. They swam in pairs, tethered together for safety because they did not use floats (see left in Fig. (1)). Thirteen diver approaches \((N_{\text{dive}} = 13)\) were staged with the commercial detection sonar in operation to detect and track them, and to vector in the unmanned surface response vessel. Measuring ultimate detection range was not an objective, so the diver starting range, \(R_0 = 324\) m from the sonar, was known to be well within the coverage of the sonar. It is the variability in detection range for repeated diver approaches that is of interest here.

The observed detection ranges are listed in Table. (1). These are the ranges at which the sonar’s automatic detection algorithms first took notice of the diver, some time before the

\(^1\) NURC military experimentation funded by NATO Allied Command Transformation (ACT) Experimentation Programme of Work 2008 (EPOW 2008).
algorithms decided to raise a warning and assign a track. Automatic track assignment was considered to be classifying a contact already detected. The sonar software gave both the detection and classification ranges in a magnified viewing window (right of Fig. (1)). The human sonar operator was alerted in advance to the divers’ approach in both range and bearing, and could often detect the divers in the sonar echograph before the automatic algorithms, but the human operator’s call was not used in this analysis because alerted or constant close and vigilant inspection of the echograph by an operator is not a realistic option for most security operations. The frequency distribution (histogram) of the sonar’s automatic detection ranges observed during the experiment is given in Fig. (2). The smooth line and uncertainty bars in Fig. (2) come from the subsequent analysis.

![Fig. (1) Right: Annotated zoom window captured from the diver detection sonar, showing how the first detection of an approaching diver was measured using the auto-detection algorithms of the sonar. The approaching paired diver track is vertically downwards. The track of the dive-tender boat is seen at the top. Left: High resolution image of divers taken from unmanned vehicle vectored for “prosecution in to 12 m range by the diver detection sonar”.

Table (1) Observed detection ranges. The uncertainties reported for average and standard deviation were estimated from the theory, after a curve was fit to the data.

<table>
<thead>
<tr>
<th>Observed Detection Range (m)</th>
<th>212.4</th>
<th>178.8</th>
<th>253.2</th>
<th>276.2</th>
<th>271.6</th>
<th>270.3</th>
<th>295.3</th>
<th>212.7</th>
<th>289.7</th>
<th>172.0</th>
<th>250.0</th>
<th>265.1</th>
<th>193.7</th>
</tr>
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<tbody>
<tr>
<td>Average</td>
<td>242.1 m ± 22.5 m</td>
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<tr>
<td>Standard Deviation</td>
<td>40.3 m ± 19.3 m</td>
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</table>
3. ANALYSIS

We treat the sonar’s inner workings (signal processing and auto-detection algorithms) as a black box unknown to us. We do know something of acoustic propagation, however, when sonar performance reverberation limited, that is, when the sonar nominal frequency is high and the target is many water depths away in range from the sonar. The diver target strength to local (diver vicinity) reverberation ratio is roughly constant over a wide ranges from the sonar if the seafloor is flat and of constant make up (gravel, sand, or mud) across those ranges. This is perhaps a little known principle for port protection and it warrants more treatment than can be given here. Although the diver (target) echo increases with decreasing range, the seafloor reverberation in the vicinity of the diver also increases with decreasing range, and it does so at much the same rate. The principle was pointed out theoretically in [1], and empirically by an expert sonar developer and operator during sea trials [2], and it was confirmed informally by an expert sonar modeller [3]. The implication is that the sonar’s ability detect the diver amid reverberation is roughly constant with range, or is at least slowly varying across relatively long distances as the diver approaches along a constant bearing. This could be shown using geometric propagation if space permitted, but for now let the justification instead be the plausible fit (such as it is) between the theoretical curve to the observed data in Fig. (2).
Given an undetected diver many water depths from the sonar, the likelihood of detection increases as the time \( t \) spent in the field of view increases. That increasing likelihood is the outcome of a growing sequence of “looks” at the approaching diver. Let the number of looks on the moment of first detection be

\[
n = \frac{R_0 - r(t)}{vT} \quad (1)
\]

were \( R_0 \) is the starting range of the diver at his or her entry point into the field of view, \( r(t) \) is the range of the diver at time \( t \), and \( v \) is the speed of the diver along constant bearing. \( T \) is the effective time between independent looks. \( T \) is greater than or equal to the ping frequency of the sonar, with equality holding if independent “looks” accumulate at the same rate as the sonar pings.

Let the probability of making a detection on a look be \( p_d \). Then the probability of detecting the diver on look \( n \) is

\[
p(r) = p_d(1 - p_d)^{n-1}. \quad (2)
\]

It can be shown that the expected value of \( n \) is \( 1/p_d \), from which it follows that the expected (average) detection range is

\[
n_{avg} = \frac{1}{p_d} = \frac{R_0 - r_{avg}}{vT}. \quad (3)
\]

The curved line in Fig. (2) is the probability distribution (2), scaled by \( 1/(vT) \) owing to (1), with \( T \) as a first guess set equal to the time between pings during the experiment, with \( v \) equal to the nominal speed of the swimming divers (one knot) \( v = 0.514 \) m/s, and with \( p_d \) derived from (3) using \( r_{avg} \) from Table (1). One finds that \( p_d = 0.019 \) for these trials. It is small because the sonar system is adjusted to keep false alarms very low. If \( p_d \) were high and false alarms were low, then we would not be dealing with a challenging surveillance problem. The overall (saturated) probability of detection may nevertheless be very high because there are many looks at the approaching diver. \( p_d \) is an effective probability of detection, apportioned across pings on the diver, based on observations of detection range. It includes the overall effect of all signal processing in the sonar. This is a dramatic simplifying departure from the complexity of the signal processing taking place in the sonar. Its justification comes above all from its fit to the observed data in Fig. (2).

The uncertainty bounds in Fig. (2) were also derived from the model distribution (2). They mark the upper and lower bounds on the histogram of observed detection ranges when, as in the experiment, only thirteen divers are used to generate the frequency distribution of observed detection ranges. In other words, if a measurement of thirteen detection ranges was repeated many times, and the distribution of the 13 detection ranges was plotted each time, then the distributions would fall between the indicated uncertainty bounds 90 \% of the time. The uncertainty bounds are integer values so it is not unusual to find the limits of the uncertainty bounds to overlap the observed number of occurrences, as they do at points in figure. The bounds are wide because the sample size is low. The model distribution nevertheless permits us to speak in general, approximate, quantitative terms about the variance of detection range, and this is what we want.

4. IMPLICATIONS OF VARIANCE IN DETECTION RANGE

Using the distribution (2), the variance of the number of looks \( n \) on detection is
\[ \sigma_n^2 = \sum_{i=1}^{N} p_d (1 - p_d)^{-1} \left( i - \frac{1}{p_d} \right)^2 = \frac{1 - p_d}{p_d^2}. \] (4)

The standard deviation of detection range \( r \) is therefore
\[ \sigma_r = \sqrt{\frac{1 - p_d}{p_d}} = \sqrt{(R0 - r_{avg})} \approx (R0 - r_{avg}). \] (5)

The approximation holds when the distance travelled \( vT \) between looks is much less than the average distance travelled \( (R0 - r_{avg}) \) before detection. Given that \( N_{divers} \) were used to create an average detection range \( r_{avg} \), the standard deviation (uncertainty) in \( r_{avg} \) is
\[ \sigma_{r_{avg}} = \frac{\sigma_r}{\sqrt{N_{divers}}} \approx \frac{(R0 - r_{avg})}{\sqrt{N_{divers}}}. \] (6)

When we assess the detection range of a diver detection sonar by staging \( N_{divers} \) approaches toward the sonar, with each approach starting from the same range \( R0 \) and along the same bearing, the uncertainty bounds (accuracy) of the resulting performance metric \( r_{avg} \) is given roughly by (6). In the present experiment with \( N_{divers} = 13 \), for instance, we find \( r_{avg} \pm \sigma_{r_{avg}} = 242 \pm 22.5 \) m—an almost 50 m uncertainty on a detection range of about 250 m, or 20%. Four times as many divers \( (N_{divers} = 52) \) would be required to reduce the uncertainty (improve the accuracy) in the measured performance to \( \pm 12.5 \) m, or 10%. This quantity would not be practical in most cases.

\( r_{avg} \) and its uncertainty bounds \( \pm \sigma_{r_{avg}} \) are both of utmost importance when comparing one sonar against another, or when comparing a given sonar against minimum performance requirements. It may be that the number of divers is simply too small to conclusively rank the observed performance of two different sonars, or to compare a sonar against specified requirements. (6) tells us when this is so. If three rather than thirteen divers were used in the present experiment, then the outcome would have been on the order of \( r_{avg} \pm \sigma_{r_{avg}} = 242 \pm 46.8 \) m—an almost 100 m uncertainty, which leaves little to be said conclusively about performance relative to other sonars or requirements.

Equation (6) can be used during the planning of a measurement of detection range to estimate the number of divers required. For instance, given a rough initial guess about the prior expected detection range \( r_{avg} \) for a given starting range \( R0 \) (\( R0 \) presumably being the largest range scale setting for the sonar, and \( r_{avg} \) some fraction of it suggested by the sonar manufacturer) one can solve (6) for the required number of divers \( N_{divers} \). The number can be quite large for high accuracy.

It is possible to estimate the range at which a given the percentage \( P \) of divers will be detected. The proportion \( P \) of detections on or before \( n \) looks is given (from (2)) by the cumulative probability distribution
\[ P(n) = \sum_{i=1}^{n} p_d (1 - p_d)^{n-1} = 1 - (1 - p_d)^{n} \] (8)

Substituting (1) and (3) into this equation and solving for range \( r \) gives
\[ r(P) = R0 - vT \frac{\log(1 - P)}{\log \left(1 - \frac{vT}{R0 - r_{avg}}\right)} \] (9)
Using the data for the present experiment, for instance, we find that one can expect that 50% of divers starting from range $R_0$ would be detected by range $r(P=0.50) = 268$ m (somewhat before the average $r_{avg}$ because the distribution is not symmetric about the mean), and 90% would be detected by range $r(P=0.90) = 137$ m. Security forces would therefore have to provide response coverage over a range of almost 200 m $(324 - 137 = 187$ m) from a diver entry point at $R_0 = 324$ m. $R_0$ was not the maximum range scale setting in this experiment, but if it were, then one would have an estimate of the window of ranges that security providers must cover for first detection of intruders.

5. CONCLUSIONS

To observe automatic detection and tracking algorithms in action (snapshot on right side in Fig. (1) for instance) gives the impression that there may be a sudden onset of detection capability with range from the sonar. That is, the detection suddenly appears and the track is continued without loss, as if the point of detection signalled the diver’s sudden entry into a zone of high detection performance. For analysts familiar with the sonar equation, moreover, the impression reinforces an overly literal vision of the sonar equation in action: that the sonar suddenly detects and locks on to the approaching target at a range completely determined by the environment (transmission loss and reverberation level), target strength, sonar power, and detection threshold, with negligible randomness. That impression, and the extreme effort it sometimes draws from analysts toward environmental and target strength assessment, is false. Track lock does not occur because of a sudden onset of detection performance at a definite range. It occurs because the system’s signal processing concentrates itself at the moment of first detection into much more efficient modes of inspection in the immediate vicinity of a detection. The data (Fig. (2)) confirms that there is a fairly uniform but random capability for first detection that is spread across rather wide ranges, perhaps hundred’s of meters, even when the environment and target strength are controlled. This variability would not occur if one were facing an easy detection problem, when false alarms are naturally low and $p_d$ was high.

The inherent variability faced in practice must be taken into consideration when designing an experiment to assess the detection range of sonars under realistic conditions. Here it was shown how the uncertainty in a measurement of average detection range can be estimated (6), and how to estimate the number of divers required in an experiment to achieve a desired accuracy in the measured sonar performance. The analysis helps this way in the design and interpretation of detection range experiments for port protection.

The analysis also highlights the critical role of the starting range $R_0$ of divers when assessing detection range. Analysts must control and record $R_0$ during runs if comparisons between sonar performances are to be made. This has rarely been done in practice. When testing the limits of sonar performance, $R_0$ should be greater than or equal to the maximum range scale setting of the sonar. The analysis also highlights the importance of having a small sonar “look” period $T$ (proportional to the time between pings). Since $T$ generally increases with the range scale setting it is always advisable during security missions to use the smallest possible range-scale setting. The setting should provide adequate range coverage in the port, but not very much more.
The analysis is admittedly preliminary, which is to say, founded on experience with diver detection using different commercial sonars and on a small data set, with some bold approximations thrown in. All of this was necessary to make analytical headway into the proper design of detection range measurements for port protection. The analysis may find application in modified form to other underwater detection problems, such as for sea mines or submarines. Insofar as there exists a significant, unavoidable randomness in detection range under realistic conditions, there also exists a practical limit to the accuracy to which the environment and target strength need to be known for operational forecasts of detection range. One has sufficient information about the environment and target strength information once the effect of their remaining uncertainty on detection range falls below its intrinsic uncertainty.

REFERENCES


[2] Private discussions with Dr. Nigel Peach, QinetiQ, who then demonstrated the principle during technical evaluations of an early version of the QinetiQ Cerberus sonar at NURC, April 2006.

ACOUSTIC RESEARCH FOR PORT PROTECTION AT THE STEVENS MARITIME SECURITY LABORATORY

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Abstract: Stevens Institute of Technology has established a Maritime Security Laboratory (MSL) as a national laboratory resource for government, industry, and universities to advance technologies for the protection of USN maritime infrastructure. Experiment instrumentation includes research vessels, a multiplicity of hydrophones and emitters, stand alone acoustic buoys, diver acoustic simulators, unmanned underwater vehicles (UUVs), and precision instrumentation placement capabilities. The in-river experiments are controlled remotely from a Visualization Centre on campus. Acoustic research is supported by sound speed profile measurements, integrated video and acoustic tracking of surface events, and global positioning system tracking of live divers. Recent results include determination of parameters defining the detection distance of a threat: source level, transmission loss, and ambient noise. The combination of acoustic noise with video data for different kinds of ships in the Hudson River enables estimation of sound attenuation in a wide frequency band. The establishment of a library of various estuarine signatures, including divers, boats, ships, UUVs, construction equipment, and so forth, is underway. This knowledge can be used in a variety of intruder detection scenarios and for optimal methods of threat detection.

Keywords: Passive acoustics, port security, intruder detection.
1. INTRODUCTION

In 2006, Stevens Institute of Technology established a research laboratory environment in support of the U.S. Navy in the area of Anti-Terrorism and Force Protection (AT/FP). Called the Maritime Security Laboratory, or MSL, it provides the capabilities of experimental verification of AT/FP research in the realistic environment of the Hudson River Estuary. The goals of MSL are:

- To continuously advance the state-of-the art in technologies key to maritime security in an estuarine environment
- To develop transportable intruder detection prototypes embodying results of new maritime security research, deployable to harbors around the world for military and commercial applications
- To become a national resource in the maritime security for the US Navy and also for the domestic security, maritime industry, and natural hazards mitigation communities.
- To develop a work force for the future through extensive involvement and education of students, post-doctoral, and academic and research faculty in the area of maritime security.

Initially, the focus of MSL was on threats posed by surface and subsurface intruders including SCUBA divers and small boats by using passive acoustic techniques [1-3]. Using these initial capabilities, MSL investigated the set of acoustic parameters fundamental to underwater acoustic threat detection including: diver acoustic signature, acoustic transmission loss, and acoustic environmental noise. The initial infrastructure has since been extended to include computer optic and infrared vision capabilities, and to enhance acoustic experiments by combining them with these capabilities.

These integrated capabilities have enabled experiments in determining the positions and trajectory of surface traffic, which may be both a sources of acoustic noise in intruder detection as well as possible targets themselves. This knowledge can be used for measurements of acoustic noise of various ships and their classification, determination of sound attenuation in a wide frequency band, and the development and testing of methods of passive acoustic triangulation and location of sources of sounds [4,5].

2. THE MARITIME SECURITY LABORATORY

Part of the uniqueness of Stevens’ Maritime Security Laboratory is its location on the Hudson River tidal estuary, which is a key waterway that defines the Port of New York/New Jersey, one of the busiest harbors in the U.S. From a scientific perspective, this harbor embodies a high degree of complexity due to variability of the current, salinity, temperature, winds, turbidity, as well as man-made factors including ambient noise due to surface and air traffic, construction noise, and various forms of electromagnetic radiation. All of these enter into the analysis of above and below surface threats.

Hence the estuary itself is an integral part of the laboratory. As discussed above, the estuary is equipped with instrumentation to collect weather and environmental data, and through modeling, to predict their characteristics. For the actual MSL execution of
experiments, the test site has been chosen based on its scientific characteristics and its accessibility both by radio communications and by safety considerations. The MSL research vessels and other MSL assets are shown in Figure 1. The larger boat is the RV Savitsky. It is specially constructed and fitted out for maritime research purposes. Towards the stern is an A-frame for loading large and heavy items onto and off of the boat. Radio antennas are affixed to the mast to transmit real-time experiment data to the MSL Visualization and Analysis Center (VAC). The smaller boat, the Phoenix, is a support boat. It is used to deploy sensors while they are cabled to the Savitsky. It is also used to deploy remote instrumentation, divers, and provide for safety. In addition, it is used as the point of radiation in experiments involving acoustic propagation between two points and measurements of temporal variability of acoustic field.

Fig.1. MSL assets applied for acoustic measurements in NY estuary

The MSL key components are shown in Figure 2. Starting at the left, various sensors are deployed at the test sites. Depending on the experiment, these may include hydrophones, sound emitters, CTD’s (for conductivity, temperature, and depth), Acoustic Doppler Current Profilers, inclinometers (to measure the roll of the boat), various radio link instrumentation, and so forth. Experiment-specific instruments are cabled to an on-board boat computer, or are connected wirelessly.
Communications with the boat is accomplished with an IEEE 802.11 (WiFi) radio link. This link is used to enable real-time data to be transmitted to the Visualization and Analysis Center (VAC). This enables experiments to be controlled from the VAC, in terms of when and how long data is recorded. Perhaps most importantly, it provides Data Quality Assurance, to assure that at the end of the day, good data has been collected. Another important application of the radio link is to maintain real-time communication with the boat crew during experiments. This is accomplished by establishing a chat line with the boat, and is critical to the logistics and administration of experiments. The radio link is connected to the VAC over the campus network.

The Visualization and Analysis Center has several major purposes:

- To provide the capability to administer and control experiments, whether on the boat, or elsewhere
- To ensure data quality assurance during experiments
- To enable the ability to reconfigure experiments in response to the data received. (This capability will become more significant as we undertake experiments in adaptive learning)
- To provide an environment for research, algorithm development, and laboratory infrastructure improvements
- To provide a demonstration capability for key stakeholders and potential customers and users.

In addition to real-time data feeds into the VAC, six video cameras have been deployed to provide real-time visual observation of experiments, as well to provide a video data source to automatically collect data, and to analyze surface and low flying aircraft traffic, as well as intruder activity. Surface Traffic Tracking System has been developed by Stevens’ scientists utilizing these video cameras, which automatically detects the entry of surface craft into a calibrated sector of the estuary. Utilizing this system, the position, bearing, and velocity of
traffic can be automatically detected and recorded. Path information on detected traffic is projected onto a Google map as well as stored in a data base. By observing the acoustic signature of passing craft via hydrophones, and correlating that with the position information recorded by the Tracker, one can determine, for example, the transmission loss of the path between the source (the passing craft) and the receiver (the hydrophone) [4,5].

The intruder detection problem is complicated by the high degree of spatial and temporal variation of an estuary due to tides, winds, currents, precipitation, traffic, power plants, and so forth. The complexity of this environment requires that real data be used in its modeling. Such data-driven mathematical models have been built by Stevens and are used to predict oceanic and atmospheric environmental factors. The model, the Stevens New York Harbor Observation and Prediction System (NYHOPS), can be found at http://hudson.dl.stevens-tech.edu/NYHOPS/. The interrelationship between NYHOPS predictions of acoustic parameters and MSL experimental measurements was studied over a 12 hour tidal cycle. Results of sound speed and Transmission Loss calculation are placed at Stevens website and Fig. 3 presents the example of this calculation.

![Image](http://hudson.dl.stevens-tech.edu/NYHOPS/)

**Fig. 3. The example of sound speed prediction and calculation of Transmission Loss placed at [http://hudson.dl.stevens-tech.edu/NYHOPS/](http://hudson.dl.stevens-tech.edu/NYHOPS/)**

### 3. BRIEF OVERVIEW MSL ACOUSTIC EXPERIMENTS

In our tests, hydrophones were placed at various heights in the water column, or on the river bottom on stands. All deployed hydrophones were connected by cable to the on-board computer for data processing and storage. The signals from the hydrophones were amplified and filtered for suppression of the high acoustic noise level in the low frequency band, which limits the dynamic range of measurements and for elimination of spurious aliasing signals
produced by electromagnetic noise at frequencies above 100 kHz. The amplified and filtered signals were digitized by an 8 channel data acquisition system and recorded. The boat computer was wirelessly connected with MSL Visualization and Analysis Center, so all information displayed on the boat computer was displayed simultaneously in VAC. This allowed scientists in the VAC to control the experiments.

The acoustic propagation experiments were conducted by radiating an acoustic wave between a transmitter and receiver. The radiating wave was generated by a calibrated emitter. The following experiments were conducted in this way:

1. Measurements of transmission loss in a wide frequency band (20-100 kHz) in Hudson River were carried out, including the effects of tidal variation.
2. Determination of the shallow channel impulse response using correlation techniques. The measured impulse response is used by Stevens’ researchers for estimation and prediction of underwater acoustic communication systems performance.
4. Ambient acoustic noise was measured in various environmental conditions and for water traffic. The joint application of acoustic measurements and video surveillance allows determination of acoustic noise produced various kinds of ships and application of ship noise for acoustic attenuation measurements. Some of these results are presented in the current paper and the paper.
5. The received data for diver source level, acoustic attenuation and noise were applied for estimation of a diver detection distance.

4. ACKNOWLEDGEMENTS

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MODEL FOR FATIGUE AND FAILURE OF HUMAN LUNG TISSUE SUBJECTED TO LOW-FREQUENCY UNDERWATER SOUND

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Abstract: A finite-element model of human lung response to low-frequency underwater sound is presented. The elastic properties of the lungs are calculated using a micromechanical model of lung parenchyma which accounts for the geometry of the alveoli, elastic fiber bundles within the alveolar septa, and lung surfactant. Simulations are compared to published results for mice and used in part to calculate a preliminary acoustic damage threshold curve for humans as a function of frequency. The model is used to study progressive lung damage that occurs over multiple acoustic cycles by modifying the elastic properties of the lungs for each iterative solution based on the amount of induced strain from the previous iteration.

Keywords: Human bioresponse to vibration, lung damage, lung resonance

1. INTRODUCTION

The objective of this research is to predict the bioresponse of human lung to low-frequency underwater sound with the goal of predicting acoustic thresholds below which injury may be avoided. Humans in water have a lung resonance frequency around 50 Hz [1], at which frequency the wavelength in water is approximately 30 m, which is two orders of magnitude larger than the human lung. The very low resonance frequency of the lungs is due to their high compressibility.

An accurate model of lung damage due to low-frequency underwater sound may also depend on exposure duration. The time-dependent nature of mammalian lung damage due to acoustic excitation at the resonance frequency of the lungs has been demonstrated experimentally [2], and this was deemed an important factor to investigate in this work.
In the present work, a computational model is presented that is focused specifically on predicting the response of human lungs to acoustic excitation near resonance. A micromechanical model of human lung parenchyma is used to calculate the effective medium properties, and these properties are used in a finite-element model of the torso that includes the ribs, bronchiolar tubes, thoracic organs, and some abdominal organs. The response of this model to acoustic excitation at various frequencies near lung resonance is calculated.

An additional capability of the finite-element model is introduced which is used for the simulation of progressive lung damage by modifying the elastic properties of the lungs based on the calculated shear strain from previous acoustic cycles. Results obtained from modelling time-dependent damage in lungs exposed to low-frequency underwater sound are presented, and attempts are made to correlate reported thresholds for lung damage in mice to computational predictions obtained from the model for humans.

2. MICROMECHANICAL MODEL OF LUNG PARENCHYMA

The micromechanical model developed in this section is used to calculate effective medium parameters that are used in the finite-element model discussed in Sec. 4.

![Fig 1: The truncated octahedron (left) and the fiber bundle arrangement (right) for the square and hexagonal faces.](image)

Following the work of Dale, et al. [3], we use a truncated octahedron to represent an individual alveolus. Elastic fiber elements are placed along the edges of the polyhedron and across the faces in such a way as to mimic the distribution of elastin and collagen throughout the alveolar septa [4]. Figure 1 depicts the geometry of the truncated octahedron and the fiber bundle arrangements. The fiber bundles are assigned the force-strain relationship used by Carton, et al. [5]: 
\[ F = \frac{-\ln(1-e/\beta)}{\alpha}, \]
where \( F \) is force, \( e \) is strain, \( \alpha \) is a regression coefficient, and \( \beta \) is the strain at which the tissue breaks. In this investigation, \( \alpha = 0.17 \) and \( \beta = 1.3 \), such that 130% is the strain at which the tissue breaks.

Another important consideration in lung modeling is lung surfactant. Surfactant decreases surface tension in the lungs, and its effect is dependent on its concentration at the surface of the thin layer of fluid on the faces of the alveoli. On the scale of an acoustic period, the surfactant does not have time to migrate from the interior of the fluid to the surface and back again, so the concentration of surfactant at the surface is assumed to be a constant in the calculations presented below.

The truncated octahedra tessellate to form a periodic lattice with cubic symmetry. With each alveolus assumed to be filled with air, and with the constitutive relation for the fibers taken into account, the macroscopic elastic moduli of the cubic crystal were calculated to be 
\[ C_{1111} = 1.68 \times 10^5 \text{ Pa}, \]
\[ C_{1122} = 1.57 \times 10^5 \text{ Pa}, \]
\[ C_{1212} = 7.39 \times 10^3 \text{ Pa}. \]
The elastic moduli for the corresponding effective isotropic medium are determined by spatial averaging [6] to obtain...
the Lamé constants $\lambda = (C_{1111} + 4C_{1222} - 2C_{1212})/5 = 1.56 \times 10^5$ Pa and $\mu = (C_{1111} - C_{1122} + 3C_{1212})/5 = 6.6 \times 10^3$ Pa. These values of the Lamé constants define the elastic properties of lung in our finite-element model calculations.

3. MACROSCOPIC MODEL

The finite-element program COMSOL Multiphysics® was used to model the geometry of the lungs and other organs and structures deemed important for this study. Figure 2 depicts the geometry of the model that was created in COMSOL. The model includes the lungs, trachea, primary bronchi, ribs, sternum, spine, and a generalized mass below the lungs which represents the diaphragm, stomach, spleen, kidneys, and liver. In order to maximize the spatial resolution of the model and minimize computation time, the geometry is kept very smooth and also two-fold symmetric. The two-fold symmetry allows for a complete solution while solving over only one-quarter of the entire geometry.

Fig 2: Geometry of the finite-element model used in this investigation. For the purposes of computational efficiency, the model is kept smooth and two-fold symmetric.

The Lamé constants for the lungs are presented above, and we chose $\lambda = 1.56 \times 10^6$ Pa and $\mu = 6.6 \times 10^3$ Pa for the bronchiole tubes and trachea, $\lambda = 1 \times 10^{10}$ Pa and $\mu = 3 \times 10^9$ Pa for the ribs, sternum and spine, and $\lambda = 2 \times 10^9$ Pa and $\mu = 1 \times 10^2$ Pa for the abdominal mass. The abdominal mass and lungs have free boundary conditions while those for the ribs, sternum and spine are fixed. The anatomical model is surrounded by a large spherical domain of water, and the response of the system to acoustic excitation is calculated.

4. SIMULATIONS OF INSTANTANEOUS LUNG DAMAGE

Direct comparisons to experimental data are very difficult given our interest in human lung damage, but investigations of lung damage to mice and rats due to low-frequency underwater sound have been reported by Dalecki et al. [7]. In these experiments, mice were submerged in an acoustic chamber and ensonified at various frequencies and intensities. After exposure, the degree to which their lungs were damaged was quantified and damage thresholds were identified as a function of exposure time, frequency, and acoustic pressure amplitude.
In order to calibrate the finite-element model to the best of our ability, the geometry of the model was rescaled to be of a size similar to that of a mouse, and the acoustic pressure used in the simulation was chosen to match a level deemed by Dalecki et al. to be sufficient for lung damage. The density of the lung, chosen to be 200 kg/m$^3$ for human lung, was changed to 600 kg/m$^3$ for the mouse lung [8]. All other aspects of the model were the same as for the human scenario.

Special attention was paid to shear strain, as this is thought to be a major cause of damage in lung parenchyma [9]. The shear strain field in the mouse lung with a quality factor of $Q = 3$ resulting from a plane wave excitation of amplitude 2 kPa travelling from the bottom of the abdominal organs to the top of the torso was calculated to be on the order of 20%, which is chosen as the threshold of instantaneous lung damage for the remainder of this study. Regions of highest shear strain are found to be near the rib-lung interface, which is to be expected considering the boundary conditions and the considerably different material properties of lung and bone.

A plot of the acoustic pressure required to cause shear strains of 20% in humans for $Q = 2$ [1] is presented in Figure 3. The general trend of this curve is to be expected, considering that the lung and surrounding tissue and water form a simple harmonic oscillator that exhibits surface displacements that are proportional to frequency squared above resonance. The acoustic pressure required to cause shear strains of 20% is at a minimum at the resonance frequency, which is approximately 30 Hz.

![Fig 3: Sound pressure level required to induce strains of 20% as a function of frequency.](image)

### 5. PROGRESSIVE DAMAGE MODEL

Since lung damage is thought to be time-dependent, a preliminary model of progressive lung damage was developed. Additional functions were added to the COMSOL model which search for regions of high shear strain and modify the elasticity of the lungs if the calculated strain exceeds a certain threshold. The procedure works in the following way: an initial solution is calculated using homogeneous elastic properties equal to those calculated using the microscopic model from Section 2, a function searches through lung volume and acoustic phase for regions which have shear strains above a level chosen as the threshold strain for the
onset of damage, those regions have their elastic properties modified to simulate damaged tissue, and the solution is recalculated with the new, now spatially-varying, elastic properties. This process is repeated, and regions found with shear strain above an absolute failure threshold are identified.

In this investigation, a shear strain of 10% is chosen as the threshold for the onset of lung damage and a shear strain of 20% is chosen as the absolute failure threshold. Properties of damaged lung tissue can be predicted using the micromechanical model described in Section 2 by first imposing a large-amplitude uniaxial strain on the periodic structure. As the structure is strained, when the fiber bundles exceed 130% strain they are considered broken and removed from the periodic structure, and the material properties of the new periodic structure are calculated. This process is repeated until all fibers oriented in the direction of the uniaxial strain are broken. The cubic symmetry of the structure studied makes the fiber bundles break in groups, and hence there are only four distinct values of $\lambda$ and $\mu$. The results of this calculation are presented in Figure 4.

![Fig 4: Calculated Lamé constants for various levels of fiber damage.](image1)

The finite-element model was run using an acoustic pressure of 2 kPa, and the simulation was run just 10 times for computational efficiency. The shear strain fields calculated for runs...
1 and 10, as well as a plot highlighting areas which have exceeded 20% shear strain, are shown in Figure 5. Both a cross-section of the lung in the coronal plane and a surface plot of the anterior of the lung are depicted. Damage is again predicted by the model to be concentrated in the regions around the rib-lung interface.

6. SUMMARY

A progressive damage model of lung response to low-frequency underwater sound has been developed using a micromechanical model of lung parenchyma, which allows for the calculation of effective medium properties based on its fine structure, paired with a finite-element solution which allows for more accurate geometrical representation of the entire system on a macroscopic level. Solutions for instantaneous and progressive damage have been given, and a preliminary threshold curve for instantaneous lung damage was presented. This work was supported by the U.S. Office of Naval Research.

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Passive acoustic tracking and classification of vessels in the Hudson River

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Abstract: Stevens Institute of Technology is conducting research aimed at the development of a low-cost passive acoustic system for detection and classification of underwater and surface threats. This system is based on extensive research of underwater acoustics in the urban environment represented by the Hudson River estuary. The experimental system is comprised of a hydrophone array and a stand alone acoustic buoy supported by a specialized Stevens computer vision system for vessel tracking. Stevens recorded a large collection of acoustic signatures of various vessels together with their video records. To track surface traffic, directionality was determined from the cross correlation of acoustic signals recorded by several hydrophones. The vessel triangulation can be done using information from two or more hydrophone arrays. For classification purposes special attention was paid to extraction of the specific parameters of vessel acoustic signatures that can be used for their classification. It was demonstrated that the vessel generates amplitude modulated noise and that the spectrum of the noise envelope contains shaft and blade frequencies and their harmonics. The measurements of modulation were conducted for vessel noisy in the band 10-60 kHz. Distribution of spectral peaks depends on the type of vessel and its speed. This distribution together with an estimation of vessel Source Level can be used for vessel classification. We considered the possible schema of this classification. This material is based upon work supported by the U.S. Department of Homeland Security under Grant Award Number 2008-ST-061-ML0002.
Structured Session 12

Temporal and Spatial Variability of Clutter, Reverberation and Propagation

Organizers: Chris Harrison, Finn Jensen & Peter Nielsen
WAVE NUMBER TRACKING OF MESOSCALE VARIABILITY IN SHALLOW WATER

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Abstract: In the summer of 2006, the US Office on Naval Research sponsored the “Shallow Water ’06” (SW06) experiment. This was a multi-discipline, multi-institutional experiment combining studies on the coastal oceanography and ocean acoustics of continental shelf and slope environments. During SW06, a low-frequency sound source, broadcasting tones between 50 – 175 Hz, was towed along radials from a vertical line array (VLA) of hydrophones and oceanographic sensors. Concurrently, and from the same ship, a towed CTD chain was used to measure water column properties as a function of time and space. The measurements provided by both the towed CTD chain and the VLA allowed for continuous monitoring of the water column sound speed field at the source and receiver during acoustic transmissions. On August 4\textsuperscript{th}, 2006 towed source measurements were made along a 5 km track oriented parallel to the shelf slope. The track was run repeatedly out and back from the VLA at tow speeds ranging from 2 – 10 knots over a span of 10 hours. Over the course of the 10 hours, the oceanographic conditions at the far end of the track changed considerably. In particular, a large intrusion of cold water moved into the area of the experiment. The cold water mass, which has a large impact on the acoustic propagation conditions, was not measured on the VLA. However, the effect of the intrusion was observed by time dependence in the range-dependent wave number estimates from track data recorded throughout the day. The resulting observations suggest ways for tracking mesoscale variability in shallow water from range-dependent wave number estimates. In this paper, these ideas are explored using acoustic simulations with a known bottom model and the measured time and spatially dependent water column properties.

Keywords: inversion, variability
1. INTRODUCTION

In the summer of 2006, the Office of Naval Research sponsored the “Shallow Water '06” (SW06) experiment [1] on the New Jersey shelf area of the North Atlantic. This is a dynamically complex environment, where cool, fresh continental shelf water interacts with warmer, more saline water from the continental slope. The sharp transition, which occurs for both properties near the 100 meter isobath, is known as the shelf break front. Intrusions of shelf water onto the continental slope are an important source of water column variability.

In these complex shallow water environments, high resolution environmental data are required for accurate predictions of acoustic propagation and scattering. Although the water column sound speed may be determined by direct measurements, it can be highly variable both spatially and temporally so that even multiple single point measurements are often inadequate. Thus, methods which can detect temporally or spatially dependent features of the water column are very desirable.

In this work, modal wave numbers are estimated from the pressure field measured on a synthetic aperture horizontal array created by a moving source and recorded on a fixed receiver array. The same 5 km source track was run repeatedly over a span of 10 hours during which time a cold water mass moved into the path. The temporal and spatial wave number estimates from the measured pressure field data clearly track the observed mesoscale feature. Combined with results from computational acoustic data, this observation suggests a way to distinguish between sources of environmental variability.

2. DESCRIPTION OF EXPERIMENT

The acoustic data collected during SW06 were recorded on a 16-channel Vertical Line Array (VLA) spanning the water column between 13.5 and 78 meters depth. Broadcasting continuous tones at 50, 75, 125 and 175 Hz, a J-15-3 low-frequency acoustic source was towed at constant depth from the R/V Endeavor. The source was towed out and back along a 5 km track oriented parallel to the shelf slope and on a radial with respect to the VLA. Repeated runs along the track were made at speeds of 2 – 10 knots over a span of 10 hours. The locations of the ship track and VLA are shown with bathymetry in Fig. 1.
3. CTD CHAIN MEASUREMENTS

During the acoustic experiment, from the same ship, a towed CTD chain was used to measure water column properties as a function of time and space [2]. Measurements provided by both the towed CTD chain and at the VLA allowed for continuous monitoring of the water column sound speed field at the source and receiver during acoustic transmissions. Over the course of the 10 hours, the oceanographic conditions at the far end of the track changed considerably. In particular, a large intrusion of cold water moved into the area of the experiment. The cold water mass, which has a large impact on the acoustic propagation conditions, was not measured at the VLA.

Water column sound speed calculated from the CTD chain measurements are shown in Fig. 2 for the time periods corresponding to the 2nd and 8th runs along the track. The measurements from the 2nd run extend deeper in the water column because the scope of the CTD chain was fixed and the ship was moving at a slower speed when these data were recorded. By comparing measurements, the persistence of specific features can be observed, particularly an intrusion present at 28 meters depth. A second intrusion, located at 50 meters depth moved from a range of 4.5 km during the 2nd run to a range of 2.5 km during the 8th. Measurements of the water column properties for runs 3 through 7 showed that these features moved en masse towards the southwest as the mass of cold water moved in from the northeast.

4. HORIZONTAL WAVE NUMBER ESTIMATION

Range dependent wave number estimation was accomplished using a sliding window auto regressive estimator [3] with an aperture of 2000 meters weighted by a Hanning window.
According to the uncertainty principle, resolution in the wave number and spatial domains cannot be determined to arbitrary precision. The closely spaced first and second wave numbers are especially difficult to estimate given the limited aperture. This difficulty is overcome by taking advantage of the mode shapes: accurate estimates are obtained by examining data from receiver depths such that the surrounding modes are not detected by the estimator due to their very low amplitudes.

The first five wave numbers estimated from the 125 Hz pressure field recorded on two different channels during the 2nd and 8th runs along the track are plotted in Fig. 3. In the top plots, data from channel 9 are used to obtain an estimate of mode one as mode two is in a null at this mid-water depth; in the bottom plots, using data from channel 15, mode one is not detected due to its very low amplitude at this near seafloor depth and an estimate of mode two is acquired. Mode four was not excited for the 2nd run and could not be estimated from either channel depth.

![Wave number estimates from 125 Hz pressure fields recorded on channels 9 and 15 during the 2nd and 8th runs along the track.](image)

**Fig.3:** Wave number estimates from 125 Hz pressure fields recorded on channels 9 and 15 during the 2nd and 8th runs along the track.

Range dependence in the water column is most clearly indicated by the wave number evolution of mode one shown in the top set of plots in Fig. 3. For the 2nd run, the value of mode one appears nearly constant over all ranges. However, closer examination of the data reveals that a slight minimum occurs in the first wave number between 4 and 4.5 km. In contrast, for the 8th run, the wave number value of mode one changes significantly with range. It increases from 0.521 m⁻¹ at near ranges to 0.525 m⁻¹ at ranges furthest from the VLA.

The greater values for mode one indicate a slower minimum sound speed in the water column. The upper bound of the discrete wave number spectrum is defined by the water wave number \( k_w = \omega / c_{w_{\text{max}}} \), where \( c_{w_{\text{max}}} \) is minimum sound speed in the water column. Thus, a decrease in the minimum sound speed of the water column causes an increase in the upper bound of the spectrum, which has the effect of shifting the entire wave number spectrum upward. Low order wave numbers are more sensitive to the water column properties because they propagate at shallower angles and refract more in the water column. As a result, these modes are more influenced by the shape of the sound speed profile. On the other hand, high
order modes propagate at steeper angles and are more influenced by the mean sound speed of the water column. Thus, these modes do not experience the same magnitude change due to the water column variability as the low order wave numbers.

In the absence of detailed water column information, based on the modal evolution described above, we would expect the water column to be relatively benign during the 2nd run along the track. This conclusion is corroborated by the CTD chain measurements. Moreover, the minimum wave number values for mode one at 4 and 4.5 km correspond to water column sound speed field maxima. For the 8th run, the dramatic increase in the value of mode one indicates much slower sound speeds at the far end of the track. Indeed, this observation is confirmed by the CTD chain measurements.

5. SIMULATED WAVE NUMBER DATA

Range-dependent wave numbers were calculated for the SW06 environment using the normal mode code Kraken [4]. A fully range-dependent seabed was used in the calculations so that the spatial evolution of the higher order modal wave numbers could be better understood. The parameters for the seabed were based on inversion results from the SW06 experiment [5]. Wave number values were calculated using the water column properties recorded at the VLA and for the measured sound speed fields shown in Fig. 2. Wave numbers resulting from the range-independent and range-dependent water column sound speed fields are shown by the dashed and solid lines, respectively, in Fig. 4. The simulated wave numbers provide a more detailed indication of environmental variability than the estimates from data because they are not averaged over the 2000 meter aperture.

Modal evolution with range for the fully range dependent environments is consistent with the wave numbers estimated from the data. Although both models had the same underlying range-dependent bottom, the water column was significantly different between the two models. For the model corresponding to the 2nd run, the mode one wave number value was relatively range invariant with a slight minimum near 4.5 km. For the 8th run, the mode one wave number showed a considerable increase with range. On the other hand, the dashed lines indicate that the value of mode one is virtually unaffected by the range dependence of the bathymetry and seabed. The effects of environmental variability on the higher order modes can also be explained. The dashed lines show a decreasing trend in the 4th and 5th wave numbers resulting from range dependence of the bathymetry and seabed. During the 2nd run, the water column was relatively benign and range-dependence in the higher order modes is dominated by seabed variability. However, for the 8th run, these changes are masked by the water column variability causing an upward shift of the total spectrum which competes with the bathymetry and the seabed that would otherwise cause the values for these modes to decrease. Consequently, wave numbers for modes four and five appear to have nearly constant values.
6. CONCLUSION

Modal wave numbers proved to be a powerful indicator of water column variability. This was illustrated by 125 Hz data from the SW06 experiment. It was shown that the relative value of the minimum value of the water column sound speed profile can be inferred from the spatial evolution of mode one. This allowed for detection of a cold water mass that moved into the region over the course of the experiment.

The relative sensitivities of modal wave numbers to environmental parameters have been suggested as a way to identify sources of environmental range dependencies [6]. In this work, the range dependence of the water column was identified based on the evolution of mode one. It was also shown that modes four and five are influenced by both the water column and by bathymetry and seabed properties.

7. ACKNOWLEDGEMENTS

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CLUTTER VARIABILITY DUE TO FISH AGGREGATIONS: MID-FREQUENCY MEASUREMENTS IN THE GULF OF MAINE

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Abstract: Clutter often drives the performance of active sonars operating in littoral environments; however, sufficiently accurate information on local environmental features that generate the acoustic clutter is rarely available. This paper presents results from a September-2008 experiment conducted in the Gulf of Maine which show that by deploying complementary assessment methodologies, the sources and characteristics of the acoustic echoes can be identified. Off one vessel, broadband mid-frequency acoustic backscatter measurements of swimbladder-bearing fish were made with two active systems, a downward-looking (short-range, < 0.2 km) sonar used to image fish at high vertical resolution, and a horizontal-looking (relatively long-range; 1-10 km) sonar used to image fish at high horizontal resolution. Off another vessel, fish were sampled both acoustically and with nets for ground truth. Spectrogram analysis showed that the echoes had resonance frequencies in the 2-3.5 kHz band, consistent with scattering by the principally-observed fish, the Atlantic herring. Examination of the long-range data (normalized match-filter output) revealed strong, spatiotemporally-variable clutter. In combination, these results provide an improved understanding of both fish behavior and the characteristics of fish aggregations as long-range clutter fields.

Keywords: Fish, ocean acoustics, clutter, volume scattering, sonar
1. INTRODUCTION

Active sonar systems use acoustic sources and receivers coupled with signal processing to detect, classify, and track undersea targets. The limiting influence of reverberation and clutter on sonar performance has long been recognized, with acoustic echoes from the waveguide boundaries and biologics capable of both masking desired signals and creating false targets [1]. Characterizing reverberation and clutter is challenging because of the need to account for 3D spatial and temporal characteristics, coupled with a generally incomplete knowledge of the environment. This paper looks at one aspect of the problem: backscattering from shoals of fish in shallow water at mid frequencies (MF; 1-10 kHz).

In-water volume reverberation at mid frequencies is primarily caused by acoustic interaction with fish that have gas-filled swimbladders [2-4]. The acoustic response of these fish depends on its bladder size, which is a function of the fish's size and depth. Since the bladder typically occupies just ~4–5% of a fish's volume, at these frequencies fish can generally be treated as point-like isotropic monopole scatterers, with diurnal depth dependence. Since they are non-directional scatterers at mid-frequencies, bladdered fish can be significant scatterers at low grazing angles, even at relatively low densities. Furthermore, since backscatter from the ocean boundaries at these angles generally decreases rapidly with decreasing grazing angle, scattering by fish can dominate the reverberation. Because of their isotropic scattering response, fish can also be significant bistatic scatterers [5-6].

Fish distribution and behavior (such as daily and seasonal migration patterns) are species dependent and linked to environmental variables, such as ocean temperature, mixed-layer depths, and topography, as well as biological variables such as spawning, food, and predators. Furthermore, commercial and climatological activities lead to fluctuations in fish abundance. Accordingly, archival fishery data alone may be insufficient to yield accurate predictions of fish-based reverberation, especially since fishery management typically only relates to species of (potential) commercial value. Thus, both up-to-date fishery information and in situ sampling are required to make reasonable estimates of the local volume reverberation.

The strength of the volume scattering due to the fish will be strongly dependent on the densities, sizes, and depths of fish present. The dynamism of fish leads to increased acoustic variability, especially for high-resolution sonar systems. It should also be noted that fish in shallow water are more likely to aggregate into isolated schools than in deep water [7] and so appear more clutter-like.

This paper describes an experiment designed to show that by deploying complementary assessment methodologies, the sources and characteristics of fish echoes can be identified. The paper concentrates on coherent echoes (derived from normalized data) rather than the reverberation or scattering-strength maps of incoherent echoes typical of most fish-mapping efforts. The next section describes the experiment, and is followed by a presentation of initial data results and then the conclusion.

2. TEST OPERATIONS

The measurements described in this paper were made in September 2008 on the northern flank of Georges Bank in the Gulf of Maine (Fig. 1a). This site was selected for several reasons: the regular occurrence of a large number of swimbladder-bearing fish; a primary spawning ground for Atlantic herring; and it is the location of previous fish census and
acoustic experiments. The specific study area described in this paper focuses on tracks conducted in water depths of 170-210 m (Fig. 1c). For these tracks, the underlying seafloor is relatively featureless and consists primarily of sandy sediments (mixed with some silt and gravel). The oceanographic conditions were stable with a typically downward-refracting sound-speed profile (SSP) (Fig. 1b).

![Fig.1: (a) General experimental site (white polygon) with NGDC bathymetry (15-s resolution), and for the data examined in this paper: (b) measured sound-speed profiles, and (c) experimental sites for the WHOI echosounder (ES) and NRL VLA/HLA data collection considered in this paper over NGDC bathymetry (3-s resolution).]

The measurements were conducted using two ships. One ship, the FR/V Delaware II, both sampled fish using nets and mapped the fish distributions using high-frequency narrowband commercial echosounders. The other ship, the R/V Endeavor, fielded a pair of towable MF acoustic sonar systems (Figs. 2-3), a downward-looking (short-range, < 0.2 km) sonar used to image fish at high vertical resolution, and a horizontal-looking (relatively long-range; 1-10 km) sonar used to image fish at high horizontal resolution. The short-range system is a commercial echosounder modified to be a broadband four-channel system [8]. In this experiment, this system was typically towed at ~5 km/hr at depths of ~20 m below the sea surface. It has conical beams focused downward toward the seafloor, with frequency-dependent beamwidths of 10-60°. The results in the paper come from the lowest-frequency (1.5-6 kHz) channel with the acoustic signals being 100-ms LFMs with a 2-s repetition period.
Fig. 2: Cartoon of NRL towed sonar geometry (HLA mode) with an image of the source.

The long-range, quasi-monostatic sonar system consisted of a 10-element, vertical source array cut for 3 kHz, and a 32-element linear receiver cut for 5 kHz. This system was deployed at depths of 50-70 m (separations < ~35 m) in both towed (at ~5.5 km/hr) and drifting modes; in former case the receiver became a horizontal line array (HLA) as illustrated in Fig. 2, and in the latter case a vertical line array (VLA). The acoustic signals were 1- or 2-s LFMs with bandwidths that varied from 1 to 8.5 kHz, and repetition periods that varied from 20 to 60 s. These long-range data were beamformed, match filtered and energy normalized (see Annexe). The long-range beam data displayed in this paper represent peak correlator values over non-overlapping 0.0625-s (~45-m) windows.

3. EXPERIMENTAL RESULTS

Fig. 3 shows 27 min (800 pings) of echo amplitudes collected with the downward-looking WHOI echosounder in the early morning of Sept. 11 after the fish had recently migrated to near the bottom. From fish trawls, these fish were estimated to be mostly Atlantic herring (*Clupea harengus*, 0.24-m long; at least 75% by numbers), along with some (probably) silver hake (*Merluccius bilinearis*, 0.22-m long; up to 25% by numbers).

Fig. 3: Echosounder amplitudes vs. depth and distance over ground collected over a 27-min time frame (800 pings), with an image of the WHOI towfish (modified echosounder).
Fig. 4 displays normalized long-range data with the sonar in drift mode in ~180 m of water (Fig. 1c). In this case, the receiver is a VLA. The reader should keep in mind that at the displayed ranges, the beams do not yield much information on the vertical positions of the scatterers within the water column (as the receiver beamwidths scale linearly with range and so quickly span the water column). However the downward-refracting SSPs (Fig. 1b) do suggest that most of the scatterers will be below the source depth of 80 m. (The receiver depth was 86 m.) Also, with increasing range, echoes will arrive at receive angles increasing closer to broadside. The acoustic signals were 1-s LFMs sweeping 2.5-3.5 kHz transmitted every 20 s. The noise floor (for energy-normalized data) for this signal is –30 dB.

Figs. 4a-c show examples of consecutive single-ping returns that illustrate short-time echo variability characteristic of milling fish. Fig. 4d is an average over 10 pings (3.3 min). That the echo characteristics remain similar with increasing range—other than decreasing SNR due to increasing propagation loss with range—is consistent with the monopole scattering nature of the swimbladders of these fish at these frequencies.

Concurrent echosounder measurements on the Delaware II (41.995°N, 68.096°W) noted a couple of fish schools at 180-200 m depth, with densities of ~0.8 per m³.

Fig. 5 displays normalized long-range data with the sonar in tow mode (receiver an HLA at 50 m; source at 78 m) in ~200 m of water (Fig. 1c). The acoustic signals were 2-s LFMs sweeping 6.0-1.5 kHz transmitted every 60 s. The noise floor (for energy-normalized data) for this signal is –34.5 dB. In Figs. 5a-c, examples of single-ping returns 4 min apart illustrate
the spatiotemporal echo variability characteristic of milling fish in shallow water. Fig. 5d is an average over 17 pings (17 min) along the straight track parallel to the northern flank of Georges Bank. (Returns are generally absent in the beams toward forward endfire due to own-ship noise preventing achieving appreciable SNRs.)

Fig. 5: Long-range data with the sonar in tow mode (9 Sept.; night-time—early in the evening) showing HLA beams vs. range for: (a)-(c) 3 pings transmitted 4 min apart and (d) a 17-ping average. (At the earliest ranges shown, the dark-blue ‘echo-free’ regions centred at broadside correspond to periods of bottom-reverberation decay.) Beam 1 is forward endfire (toward the ship), beam 17 is broadside, and beam 33 is aft endfire. (The box in (d) includes beams 28-32.)

Concurrent echosounder measurements on the Delaware II (42.0772°N, 67.8663°W) noted a dispersed layer of fish at 180-200 m depth, with densities of ~0.02 per m³. This lower density is consistent with the lower reverberation levels seen in Fig. 5 than in Fig. 4. Fig. 6 displays spectral representations of the boxed areas in Figs. 3 and 5d comparing the echosounder and the long-range data (with system and propagation effects removed). Although the data were collected on different days, they are generally consistent, indicating a resonance in the 2.5-3.25 kHz range, and, in turn, consistent with previous studies and modeling of the resonant frequency characteristics of the Atlantic herring at these depths in the Gulf of Maine [8-9]. (Exactly how the Atlantic herring swimbladder volume changes as they change depth remains an open question, as discussed in [9].)

Fig. 6b also displays the volume backscattering strength $S_v$ vs. frequency derived from the echosounder data in the boxed region of Fig. 3. The density of fish within this box was estimated (acoustically) to be ~0.5 per m³.
4. CONCLUSIONS AND FUTURE WORK

These initial results help improve the understanding of both fish behavior and the characteristics of fish aggregations as long-range clutter fields. While fish in shallow water present many challenges to sonar systems because of their spatiotemporal variability and strong coupling between scattering and propagation structures, they should also be viewed as both potentially manageable and presenting opportunities (e.g., changing the operating frequency or signal processing in response to in situ fish assessments).

Future work includes examination of more data sets from the 2008 experiment. For all these data, statistical measures of spatiotemporal characteristics (such as spatial/temporal correlation lengths and probability-density functions) will be examined to extract physical insights into the clutter process.

5. ACKNOWLEDGEMENTS

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ANNEXE – ENERGY NORMALIZATION

“Energy normalization” was applied to the nonstationary time-series data for each beam to remove the mean decay. It uses a sliding window, matched to the duration of the transmitted signal, and the normalization obtained with an incoherent sum of total received energy within that window. Energy-normalized correlator output is equivalent to a normalized cross-correlation function, ranging in amplitude from 0 to 1. The matched filter estimate of the total coherent energy $e_{coh}$ in the received signal is given by:

$$e_{coh}(k) = \Delta t \ast \sum_{n=0}^{N-1} r^\dagger(n) \ast s(k+n)$$

for $k = 0, 1, \ldots, M - N$; $\Delta t \ast \sum_{n=0}^{N-1} r(n)^2 = 1$. (1)

In (1) $r$ is the replica of the transmitted signal of sample length $N$, $s$ is the received signal (plus noise) of sample length $M$, $\Delta t$ is the common time sample spacing, and $\ast$ represents complex magnitude. Here it is assumed that the signal replica has unit energy as indicated by the rightmost equation of (1). In this case, the variance of the matched filter output is the same as the input variance for the coherent signal, while the uncorrelated noise component receives a gain inversely proportional to the time-bandwidth product of the signal replica.

The energy-normalized correlator output $c_{en}$ is computed by dividing the total coherent energy $e_{coh}$ by the total received energy (incoherent sum) $e_{incoh}$ at each sample delay:

$$c_{en}(k) = \sqrt{\frac{e_{coh}(k)}{e_{incoh}(k)}} \quad \text{where} \quad e_{incoh}(k) = \Delta t \ast \sum_{n=0}^{N-1} |s(k+n)|^2$$

(2)
THE INFLUENCE OF PROPAGATION FOCUSING ON CLUTTER STATISTICS

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Abstract: Fluctuations in reverberation from scatterers at long range in shallow water (i.e. clutter) can be affected by many mechanisms. Their statistics, in particular the scintillation index, depend not only on the statistics of the scatterers (their physical distribution, shape, orientation, etc.) but on the outward and return propagation paths. This paper investigates the focusing effects due to forward scattering from weak undulations in the seabed. Using a Fresnel-Kirchhoff simulation of bistatic geometry and a one dimensional surface the intensity statistics are calculated including scintillation index, autocorrelation function and probability distribution. The behaviour of these properties is investigated for various Gaussian-correlated roughnesses. Bistatic geometries include source and receiver moving with constant separation, and moving source with fixed receiver. Fairly modest vertical undulations can result in significant fluctuation due to reflection focusing with scintillation indices possibly greater than unity.

Keywords: clutter, forward scatter, focus, statistics, scintillation index
1. INTRODUCTION

The statistics of the fluctuating part of reverberation (clutter) has a strong influence on the development of active sonar target detection algorithms since it constitutes the interfering background. Consequently much experimental and theoretical effort has recently been spent on understanding the link between these statistics and the scatterers themselves [1-4]. In this paper we highlight some long range propagation effects that may complicate this issue. This paper investigates potential propagation fluctuations by numerically simulating forward scattering from a synthetic surface using the Kirchhoff approximation. A more detailed version of this paper [5] also includes evaluation of a formula for scintillation index.

2. SIMULATION

In this simulation the source and receiver take on (1024) discretized positions at depth zero, and the 1 km long rough surface is represented by 8192 points at a mean depth below of 100m. Two bistatic arrangements are chosen: in the first, the source and receiver have fixed horizontal separation as they move; in the second, the receiver is fixed while the source moves. The intensity is calculated with the 2D Kirchhoff formula by integrating over all possible paths from source to surface \( r_s(x_s, x_1) \) and from surface to receiver \( r_r(x_r, x_1) \). The reflection coefficient is assumed to be unity. Without any special normalisation this yields an intensity of exactly the inverse square of the round trip range for a perfectly flat (specular) reflector.

\[
\psi(x_s, x_r, t) = \frac{i \sin \theta}{\lambda r_{os} r_{or}} \sqrt{\frac{\lambda}{r_{os} + r_{or}}} \times \left( \exp \left( ik(r_{os} + r_{or}) \right) \right) \left( - \frac{ct - (r_{os} + r_{or})}{ct_p} \right)^2 dx_1
\]

\[
\sin \theta = \frac{z_{os}}{r_{os}} = \frac{z_{or}}{r_{or}} = \frac{z_{os} + z_{or}}{r_{os} + r_{or}}
\]

The square root term in Eq. (1) results from integrating out the third dimension \((y)\) assuming it to be flat, and the additional “o” subscripts denote values in the absence of roughness. Equation (1) allows for a Gaussian pulse shape with half-width \(t_p\) \((ct_p\) spatially) centred on a travel time \(t\). The 1D surface is generated with a given exact autocorrelation function and vertical standard deviation by filtering a Gaussianly distributed random number sequence.

The direct output of this simulation is acoustic intensity as a function of source and receiver position. From this it is straightforward to compute the variance of the intensity, the scintillation index, the intensity’s autocorrelation function and characteristic width, and the PDF of the intensity (or any function of intensity) via a histogram. Because this simulation uses a one-dimensional surface the variances tend to be somewhat smaller than predicted by the theoretical approach in [5].
2.1. **Geometry 1: moving source and receiver; constant separation**

Maintaining a constant source-receiver separation ensures that if there are focusing effects they are statistically stationary as the source-receiver moves. Here the separation is 200m so that the mean grazing angle is 45°.

2.1.1. **Intensity vs position**

The lower panel of Fig. 1 shows that the Gaussian autocorrelation function results in a rather smooth, “wavy” scattering surface $\zeta(x)$. The upper panel demonstrates the spiky behaviour of the corresponding received intensity as a function of the mean of source and receiver position (so that the intensity peaks can be seen to line up with the concave parts of the surface). This example was deliberately chosen so that the expected curvature produces focusing with this geometry. The vertical standard deviation is $\sigma = 0.3$m, and the horizontal correlation length is $T = 10$m, resulting in a radius of curvature (inverse of expected curvature) of 96m.

![Fig. 1: A simulation of intensity fluctuations plotted against mean position of a source and receiver (upper panel) as they move at constant horizontal separation (200m) and constant height (100m) above the surface shown in the lower panel.](image)

It is worth noting, however, that the obvious spikyness does not necessarily indicate a scintillation index greater than unity. It is easy to generate an exponentially distributed sequence, which has SI = 1 exactly, by squaring and adding a pair of Gaussian random sequences, and this has a similar spiky appearance. Nevertheless, from the point of view of any experiment designed to measure other acoustic quantities (such as reflection coefficients) or variability of other quantities (such as scattering coefficient), any variability is undesirable, i.e. one would hope for SI = 0 in the outward and return paths.
2.1.2. Scintillation Index

Reference [5] demonstrates, using different techniques, that the variance of intensity (or the scintillation index) depends on the height of the source and receiver above a fixed wavy surface. Close up there is typically only one quasi-specular path and SI is small; far away there are many paths and SI → 1; in between, there is a focusing region where SI may be much greater than unity. Here the vertical and horizontal scale can be adjusted so that the radius of curvature remains constant, i.e. $\sigma / T^2 = \text{constant}$, and there is always the same degree of focusing. In this case the constant was chosen to be $0.003 \text{ m}^{-1}$ which provides radii of curvature of order 100m. Scintillation index is plotted against roughness phase ($\phi_o = 2k\sigma \sin \theta$) in Fig. 2. This plot shows (by simulation) that with a one-dimensional surface it is possible to attain high scintillation indices provided $\phi_o$ is greater than about 3. With the assumed wavelength of 0.5m and grazing angle of 45° this corresponds to a roughness of $\sigma = 0.17\text{m}$.

![Fig. 2: Variation of scintillation index with surface roughness $\sigma$ specified as the phase $\phi_o$. In the simulations $\theta = 45^\circ$, acoustic wavelength is 0.5m, and the horizontal roughness scale $T$ is always adjusted so that $\sigma / T^2$ is constant to retain the same degree of focusing.](image)

2.1.3. Intensity Probability Distribution Function (PDF)

By repeating the realizations like the one in Fig. 1 it is possible to plot a histogram which in the limit (with normalisation) converges on the PDF. An example for 100 different seeds for a 8192 point surface is shown in Fig. 3. Plotting as log of probability highlights the differences for PDFs near to exponential. For comparison, the straight lines are the exponential PDFs $\exp\{-I / (2\sigma^2)\} / (2\sigma^2)$ for $\sigma^2 = 0.4$ to 0.8 in 0.1 steps. The vertical standard deviation is $\sigma = 1.0\text{m}$ which results in an SI of 1.58 and a strong tail.

2.1.4. Correlation length

Figure 4 shows the initially given Gaussian autocorrelation function superimposed on the normalised autocorrelation averaged over 100 realizations of the surface and the corresponding normalised autocorrelation function of the received intensity for the same case as Fig. 1. The rapid decorrelation of the intensity compared with the surface height is striking.
Fig. 3: PDF of intensity derived from a normalized histogram of 100 simulations with a 8192-point rough surface with parameters $\sigma = 1.0\text{m}$, $T = 18.2\text{m}$ resulting in $SI = 1.58$. The straight lines are the exponential distributions, for reference, $\exp\{−1/(2\sigma_I^2)\} / (2\sigma_I^2)$ with $\sigma_I^2$ in 0.1 steps between 0.4 to 0.8.

Fig. 4: Normalized roughness and intensity correlation functions for 100 realizations of a 8192-point surface with $\sigma = 0.3\text{m}$, $T = 10\text{m}$. The originally specified Gaussian correlation function is superimposed.

2.2. Geometry 2: fixed receiver; moving source

The fixed end point geometry is of less interest from the point of view of determining expected values and PDFs because the angles change and the ratio of range to radius of curvature changes as the source-receiver separation changes. It is included here because it is a convenient geometry for doing at-sea experiments, and is used, for example, as the “move-out” technique to measure reflection loss [6,7]. Obviously fluctuations due to focusing need to be minimized or eliminated somehow when measuring any property of the seabed (other than curvature). It seems extremely unlikely that one could ever find a location where the bottom was absolutely flat – the question is, how flat does it have to be to ensure fluctuations
less than a given amount? First, a simple order of magnitude calculation for a monostatic and a bistatic sonar. Obviously a monostatic sonar at height $z$ above the surface requires curvature equal to $1/z$ for perfect focus. If, on the other hand, the source and receiver are widely separated by a distance $L$ (still at height $z$) then the surface needs to match an ellipse with source and receiver at the ellipse foci. It is easy to show that the curvature at the deepest point on the ellipse (in terms of the semi-major and semi-minor axes $a$, $b$) is $R = a^2 / b = [(L/2)^2 + z^2] / z = z / \sin^2 \theta$. Since the order of magnitude of the radius of curvature in terms of the vertical and horizontal scales of the roughness is $R = T^2 / \sigma$ one finds that very small roughness heights can result in significant focusing. For instance, if $z = 100\text{m}$ and $L = 200\text{m}$ (grazing angle $= 45^\circ$), then $R = 200\text{m}$. If the horizontal scale $T$ is $3\text{m}$ then the required vertical deviation is only $0.045\text{m}$. Extending $L$ to $1\text{km}$ (grazing angle $= 11.3^\circ$) one finds $R = 2.6\text{km}$, leading to a vertical deviation of $3.5\text{mm}$, still with a $3\text{m}$ horizontal scale. Even with a $10\text{m}$ horizontal scale one only needs $38\text{mm}$. This calculation alone shows that it is almost impossible to avoid these effects at some range or other.

![Fig. 5: Simulated intensity fluctuations plotted against source position (100m height) with fixed receiver (20m height) with roughness horizontal scale $T = 10\text{m}$ and vertical scale $0.1\text{m}$.

2.2.1. Typical intensity vs. position

Using the same $1\text{km}$ long surface realisation but varying its standard deviation, $\sigma$, the receiver is fixed at $20\text{m}$ above the seabed and the source moves between $0$ and $900\text{m}$ horizontally at height $100\text{m}$ above the seabed. Figure 5 shows intensity plotted against source-receiver separation for roughness height $\sigma = 0.1\text{m}$ and horizontal correlation length $T = 10\text{m}$. Although the scintillation index is small ($0.21$) the fluctuations are still significant. Notice that the fluctuations appear slower than in Fig. 1. This is simply because the reflection point is at only a fraction of the source range, its maximum being $150\text{m}$ rather than $900\text{m}$. Because the focal length and the slant-ranges from source and receiver to the specular point vary in different ways the fluctuations are not uniform with move-out. In these examples fluctuations are slightly more rapid and larger amplitude at long range. Incidentally, at long ranges and small grazing angles shadowing may be more pronounced, depending on the roughness. In these examples the effects, i.e. inclusion or exclusion of geometrically shadowed points, is within a line thickness.

2.2.2. Typical intensity vs. angle
If the same intensities as in Fig. 5 are now plotted against the assumed specular angle $\theta$, as in Fig. 6, using

$$\tan \theta = (z_s + z_r)/ L$$

(3)

then the already slight bunching up of features at long range translates to more extreme bunching at low angles because of the arccotangent operation. The higher depth of modulation at low angles is also more apparent in these plots.

Whilst acknowledging that there are many other mechanisms at play when investigating layered bottom reflection properties with the move-out technique (see e.g. [6]) the similarity of this curve to many of the experimental curves at low angles, say $10^\circ$-$30^\circ$ (see, for example, Figs. 15, 16, *ibid.*) could be interpreted as further strong evidence for the importance of the reflection focusing phenomenon.

Fig. 6: The same simulated intensity fluctuations as in Fig. 5 plotted against calculated move-out angle with roughness horizontal scale $T = 10m$ and vertical scale $0.1m$.

3. CONCLUSIONS

The fluctuating part of long range reverberation, known as clutter, depends partly on the scattering objects and scattering areas, and partly on the outward and return propagation paths which may pass through inhomogeneous media or be subject to forward scatter and focusing effects at the boundaries. In this context, this paper has concentrated on focusing from a single boundary and the statistics of forward scatter.

The approach to investigating focusing statistics was to build a one-dimensional surface simulation in which one could control the illuminating pulse and its chosen arrival time. Given realizations of a surface with specified Gaussian correlation function it was possible to calculate the fluctuating intensity and various statistics including SI, autocorrelation function, and probability distribution. Keeping the expected curvature constant and moving source and receiver at constant separation it was shown that SI reaches unity with a roughness phase of 2 or 3 and finally settles into a more or less linear increase with roughness phase (Fig. 2). Probability distributions with clear upward trending tails (Fig. 3) and $SI > 1$ were derived from histograms for 100 realizations with roughnesses greater than $0.3m$. The autocorrelation function of the intensity was shown to be very narrow compared with that of the seabed (Fig. 4). In order to counter the objection that the Gaussian correlation function makes the surface
unrealistically smooth, an uncorrelated roughness was superimposed in [5] to investigate the
degradation. Whether the original smoothly undulating surface has a high or low roughness
phase (and corresponding high or low SI) the effect of adding uncorrelated roughness was to
pull the SI towards unity. Adding a roughness of 0.06m (roughness phase of about one)
produced a significant difference in SI although focusing was still strong.
An alternative bistatic geometry with moving source and fixed receiver was also
investigated because it is a commonly used experimental arrangement. Because the source
may move through foci at intermediate ranges but yet be well away from foci at long or short
ranges, plots of intensity fluctuations vs. range are distinctly non-stationary. At short ranges
fluctuations are weak so the SI is low; at long ranges the SI tends to unity (Fig. 5). When the
same intensities are plotted against calculated grazing angle there is a very strong tendency
for the fluctuations to bunch up at low angles (Fig. 6). In addition the change of SI from long
to short range shows as a stronger fluctuation amplitude at low angles.

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A PHASE SPACE VIEW OF DISPERSIVE PROPAGATION, MOMENT VARIABILITY, AND ENVIRONMENTALLY-INVARIENT FEATURES FOR CLASSIFICATION

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Abstract: Propagation of sound in shallow water is impacted by interactions with the ocean surface and bottom, which give rise to frequency-dependent spreading and attenuation (dispersion and damping). In active sonar for automatic target recognition, these propagation-induced effects can be detrimental to classification because the observed backscatter depends not only on the target but also on the propagation environment and how far the wave has traveled, resulting in increased variability in the received sonar signals. We present an approach, based on a recently developed phase space approximation of dispersive propagation, for analyzing these effects on temporal moment features of the propagating signal, as well as the effect of random variability in certain channel parameters, in particular target distance. We also note new features for classification that are invariant to the propagation effects of dispersion and damping.

Keywords: dispersive propagation, Wigner distribution, moments, target recognition
1 INTRODUCTION

When a sound propagates in shallow water, interactions with the ocean surface and bottom can induce propagation effects, characterized by the structural (or geometric) dispersion of the channel. These effects are frequency-dependent, such that different frequencies in the wave propagate at different velocities and are attenuated at different rates. As such, the wave changes as it propagates. When using active sonar for automatic target recognition, these propagation-induced effects can be detrimental to classification because the observed backscatter depends not only on the target but also on the propagation environment and how far the wave has traveled, resulting in increased variability in the received sonar signals. We examine the effects of dispersion and damping on various moments of the wave, since moments are often used as features for classification. Because of the frequency-dependent propagation effects that cause the wave to change over time and distance, we use a joint time-frequency approach to formulate dispersive propagation in phase space. This approach allows for a simple but powerful approximation, which we utilize to examine the effects of uncertainty in target distance in a dispersive channel with damping. We also discuss how this approach suggests new moment-like features for classification that are invariant to the propagation effects of dispersion and damping.

2 LINEAR WAVE PROPAGATION AND PHASE SPACE CHANNEL MODEL

In linear wave propagation, the wave at position \( x \) and time \( t \) is given by

\[
   u(x, t) = \frac{1}{\sqrt{2\pi}} \int F(0, \omega) e^{j(\kappa(\omega)x - \omega t)} \, d\omega
\]

for each mode, where \( F(0, \omega) \) is the spectrum of the initial wave \( u(0, t) \),

\[
   F(0, \omega) = \frac{1}{\sqrt{2\pi}} \int u(0, t) e^{j\omega t} \, dt
\]

and \( \kappa(\omega) \) is the dispersion relation, which couples spatial frequency \( k \) and radial frequency \( \omega \). In many propagation environments, \( \kappa(\omega) \) is a nonlinear function of \( \omega \), and further can be complex, \( \kappa(\omega) = \kappa_R(\omega) + j\kappa_I(\omega) \), which is the case when there is damping (frequency-dependent attenuation, characterized by non-constant \( \kappa_I(\omega) \)). Eq. (1) can be expressed equivalently as

\[
   u(x, t) = h(x, t) \ast u(0, t)
\]

where \( h(x, t) = \int e^{j(\kappa(\omega)x - \omega t)} \, d\omega \) is the channel impulse response. The frequency-dependent propagation effects, characterized by \( \kappa(\omega) \), cause the wave to change as it propagates. In addition, there can be inherent dispersion in the acoustic scattering from elastic objects [2, 3]. Because of these phenomena, phase space methods, such as time-frequency distributions, have been applied to study wave propagation and acoustic scattering [2, 3, 4, 5, 6, 7]. The Wigner time-frequency distribution is defined as

\[
   W_u(t, \omega; x) = \frac{1}{2\pi} \int u\left(x, t + \frac{\tau}{2}\right) u^*\left(x, t - \frac{\tau}{2}\right) e^{j\omega \tau} \, d\tau
\]

\footnote{For consistency with the sign conventions of the Fourier transforms in \( x \) and \( t \) as defined previously, we take the sign of the exponent here to be positive.}
Substituting in Eq. (3), one obtains [8]

\[ W_u(t, \omega; x) = W_u(t, \omega; 0) *_t W_h(t, \omega; x) \]  

(5)

where \( W_u(t, \omega; 0) \) is the Wigner distribution of the initial wave, \( u(0, t) \), and \( W_h(t, \omega; x) \) is the Wigner distribution of the channel impulse response. This result is exact; an accurate approximation is obtained by approximating the channel Wigner distribution as [5, 9]

\[ W_h(t, \omega; x) \approx e^{-2\kappa I(\omega) x} \delta(t - \kappa' R(\omega) x) \]  

(6)

To introduce random variations, we let the channel be defined by realizations of random variables. For example, in Eq. (1), the parameter \( x \) could be taken to be a random variable, representing uncertainty in the target distance or how far the wave propagates. If the initial wave \( u(0, t) \) is deterministic, then Eq. (5) becomes, in the ensemble sense,

\[ \langle W_u(t, \omega; x) \rangle = W_u(t, \omega; 0) *_t \langle W_h(t, \omega; x) \rangle \]  

(7)

where the brackets \( \langle . \rangle \) denote the ensemble average. We use this propagation model to calculate the statistical behavior of the moment features of a propagating wave.

3 TEMPORAL MOMENTS AND VARIABILITY

Moments have been used as features for target classification. Ideally, the sonar echo from a particular target of interest would produce the same feature value each time the target was insonified. However, in practice this is not the case, as noise and other environmental factors cause the features to take on random values. In this paper, we ignore the effect of noise and instead focus on variability introduced by uncertainty in the environment. We also focus on temporal moments, defined as

\[ \langle t^n_x \rangle = \int t^n |u(x, t)|^2 dt = \int \int t^n W_u(t, \omega; x) d\omega dt \]  

(8)

where the last equality holds because the Wigner distribution satisfies the marginals [8].

3.1 Moments of Moments of the Wigner Distribution

When there is uncertainty in the channel, such as in the target distance, the moments will exhibit variability. We examine this variability by calculating the “moments of the moments,” meaning the expectation values of the moments, following an approach similar to Liu and Yeh [10]. However, we do so from the Wigner distribution and approximation, and we do not consider turbulent or random media, but rather a channel model characterized by random parameters.

The expected value of the temporal moments of the signal at \( x \) is given by

\[ \langle \langle t^n_x \rangle \rangle = \int \int \int \int \ldots \int \langle t^n \rangle \langle W_u(t, \omega; x) \rangle d\omega dt \]  

(9)

Higher-order expected values of the temporal moment features are given by

\[ \langle \langle \langle t^n_x \rangle \rangle \rangle = \int \int \int \int \ldots \int \langle t^n \rangle \langle W_u(t_1, \omega_1; x) W_u(t_2, \omega_2; x) \ldots W_u(t_i, \omega_i; x) \rangle dt_1 d\omega_1 dt_2 d\omega_2 \ldots dt_i d\omega_i \]  

(10)
We may re-write Eq. (10) in terms of the temporal moments of the initial wave (at $x = 0$) and the expected values of the temporal moments of the channel impulse response, as

$$
\langle \langle (t_u^m) \rangle \rangle = \sum_{m_1,m_2,...,m_i=0}^{n} \binom{n}{m_1} \binom{n}{m_2} \cdots \binom{n}{m_i} \int \cdots \int \left( \langle t_{u}^{n-m_1} \rangle_{0,\omega_1} \langle t_{u}^{n-m_2} \rangle_{0,\omega_2} \cdots \langle t_{u}^{n-m_i} \rangle_{0,\omega_i} \right) \times \\
\langle \langle (t_h^m) \rangle \rangle_{x,\omega_1} \langle (t_h^m) \rangle_{x,\omega_2} \cdots \langle (t_h^m) \rangle_{x,\omega_i} \rangle \ d\omega_1 d\omega_2 \cdots d\omega_i
$$

(11)

The random quantities in Eq. (11) are the local temporal moments of the channel, given by $\langle t_h^m \rangle_{x,\omega} = \int t^k W_h(t,\omega,x) \ dt$.

The problem of solving for the statistics of the temporal moment features of the signal, then, essentially reduces to finding the ensembles of products of the moment features of the channel. The Wigner approximation of the channel, Eq. (6), may be used to obtain approximate values of the temporal moment features of the channel:

$$
\langle \langle (t_h^m) \rangle \rangle_{x,\omega} \approx \int t^m e^{-2\kappa_1(\omega)x} \delta(t - \kappa'(\omega)x) \ dt = e^{-2\kappa_1(\omega)x} \left( \kappa'(\omega)x \right)^m
$$

(12)

Ensemble averages of products of Eq. (12) may be used to obtain approximate moments of the temporal moment features. For example, if $x$ is described by probability distribution $P(x)$, then the expected value of the product of $i$ moments is approximately given by

$$
\langle \langle (t_h^m) \rangle \rangle_{x,\omega} \approx \int e^{-2x(\kappa_1(\omega_1) + \kappa_1(\omega_2) + \cdots + \kappa_1(\omega_i))} \left( \kappa'(\omega_1) \right)^{m_1} \left( \kappa'(\omega_2) \right)^{m_2} \cdots \left( \kappa'(\omega_i) \right)^{m_i} P(x) dx
$$

(13)

4 EXAMPLE

We derive approximate expressions for the first two moments of the moment features in a channel where the range $x$ is uncertain and modelled by a Gaussian probability density function with mean $\mu$ and variance $\sigma^2$. The expected value of each of the temporal moments is given exactly by Eq. (11) with $i = 1$. Using Eq. (13), we have an approximate expression for the temporal moments of the channel,

$$
\langle \langle (t_h^m) \rangle \rangle_{x,\omega} \approx \left( \kappa'(\omega) \right)^m d_1(\omega) \left[ \frac{1}{\sigma \sqrt{2\pi}} \int x^m e^{-\frac{(x-\left(\mu - 2\kappa_1(\omega)\sigma^2\right))^2}{2\sigma^2}} dx \right]
$$

(14)

where $d_1(\omega) = e^{2\kappa_1(\omega)\sigma^2 - 2\kappa_1(\omega)\mu}$. For $m = 0, 1$, we have

$$
\langle \langle (t_h^0) \rangle \rangle_{x,\omega} \approx d_1(\omega) ; \ \langle \langle (t_h^1) \rangle \rangle \approx \left( \kappa'(\omega) \right) d_1(\omega) \left[ \mu - 2\kappa_1(\omega)\sigma^2 \right]
$$

(15)

4.1 First Temporal Moment Feature

The general expression for the expected value of the first temporal moment feature of the signal is given by Eq. (11) with $i = 1$ and $n = 1$:

$$
\langle \langle (t_u^1) \rangle \rangle = \sum_{m_1=0}^{1} \binom{1}{m_1} \int \langle t_{u}^{n-m_1} \rangle_{0,\omega_1} \langle (t_h^m) \rangle_{x,\omega_1} d\omega_1
$$

(16)

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Utilizing the approximate expressions in Eq. (15), we have

$$\langle (t_u)_x \rangle \approx \int (t_u)_{0,\omega_1} d_1(\omega_1) d\omega_1 + \int (\kappa'_R(\omega_1)) d_1(\omega_1) \left[ \mu - 2\kappa_I(\omega_1)\sigma^2 \right] |F(0,\omega_1)|^2 d\omega_1$$  \hspace{1cm} (17)

For the second moment of the first temporal moment feature, we use the approximation

$$\langle (t^{m_1}_h)_{x,\omega_1} (t^{m_2}_h)_{x,\omega_2} \rangle \approx \int e^{-2x(\kappa_I(\omega_1) + \kappa_I(\omega_2))} (\kappa'_R(\omega_1)x)^{m_1} (\kappa'_R(\omega_2)x)^{m_2} P(x) dx$$  \hspace{1cm} (18)

$$\approx \left( \kappa'_R(\omega_1) \right)^{m_1} \left( \kappa'_R(\omega_2) \right)^{m_2} d_2(\omega_1, \omega_2) \times \frac{1}{\sigma \sqrt{2\pi}} \int x^{m_1+m_2} e^{-\frac{(x-\mu-2\kappa_I(\omega_1))\sigma^2}{2\sigma^2}} dx$$  \hspace{1cm} (19)

where \( d_2(\omega_1, \omega_2) = e^{2\kappa_I^2(\omega_1) \sigma^2 + 4\kappa_I(\omega_1)\kappa_I(\omega_2)\sigma^2 + 2\kappa_I^2(\omega_2)\sigma^2 - 2\kappa_I(\omega_1)\mu - 2\kappa_I(\omega_2)\mu} \). Therefore, we have

$$\langle (t^0_h)_{x,\omega_1} (t^0_h)_{x,\omega_2} \rangle \approx d_2(\omega_1, \omega_2)$$  \hspace{1cm} (20)

$$\langle (t^1_h)_{x,\omega_1} (t^1_h)_{x,\omega_2} \rangle \approx \kappa'_R(\omega_1)d_2(\omega_1, \omega_2) \left( \mu - 2\sigma^2 (\kappa_I(\omega_1) + \kappa_I(\omega_2)) \right)$$  \hspace{1cm} (21)

$$\langle (t^1_h)_{x,\omega_1} (t^1_h)_{x,\omega_2} \rangle \approx \kappa'_R(\omega_1)\kappa'_R(\omega_2)d_2(\omega_1, \omega_2) \left[ \left( \mu - 2\sigma^2 (\kappa_I(\omega_1) + \kappa_I(\omega_2)) \right)^2 + \sigma^2 \right]$$  \hspace{1cm} (22)

The second-order expectation of the first-order temporal moment feature is given by Eq. (11) with \( n = 1 \) and \( i = 2 \), which may be approximated by plugging in the expressions from Eqs. (20)-(22):

$$\langle ((t_u)_x)^2 \rangle \approx \int \int (t_u)_{0,\omega_1} (t_u)_{0,\omega_2} d_2(\omega_1, \omega_2) d\omega_1 d\omega_2 \hspace{1cm} (23)$$

4.2 Second Temporal Moment Feature

The expected value of the second temporal moment feature is given by Eq. (11) with \( n = 2 \) and \( i = 1 \):

$$\langle (t^2_u)_{x} \rangle = \sum_{m_1=0}^{2} \binom{2}{m_1} \int \langle t^{n-m_1}_u \rangle_{0,\omega_1} \langle t^{m_1}_h \rangle_{x,\omega_1} d\omega_1$$  \hspace{1cm} (24)

To find an approximate expression for this moment, we require the approximate temporal moments of the channel to second order. The zeroth and first order approximate moments are given in Eq. (15), while the second order moment is approximately

$$\langle (t^2_h)_{x,\omega_1} \rangle \approx \left( \kappa'_R(\omega) \right)^2 d_1(\omega) \left[ \left( \mu - 2\kappa_I(\omega)\sigma^2 \right)^2 + \sigma^2 \right]$$  \hspace{1cm} (25)

Plugging these values into Eq. (24), we obtain the general expression

$$\langle (t^2_u)_x \rangle \approx \int (t^2_u)_{0,\omega_1} d_1(\omega_1) d\omega_1 + 2 \int (t_u)_{0,\omega_1} (\kappa'_R(\omega_1)) d_1(\omega_1) \left[ \mu - 2\kappa_I(\omega_1)\sigma^2 \right] d\omega_1 \hspace{1cm} (26)$$
The second-order expectation of the second temporal moment feature is given by Eq. (11) with \( n = 2 \) and \( i = 2 \):

\[
\langle ((t^2_x)_x)^2 \rangle = \sum_{m_1,m_2=0}^{n} \left( \frac{2}{m_1} \right) \left( \frac{2}{m_2} \right) \int \int \langle t_n^{m_1} \rangle_{0,\omega_1} \langle t_n^{m_2} \rangle_{0,\omega_2} \langle \langle t^{m_1}_x \rangle_{x,\omega_1} \langle t^{m_2}_x \rangle_{x,\omega_2} \rangle d\omega_1 d\omega_2
\]  

(27)

The above exact expression may be approximated by using approximations to \( \langle \langle t^{m_1}_x \rangle_{x,\omega_1} \langle t^{m_2}_x \rangle_{x,\omega_2} \rangle \) for the various values of \( m_1 \) and \( m_2 \). In addition to the expressions in Eqs. (20)-(22), the necessary quantities are:

\[
\langle \langle t^0_{h,\omega_1} t^2_{h,\omega_2} \rangle \rangle \approx (\kappa'_{R}(\omega_2))^2 d_2(\omega_1,\omega_2) \left[ (\mu - 2\sigma^2 (\kappa_{I}(\omega_1) + \kappa_{I}(\omega_2)))^2 + \sigma^2 \right]
\]

(28)

\[
\langle \langle t^1_{h,\omega_1} t^2_{h,\omega_2} \rangle \rangle \approx \kappa'_{R}(\omega_1) (\kappa'_{R}(\omega_2))^2 d_2(\omega_1,\omega_2) \left[ (\mu - 2\sigma^2 (\kappa_{I}(\omega_1) + \kappa_{I}(\omega_2)))^3 + 3\sigma^2 (\mu - 2\sigma^2 (\kappa_{I}(\omega_1) + \kappa_{I}(\omega_2))) \right]
\]

(29)

\[
\langle \langle t^2_{h,\omega_1} t^2_{h,\omega_2} \rangle \rangle \approx (\kappa'_{R}(\omega_1))^2 (\kappa'_{R}(\omega_2))^2 d_2(\omega_1,\omega_2) \left[ (\mu - 2\sigma^2 (\kappa_{I}(\omega_1) + \kappa_{I}(\omega_2)))^4 + 6\sigma^2 (\mu - 2\sigma^2 (\kappa_{I}(\omega_1) + \kappa_{I}(\omega_2)))^2 + 3\sigma^4 \right]
\]

(30)

For the general case of an arbitrary complex dispersion relation, most of these integrals cannot be easily evaluated. However, when there is no damping, the dispersion relation is purely real, and the calculations are simpler, as considered next.

4.3 Real Dispersion Relation

For the special case of a purely real dispersion relation (no damping), we have

\[
\kappa_{I}(\omega) = 0 \quad d_1(\omega) = 1 \quad d_2(\omega_1,\omega_2) = 1
\]

(31)

by which the approximate expected value of the first temporal moment feature becomes

\[
\langle \langle t_u \rangle \rangle_x \approx \int \langle t_u \rangle_{0,\omega_1} d\omega_1 + \int \kappa'_{R}(\omega_1) |F(0,\omega_1)|^2 d\omega_1 = \langle t_u \rangle_0 + \mu \langle \kappa'_{R}(\omega) \rangle_0
\]

(32)

and the variance of the first temporal moment feature is

\[
\sigma^2_{(t_u)_x} \approx \langle \langle (t_u)_x^2 \rangle \rangle - \langle \langle t_u \rangle \rangle_x^2 = \sigma^2 (\langle \kappa'_{R}(\omega) \rangle_0)^2
\]

(33)

For the second-order temporal moment feature, the approximate expected value is

\[
\langle \langle t^2_{u,\omega_1} \rangle \rangle \approx \langle t^2_{u,\omega_1} \rangle_0 + 2\mu \langle t \kappa'_{R}(\omega) \rangle_0 + (\mu^2 + \sigma^2) \langle (\kappa'_{R}(\omega))^2 \rangle_0
\]

(34)

and the variance of the second temporal moment is

\[
\sigma^2_{(t^2_{u,\omega_1})_x} \approx \langle \langle (t^2_{u,\omega_1})^2 \rangle \rangle - \langle \langle t^2_{u,\omega_1} \rangle \rangle_x^2 \approx \sigma^2 (\langle t \kappa'_{R}(\omega) \rangle_0)^2 + 8\mu \sigma^2 (\langle t \kappa'_{R}(\omega) \rangle_0 \langle (\kappa'_{R}(\omega))^2 \rangle_0)
\]

\[
+ (4\mu^2 \sigma^2 + 2\sigma^4) \langle \langle \kappa'_{R}(\omega) \rangle^2 \rangle_0
\]

(35)
5 CONCLUSION

Shallow water propagation effects such as dispersion and damping may degrade sonar classification performance by increasing the variability of features extracted from a received waveform. Indeed, the results given in this paper show that in general the temporal moments change as the wave propagates when there is dispersion and damping. Moreover, the variability in these moment features increases when there is uncertainty in certain channel parameters, such as propagation (or target) distance, as analyzed in detail in this paper.

In previous work, we have developed signal processing algorithms for extracting moment-like classification features that are invariant to the effects of dispersion and certain forms of damping \[11\], and we have shown that these propagation-invariant features can improve classification performance, compared to ordinary moments. Analysis of the effects of channel variability on these invariant features is ongoing.

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References


CLUTTER ON THE MALTA PLATEAU

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Abstract: Acoustic and environmental measurements along a particular track on the Malta Plateau, Mediterranean Sea, are of interest for presentation in this paper because high scattering strength and reverberation deviation is persistently observed from a confined location on the Plateau. The diffuse reverberation and clutter data were acquired by a towed source and towed cardioid array with the source transmitting 1-s linear frequency modulated sweeps in the band from 800-1800 Hz every minute along the track. The received signals were acquired for 55 s to cover a large area. Environmental surveys (multibeam echosounder, sub-bottom profiling, side scan sonar, wide-angle reflection measurements) were conducted at this strong scattering location with no clear indication of significant scattering features, i.e. no evidence of bathymetric features, variations in sub-bottom stratification or changes in bottom geoacoustic properties. However, the sub-bottom profiling shows inclusions in the upper 5 m of the sediment which have not been detected by any of the other survey techniques. It is hypothesized that these inclusions have slightly different geoacoustic properties than the surroundings which results in strong clutter-like backscattering. This hypothesis is presented and verified by backscattered time-series computations using a two-way prediction model applied in an environment similar to that of the acquired acoustic data.

Keywords: Clutter, reverberation, measurement, modelling
1. INTRODUCTION

Regions containing target-like returns, i.e. clutter, present an important problem in mid-frequency active sonar activities. In order to further understand the physical properties of natural and man-made clutter features close to the seabed, NURC has conducted a series of experiments on the Malta Plateau, Mediterranean Sea, with the focus on collecting environmental and acoustic clutter data. Strong correlation between the environmental and processed acoustic data has been established which provides important information to enhance prediction capabilities, signal processing algorithms and to eventually reduce the clutter returns in the sonar processing chain. A particular experimental track is of interest for presentation in this paper because high scattering strength and reverberation deviation is persistently observed from a confined location on the Plateau. No strong evidence of scattering features has been identified by environmental surveys at this location. Numerical modelling of the sound propagation and back scattering may give a clue on the clutter sources by assuming a certain underwater environment derived from measurements. In this paper a two-way wave-based prediction modelled is applied to such an environment for time-series calculations, and thereby to propose additional natural clutter sources besides the already known clutter sources on the Malta Plateau.

2. ACOUSTIC AND ENVIRONMENTAL MEASUREMENTS

In 2004 NATO Undersea Research Centre conducted the BASE’04 sea trial on the Malta Plateau, Mediterranean Sea, as a part of a demonstration of an Environmentally Adaptive Sonar Concept. One of the tracks followed during this experiment was designed as a hexagon centred at 36.34°N 14.25°E with straight legs of around 8.5 km. The NRV Alliance deployed a sound source and a Cardioid array at depths 65 m and 70 m, respectively, below the thermocline to enhance bottom interaction of the acoustic signals. The transmitted signals were 1-s Linear-Frequency-Modulated (LFM) sweeps in the band 800-1800 Hz shaded by a 5% Tukey window. The acoustic signals were acquired on the 258-element Cardioid array at a sampling frequency of 12.8 kHz for 55 s to capture diffuse and clutter-like reverberation over a large area. The raw reverberation data were Cardioid beamformed into 120 beams equally spaced in cosine to the steering angle, complex band shifted and matched-filtered before stored to disk for post-processing analysis.

A post-processing algorithm was developed and is described in details in [1]. The reverberation intensity envelope time series were divided into constant corrected range intervals (or equivalent time intervals) for each of the 120 beams and the levels corrected for the beam width. The reverberation intensity envelope for each range interval and a 360° bearing was then normalized by the minimum of the maximum intensity level found in the entire range-bearing annulus. The ratio of the standard deviation to mean (SMR) value of the normalized reverberation intensity was calculated for each range-bearing interval and geo-referenced. This procedure was applied for each acoustic transmission along the hexagon track and then averaged over all transmissions. The result is a scatterer map which indicates regions with clutter-like (spiky) returns and diffuse background reverberation, see [1] and Fig. 1. Higher values in Fig. 1 indicate clutter and lower values diffuse reverberation or noise.
Fig. 1: Scatterer map on the Malta Plateau based on the standard deviation to mean reverberation intensity ratio. A: Campo Vega oil rig, B: Tender ship, C-F: ship wrecks, G: Ragusa Ridge, H: Gozo Island and I: Possible rock outcrops.

The values in the scatterer map are only used as an indication of the clutter-diffuse reverberation distribution on the Malta Plateau. The reason is that at present it is unknown what contribution of imperfect geo-referencing has on the values of the scatterer map. However, the scatterer map still provides information about classes of reverberation sources, i.e. clutter-like or diffuse background reverberation. The scatterer map in Fig. 1 shows anthropogenic features like the Campo Vega oil rig (A), tender ship (B) and wrecks (C-F), and natural features, for instance, Ragusa Ridge (G), the Island Gozo (H) and possible rock outcrops (I). The wrecks and their positions have been verified by side-scan sonar images. There is one particular region on the Plateau of interest which is believed to be either natural bottom/sub-bottom features or of biological origin (J). Note that the SMR is comparable to what is observed from Ragusa Ridge. No strong evidence of scattering sources around region J has been identified at present. Fish has been considered as a possible scattering source, but the extent of the scattering area would mean a huge fish shoal, and the fact that the scattering from region J only appears at a confined area on the entire Plateau makes this hypothesis less likely. Seepage of oil or gas from the bottom has also been considered as a possible scattering source, but Synthetic Aperture Radar images do not indicate oil seepage as a plausible source, and no extensive effort has been made till now measuring gas seepage.

Significant amount of “ground truth” measurements have been performed during the BABO’06, CLUTTER’07, BASE’07 and BASE’07 Extended sea trials involving NRV Alliance, RV Planet [led by Dr Arne Schulz, Forschungsbereich Wasserschall und Geophysik (FWG), Germany], RV Beautemps-Beaupré [led by Dr Yann Stephan, Service Hydrographique et Océanographique de la Marine (SHOM), France], and RV Oceanus [led by Dr John Preston (Pennsylvania State University), Woods Hole Oceanographic Institution (WHOI), USA] with the objective to find evidence of strong bottom scattering features around region J (Fig. 1). In summary, the multibeam bathymetric surveys show that the bathymetry is smooth and the side-scan images indicated only sparse pockmark-like features in the region of interest. Though, the sub-bottom profiling shows inclusion-like features in the upper 5-m sediment layer of the bottom. These inclusions are hypothesized to have
different geoacoustic properties than the surrounding bottom and will, therefore, change the reflection coefficient and scattering properties locally. During the BABO'06 trial RV Planet found higher concentrations of the inclusion in the region J than anywhere else during the survey on the Plateau. RV Oceanus and NRV Alliance performed a more detailed sub-bottom profiling during the BASE'07 Extended and CLUTTER'07 trials to map the inclusions. The result of the lawn-mow pattern survey by RV Oceanus is shown in Fig. 2(a), and the inclusions are clearly visible in the 3 northern most profiles and the presence of these features were verified by the sub-bottom profile tracks performed by NRV Alliance. Two time traces from the NRV Alliance profiling track is shown in Fig. 2(b). Ping No. 1 is acquired in the turbid area outside the inclusion [Fig. 2(a)] and Ping No. 2 is a time trace going through the inclusion. There are clearly stronger reflections from inside the 5-m sediment layer in the turbid area (Ping No. 1) than at the inclusion (Ping No. 2), which most likely is caused by an acoustically harder sediment layer in the turbid regions than at the inclusion.

Fig.2: Sub-bottom profiles (a) acquired on RV Oceanus during BASE’07 Extended and (b) single profiles acquired on NRV Alliance during CLUTTER’07 outside (Ping No. 1) and at (Ping No. 2) the observed sediment inclusions.

3. MODELLING APPROACH

Clutter-like returns from varying bottom reflection properties are investigated in this section by applying a modified version of the two-way Parabolic Equation (PE) model developed by Collins and Evans [2]. The significant modifications consist of implementation of the PE self-starter [3], an extension from only handling discrete bathymetric scattering features to include continuous scattering from sea surface, bathymetry and sub-bottom, implementation of the complex Padé coefficients presented in [4] to obtain numerical stability and to accurately support steep-angle scattering.

Calculation of the forward and backward propagating field in range-dependent environments is performed by dividing the environment into range-independent sectors. The boundary conditions, i.e. continuity of pressure and particle velocity, are fulfilled at the interface between each sector. The backward propagating field is calculated by adding the backscattered field at each sector interface as the field is marched from maximum range towards the source location at zero range. The time series for both the forward and backward field are determined by Fourier synthesis. PE modelling results have been compared to
solutions from the coupled-mode model COUPLE [5] with good agreement and the performance is similar to the PE model PERM [6].

A synthetic environment has been defined to mimic the experimental conditions during the BASE’04 trial including measured water column sound speed and inclusions observed in Fig. 2(a) in the bottom (Fig. 3). Only the most Eastern inclusion apparently extending across the 3 Northern profiles in Fig. 2(a) is modelled. The horizontal shape of the inclusion is assumed elliptic with major and minor axis of 1389 and 463 m, respectively, and 5 m thick. The bottom properties are shown in Fig. 3 with a across section of the inclusion (white area in the bottom), and consists of a 5-m sediment layer over a halfspace. The sediment geoacoustic properties outside the inclusion are derived from [7], and the inclusion is assumed to have slightly lower values of the geoacoustic properties.

![Fig. 3: Modelled back propagating time series (upper part) in an environment with sediment inclusions downrange (lower part).](image)

![Fig. 4: Modelled standard to mean reverberation intensity (SMR) level from a single (dashed line) and 20 realizations (solid line) of back propagating time series.](image)

The back-scattered time series calculated in the band 800-1800 Hz is shown in the upper part of Fig. 3 in a 2-s time window and received at a single hydrophone located 1 m from the source, i.e. no beamforming. Uniform random noise has been added to the time series requiring equal signal-to-noise ratio as observed in the data. Similar computations were performed along 20 radials covering ¼ of the elliptic inclusion centred at a constant range of 5 km from the source. Unfortunately, the 2-s time window used in these 20 calculations was not sufficient to prevent time wrap and interference of the arrivals. However, the 20 sequences obtained were used to determine the SMR of the matched-filtered intensity envelope as for the experimental data in Fig.1. This ratio was calculated within a 0.122-s sliding window (~190 m) for the 2-s time series. The result is shown in Fig. 4 for the single time series (dashed line) in Fig. 3 and the averaged over the 20 sequences (solid line). The SMR is higher for the modelling results than observed in the data. The SMR strongly depends on the signal-to-noise ratio, i.e. increased background noise level or less variation in
the geoacoustic properties of the inclusion compared to the surrounding sediment will lower the SMR. However, the modelling results demonstrate that slight local changes in the bottom reflection properties is a plausible source of clutter-like returns detected in sonar processing display.

4. CONCLUSIONS

Acoustic reverberation data acquired during the BASE’04 on the Malta Plateau, Mediterranean Sea, were analysed to obtain a scatterer map based on multiple transmissions and the standard deviation to mean reverberation intensity ratio. This ratio provides indications of region on the Plateau with diffuse and clutter-like reverberation. The majority of the sources causing clutter-like returns have been identified through supporting sea trial experiments. However, the cause of clutter-like reverberation from a particular area has not been identified although severe environmental surveys have been conducted in the area. The only clear evidence of clutter sources is from sub-bottom profiles which indicate possible inclusions in the upper sediment layer with geoacoustic properties different to the surroundings. Predictions of back propagating time series in the experimental region show that these inclusions are plausible clutter sources.

5. ACKNOWLEDGEMENTS

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REFERENCES

Structured Session 13

Acoustic Measurements of Sediments and their Transport

Organizers: Alex Hay, Peter Thorne & Mike Richardson
EVALUATION AND PRELIMINARY RESULTS FROM AN AUTONOMOUS 3-AXIS COHERENT ACOUSTIC DOPPLER VELOCITY PROFILER

Richard D. Cooke, Peter D. Thorne, Benjamin D. Moate, Paul S. Bell

Abstract: Laboratory based Acoustic Doppler Velocity Profilers (ADVP) have found recent application to sediment transport and turbulence studies, due to their ability to provide vertical profiles with sub-centimetric spatial resolution at turbulent and inter-wave timescales. Adaptation of these instruments to autonomous units capable of prolonged deployment in marine environments is not straightforward due largely to the necessary reductions in power consumption, and the resulting trade off between operating frequency and signal to noise ratio, impacting upon the systems ability to de-alias ambiguous velocities over the whole profile. In the present paper, the system design, laboratory evaluation, and preliminary field trial results for an autonomous, 3-axis coherent ADVP will be presented. The system described operates at 1 MHz utilising a dual pulse repetition frequency to combat phase wrap velocity aliasing, and provides profiles over a 1.5 m range recording at 10 Hz, with a deployment duration of up to 5 weeks. In the laboratory, the instrument has been evaluated in a sediment tower, by comparing measured settling velocities obtained from spheres to theoretical estimates. The field trial was carried out in a shallow estuary, where the ADVP was evaluated relative to independent measurements of the three velocity components, and an assessment of the instruments dynamic range was conducted. The implications of the results to using autonomous ADVPs to probe sediment transport processes in marine bottom boundary layers will be discussed.

Keywords: Autonomous, Coherent, Acoustic, Doppler, Velocity, Profiler, sediment, transport, instrument
1. INTRODUCTION

Traditional methods of obtaining velocity profiles have been limited to a small number of point measurements taken using intrusive instruments such as ADVs. In recent years the development of lab based coherent acoustic Doppler velocity profilers has enabled the non-intrusive measurement of velocity profiles at sub-centimetre resolution and inter-wave timescales, though the first generation of instruments has had limited success in the field [1]. The main limitation of early instruments was the requirement for a cabled connection to a shore-based computer, hence restricting the type and duration of experiments. Cabled instruments have been employed by other research bodies such as LEGI at University of Grenoble [2,3] but this cabling still limits the deployment opportunities. As no suitable autonomous instrument is at present available commercially, an autonomous battery powered system has been developed at POL. The initial development of this instrument was described earlier [4]. In the present paper, a performance evaluation of the instrument is summarised.

2. THE ADVP INSTRUMENT.

The ADVP (also referred to in previous publications [4] as the CDVP - Coherent Doppler Velocity Profiler) is designed to take a range gated profile of measurements in 3-axes, permitting sediment velocities and potentially concentrations to be obtained in the 2m near-bed region.

![Figure 1 – POL ADVP instrument](image)

It achieves this by recording Doppler shift in terms of phase shift relative to a known reference signal [5] hence the term coherent. Velocity vectors can be derived from this Doppler phase shift using a dual pulse repetition frequency technique [6]. The magnitude of the Doppler phase shift relates to acoustic backscatter signal (ABS) strength, thus providing information regarding sediment concentration.
The POL ADVP was configured to record unprocessed data to ease performance evaluation. Under normal use, the instrument is intended to process Doppler phase shift and ABS in-situ to reduce the enormous data overhead involved. For evaluation, the ADVP was configured to take measurements at 5cm intervals over 1.5m, using a dual-PRF of 410Hz/390Hz, with sufficient profiles taken to provide a post-processed recording rate of 10Hz. In normal use the system should typically be up to 16Hz recorded profiles at 1cm range gated measurements over a range of 1.5m.

3. LABORATORY EVALUATION.

The first evaluation stage was a laboratory based comparison with a Sontek-YSI Acoustic Doppler Velocimeter (ADV). The ADV provided single point velocity measurements, using a transmit frequency of 10MHz. The intention was to see if the ADVP (a) generates realistic data and (b) highlight any performance issues that need to be accounted for. The chosen form of comparison was measured settling velocity of particulates versus predicted settling velocity based on the work of Gibbs [7]. This was the simplest way configurable sediment flows of known concentration and velocity, could be generated. Evaluation was carried out using a Sediments Tower designed for acoustic studies and is described elsewhere [8,9,10]. This facility was designed to generate sediment suspension for observation with acoustic instrumentation. For the evaluation, sediment flows were generated using glass spheres finely sieved to give $\frac{1}{4}\Phi$ distributions with mean radii of 195µm, 137.5µm, 98µm and 34.5µm. These sizes were chosen to represent sediments from course sand to silt. Each $\frac{1}{4}\Phi$ distribution was evaluated separately by adding 160gm of it to the Sediment Tower water column with no water flow. A sample ADVP dataset is shown in Figure 2.

![Figure 2 - Settling velocity test using 390µm glass spheres](image)

Analysis of the settling velocity only ADVP datasets showed that a range of velocities could be detected varying over time and distance, due to the different particle sizes within each $\frac{1}{4}\Phi$ distribution. Also the ADV takes a point measurement hence is only exposed to the settling particulates for a very short time. Thus evaluation seemed best performed by comparing mean velocities for the vertical axis (Figure 3).
The ADVP is represented by an upper and lower mean velocity on account of the variation with distance and time. This result compares favourably with mean ADV velocity showing the ADVP to be giving sensible values. Differences from the Gibbs predicted velocities maybe due to fluid dynamics of the Sediment Tower, but the overall trend is similar.

4. FIELD TRIAL.

The second stage of evaluation was a field deployment at a suitable test site chosen in a local estuary, providing good site access, shallow depth of water and plenty of sediment transport. Each trial deployment involved the use of three instruments – the ADVP, a Sequoia Scientific LISST100X and a Sontek/YSI Hydra ADV. All instruments were deployed in close proximity on a tubular metal frame for comparison (Figure 4).

![Figure 3 - Comparison of ADV and ADVP mean settling velocities](image1)

![Figure 4 - Hilbre Island Field Trial Instrument Frame (aerial photo by Mills Photographic)](image2)
The ADV provided comparison measurements of velocity whilst the LISST quantified the amount and size distribution of suspended sediment. For each trial the instruments were deployed for a 24-hour period allowing measurements to be taken over two tidal cycles.

Initial analysis of data from the field trial is encouraging. Direct comparison of vertical velocities from the ADCP and ADV is possible at this time due to the slower vertical velocities not exceeding the first aliasing limit for both ADVP PRFs (Figure 5). By taking ADVP data from the range bin closest to the ADV sampling volume height, we can see that the measured velocities are visually comparable even if they are not identical (Figure 6). This lack of identical profiles could be due to the 12cm horizontal spatial difference between the ADVP and ADV measuring volumes and the ADV operating at 10MHz compared to the ADVP’s 1MHz. To broadly compare the two datasets the distribution of velocities measured was examined (Figure 7). As can be seen, these are in close agreement, suggesting the ADVP instrument’s performance is close to that of the ADV.
5. SUMMARY AND FUTURE WORK.

To summarise, an autonomous coherent acoustic Doppler velocity profiling instrument has been developed at POL to assist in furthering sediment transport studies. The POL ADVP (a.k.a. CDVP) has undergone laboratory tests and field trials to examine its performance. Preliminary results show the ADVP to be yielding velocity results comparable to a single point commercial instrument. Future work on the field trial datasets will produce full velocity profiles in the near-bed region. Dealing with will require careful application of techniques developed at POL for the previous cabled ADVP [8]. More rigorous and reliable de-aliasing techniques will also be developed.

6. REFERENCES.

[6] Peter Hardcastle, Proudman Oceanographic Laboratory, private correspondence.
MEASUREMENTS OF THE SCATTERING CHARACTERISTICS OF SEDIMENT SUSPENSIONS WITH DIFFERENT Mineralogical Compositions

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Abstract: Acoustic studies of suspended sediments often assume the dominant mineral in suspension is quartz, the density and intrinsic scattering properties of which are implemented when inverting acoustic backscatter data collected at sea. However, compositional analysis studies of suspended and sea-bed particulate material show a wide range of mineral species contribute to the inorganic fraction of sediments in the marine environment. Whilst no theoretical framework exists to predict the acoustic properties of irregularly shaped sediment grains, the density, compressional, and shear wave velocities of common marine mineral species can vary by up to a factor of two. In this study, we present and compare measurements of the intrinsic scattering parameters, namely the normalized total scattering cross section, $\chi$, and the backscatter form function, $f$, obtained from homogenous suspensions of irregularly shaped sand sized grains of both magnetite and quartz. Our preliminary measurements suggest that in the geometric scattering regime, $\chi$ is enhanced for magnetite sands by ~ 100 % relative to quartz. Similarly, measurements of the form function for magnetite sands are enhanced by ~ 33 % relative to quartz in the geometric regime, though no measurable difference was observed in the Rayleigh regime. The implications of these results for acoustic backscatter data collected at sea are discussed.

Keywords: Scattering, form function, cross section, sand, sediment, magnetite, mineralogy.
1. INTRODUCTION

Suspended marine sands significantly scatter underwater sound at MHz frequencies, with the suspended concentration and size controlling the backscattered intensity\textsuperscript{1,2,3}. Utilising this premise, monostatic Acoustic Backscatter Systems (ABS) have been developed over the past two decades, designed to collect profiles of suspended sediments in the bottom 1 – 2 m above the bed\textsuperscript{4}. Acoustics offer the advantages of non-intrusive measurements, with centimetric resolution, at turbulent and inter-wave timescales\textsuperscript{5}.

In recent years, analytical inversions of multi-frequency ABS data have facilitated non-empirical estimates of suspended concentration and size\textsuperscript{4,6}. Such inversions require knowledge of the acoustic scattering properties of the particles, typically characterised by two dimensionless parameters; the backscatter form function, $f$, and normalised total scattering cross section, $\chi$. Physically, $f$ describes the backscattering characteristics of a particle relative to its geometrical size, whilst $\chi$ quantifies a particles total scattering over all angles, relative to its geometrical cross section, and is proportional to particle scattering attenuation losses. For irregularly shaped particles such as natural sands, no analytical theoretical solution exists to describe $\chi$ and $f$. Consequently, to facilitate inversion of marine ABS data, $\chi$ and $f$ have been determined experimentally for irregularly shaped quartz based sediments\textsuperscript{1,2,3}.

Compositional analysis studies of suspended and sea-bed particulate material however, show a wide range of mineral species contribute to the inorganic fraction of sediments in the marine environment\textsuperscript{7,8,9}. Minerallogically, sediments are usually divided into light and heavy fractions, based upon grain density, $\rho$. In marine sediments, commonly occurring light minerals include quartz, feldspar, calcite, mica and finer clay minerals, with common heavy minerals including garnet, ilmenite, magnetite, and zircon\textsuperscript{7,9,10,11,12}. Whilst many marine sediments consist of up to 95 % light minerals by mass, feldspar and calcite are often in equal or greater abundance then quartz in the silt to course sand size range\textsuperscript{8,9,13}. At the other extreme, heavy minerals such as magnetite and ilmenite can dominate the bulk sediment in some coastal and inter-tidal regions, giving rise to so-called black sands\textsuperscript{7,12,14}.

To date, no measurements of $\chi$ and $f$ for non-quartz based sediments have been reported. Modelled scattering predictions for spheres having $\rho$, compressional, and shear wave velocities ($V_P$ and $V_S$ respectively) equal to those for garnet have been presented\textsuperscript{6}, though it is unclear whether the differences imparted to $\chi$ and $f$ due to mineralogy are significant, when compared to the enhanced scattering that occurs due to the irregular shape of marine particles relative to spheres. The aim of this study was to assess to what extent, if any, the intrinsic scattering properties of magnetite sands differ to those of quartz. Magnetite was chosen for study here as it is one of the heaviest minerals known to occur abundantly at some locations in the marine environment, with $\rho = 5196 \text{ kgm}^{-3}$, compared to $2650 \text{ kgm}^{-3}$ for quartz.

2. MODELLING $\chi$ AND $f$

For a single size sphere of radius $a$, the theoretical far field $\chi$ and $f$ are respectively\textsuperscript{15,16}:

$$\chi(x) = \frac{2}{x^2} \sum_{n=0}^{\infty} (2n+1) \left| b_n \right|^2$$

$$f(x) = \frac{2}{lx} \sum_{n=0}^{\infty} (2n+1)(-1)^n b_n$$

where $b_n$ is taken from the literature\textsuperscript{17}, and $x = ka$, where $k = 2\pi/\lambda$, with $\lambda$ the wavelength of sound in water. Evaluation of Equations 1 and 2 requires knowledge of $\rho$, $V_P$ and $V_S$ for each mineral, which were taken from the literature\textsuperscript{18,19}.

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3. OBTAINING $\chi$ AND $f$ FROM ABS MEASUREMENTS

Scattering measurements were obtained in a sediment tower, capable of generating homogenous suspensions for ABS studies. An Aquatec® AQUAscat ABS operating at 0.5, 1, 2 and 4 MHz, was used to collect backscatter data at 0.01 m intervals over a range of 1.28 m. A pulse repetition frequency of 4 Hz was used to allow each transmission to fully dissipate before the next, with a system generated average produced every 32 pings, providing 1 recorded profile every 8 seconds. ABS data was collected from sands sieved into narrow ¼ $\phi$ size fractions, where $\phi = -\log_2(d)$, with $d$ the particle diameter in mm. ¼ $\phi$ size fractions comprise a nominally single size of sediments, thus enabling the measurement of the intrinsic scattering properties. To ensure the necessary size, shape, and purity characteristics were obtained, all sediments were sourced from commercial suppliers.

If the phase of the backscattered signal is randomly distributed over $2\pi$, the root mean square backscattered voltage, $V_{RMS}$, can be shown to be:

$$V_{RMS} = \frac{K_t M^{1/2} f}{r \psi \sqrt{a \rho}} e^{-2r\alpha}$$  \hspace{1cm} (3)

where $M$ is the mass concentration of suspended sediment, $\psi$ accounts for the departure from spherical spreading in the transducer near field, $\alpha$ is the total attenuation over the range $r$, and $K_t$ is a system constant. $K_t$ is obtained by calibration and incorporates the receive sensitivity, electronic gain, and directivity of the transducer. The total attenuation includes contributions from water absorption, $\alpha_w$, and sediment scattering losses, $\alpha_s$, which add linearly, $\alpha = \alpha_w + \alpha_s$. The sediment scattering attenuation is proportional to $\chi$:

$$\alpha_s = \frac{3\chi M}{4a\rho}$$ \hspace{1cm} (4)

where all terms are as previously defined.

For a homogenous suspension, rearranging and taking the natural log transformation of Equation 3 yields a linear function of $\log_e(V_{RMS}r\psi)$ with range $r$ from the transducer:

$$\log_e(V_{RMS}r\psi) = \log_e \left( \frac{K_t M^{1/2} f}{\sqrt{a \rho}} \right) - 2r\alpha$$  \hspace{1cm} (5)

Therefore, $\alpha_s$ can be obtained from the slope of Equation 5 by subtracting $\alpha_w$ (which can be taken from the literature). Providing $M$, $\rho$, and $a$ are known, a profile mean value of $\chi$ can thus be calculated from Equation 4. In this way, accurate estimates of $\chi$ can be obtained providing $\alpha_s \geq \alpha_w$. Where $\alpha_s < \alpha_w$, small errors in the slope of Equation 5 reduce the accuracy of the estimated $\chi$. This limitation, combined with the maximum sand concentration that would not damage the sediment tower pumps being $\sim 2$ gl$^{-1}$, resulted in measurements of $\chi$ being obtainable at 4 MHz only. To utilise ABS data obtained at lower operating frequencies, $f$ was calculated from the measured $V_{RMS}$ by re-arranging Equation 3, and using a heuristic expression fitted to the measured $\chi$ to enable the computation of $\alpha_s$ at all frequencies.

4. RESULTS

Measurements of $\chi$ obtained from suspensions of the ¼ $\phi$ size fractions of magnetite and quartz sands are presented in Fig. 1a, along with heuristic fits to illustrate the general trends in the measurements. Theoretical predictions of $\chi$ for a ¼ $\phi$ distribution of spheres with the
same \( \rho, V_P, \) and \( V_S \) characteristics as both magnetite and quartz are also shown in Fig. 1a. These theoretical predictions were obtained by evaluating Equation 1 over a uniform size distribution for \( \frac{1}{4} \Phi \) size ranges of \( a \pm 10\% \). Fig. 1a shows the measured \( \chi \) for the magnetite sands were greater by up to \( \sim 100\% \) in the geometric scattering regime \( (x \gg 1) \), relative to quartz sands. The measured \( \chi \) for magnetite sands also showed significantly greater elevation when compared to the sphere predictions, then was observed for quartz.

Fig. 1: Measurements of (a) \( \chi \) and (b) \( f \), obtained from \( \frac{1}{4} \Phi \) size fractions of magnetite and quartz sands. Heuristic fits to the measurements are also shown, along with theoretical predictions for \( \frac{1}{4} \Phi \) distributions of magnetite and quartz spheres.
Measurements of $f$ obtained from the same suspensions are presented in Fig. 1b, again with heuristic fits to the measurements, and theoretical predictions obtained as above, using Equation 2. Fig. 1b shows the measured $f$ obtained from magnetite sands were elevated relative to those obtained from quartz sands, by up to $\sim 33\%$ at $x \sim 2$, in the geometrical scattering regime. Measured $f$ for magnetite sands also showed significantly greater elevation when compared to the sphere predictions, then was observed for quartz, at $x \geq 0.8$. However, below $x \sim 0.6$, in the Rayleigh scattering regime, there was no measurable difference in $f$ between the magnetite and quartz, with measured $f$ for both materials also being in closer agreement to the sphere predictions then at larger values of $x$.

5. DISCUSSION

These early results suggest that significant differences exist in the intrinsic scattering properties of irregularly shaped sands having different mineralogical compositions. Given the sphere predictions are similar in the region $x = 2 – 5$ (Fig. 1) for both materials studied, it is unclear why this similarity was not observed in the measured values of $\chi$ and $f$, with both being significantly elevated for the magnetite sands relative to quartz in this region. Some of the differences in observed scattering properties could conceivably be attributed to relative differences in the degree of departure of the magnetite and quartz grains from a spherical shape. Previous analysis of the quartz sands however have shown them to be highly irregularly shaped\(^3\), so this is considered to be an unlikely source of disagreement.

The observed differences in intrinsic scattering properties (Fig. 1) suggest that differences in mineralogical composition can significantly impact ABS data collected at sea. Given that at low values of $x$, $\chi$ and $f$ for magnetite are similar to those for quartz, Equation 3 suggests that for suspensions of equal mass concentration and size, $V_{RMS}$ for magnetite would be reduced by $\sim 1/\sqrt{2}$ relative to quartz, due to the higher density of magnetite (see Section 1). In contrast, at larger $x$, Equation 3 suggests $V_{RMS}$ would be only slightly reduced, by $\sim 1.33/\sqrt{2}$. Further work is required to establish if differences in scattering properties exist between these materials and other commonly occurring marine mineral species, as well as to determine if these differences are significant in the presence of broad size distributions of sediment suspensions with mixed mineralogical compositions.

6. ACKNOWLEDGEMENTS

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REFERENCES

Abstract: In shallow marine environments, the transfer of momentum in the near bed layer, over steep wave induced sand ripples and under regular two-dimensional flows, is dominated by the process of vortex formation and shedding. Sediment entrainment is highly influenced by this process, with maximum pickup associated with lee wake vortex shedding events around flow reversal. In contrast, maximum pickup over flat beds occurs at times of peak velocity amplitude. Whether this process occurs under irregular waves is an important question because vortex shedding is a highly effective mechanism for suspending sediments and in nature irregular waves are more common than regular waves. The Deltaflume of Deltares, Delft Hydraulics, the Netherlands, is a large scale flume facility where full scale irregular waves were generated and velocities and suspended sediments were monitored using an Acoustic Doppler Velocimeter (ADV) and Acoustic Backscatter System (ABS) respectively under a variety of wave conditions. The bedforms were monitored using a Sector Scanning Sonar (SSS) and Acoustic Ripple Profiler (ARP). An experiment where the ABS was above a 2D ripple crest has been identified and the data used to investigate the intra-wave structure of the sediment suspensions over a sand ripple crest by phase averaging the high resolution suspended sediment concentrations. The results show that over a ripple crest, peak sand pickup occurs around flow reversal during 60% of the wave-half cycles.

Keywords: Acoustics, ARP, ADV, ABS, sediments, suspensions, irregular waves, ripples
1 INTRODUCTION

Under waves and above steep wave induced ripples the wave boundary layer can separate in the lee of a ripple crest, thus forming a vortex which at flow reversal is ejected over the ripple crest into suspension. The near bed hydrodynamics are thus dominated by this coherent and repeatable process [1] which in turn strongly influences the suspended sediments. Sediment can be trapped in the vortex, which is advected both horizontally and vertically [2], and released as the vortex dissipates. Vortex shedding has been directly observed in the laboratory under regular [3] and irregular [4] waves and its existence inferred in the field [5]. Recent measurements of the suspended sediment concentration on an intra-wave time scale in a full scale wave flume using an Acoustic Backscatter System (ABS) [6] confirmed the process of vortex shedding at large scale under regular waves. However, whether vortex shedding occurs at full scale under irregular waves, more representative of nature, is still debatable. In this paper acoustic measurements taken at full scale under irregular waves and over a medium-grained sandy bed are examined. During the experiment the acoustic devices were moved above a ripple crest for 10 minutes and the ABS results from this time period are studied here on an intra-wave time scale. Whether vortex shedding occurs under each wave in an irregular wave sequence is questionable and this has been investigated.

2 EXPERIMENTAL INSTRUMENTATION AND METHODOLOGY

The Deltaflume of Deltares, Delft Hydraulics, the Netherlands, is a large scale flume 230m long, 5m wide and 7m deep. Irregular surface waves with a JONSWAP spectrum were generated over a sandy bed with a median grain diameter of \(D_{50} = 350\, \mu m\) in 4m of water. Under irregular wave forcing, parameterised by a significant wave height, \(H_s\), and peak spectral period, \(T_p\), a rippled bed formed above which there was substantial sediment suspended. To allow the rippled bed to reach equilibrium, measurements were made one hour after the onset of wave forcing. The measurements were made for 25 minutes and comprise the water surface elevation, near bed water particle velocities, bedforms and the concentration of suspended sediment. The water surface elevation was monitored by a wave staff suspended from a gantry above the flume. The acoustic devices were mounted in line with one another across the Deltaflume on a sliding platform which in turn was mounted to an instrument frame positioned over the sandy bed. Measurements of the near bed water particle velocities were made using a Nortek Acoustic Doppler Velocimeter (ADV) the sampling volume of which was nominally 10cm above the bed. The 25 minute velocity time series obtained from the ADV consisted of 3 orthogonal components each recorded at 16Hz. The bedforms were observed using a Sector Scanning Sonar (SSS) and an Acoustic Ripple Profiler (ARP) and were mounted 0.56m and 1m respectively above the bed. The SSS operated at 1.2MHz mechanically scanning through 400 angular steps during each revolution taking approximately 60s [7]. The ARP operated at 2MHz scanning a 4m profile of the bed approximately every 63s [7] such that a 2D profile of the bed, and how it changed with time, was built up over the 25 minute period. Measurements of the suspended sediments were made using an Acoustic Backscatter System (ABS). The ABS comprised 3 transducers operating at 1, 2 and 4MHz each of which recorded backscatter profiles at a vertical resolution of approximately 1cm and at a sample rate of 128Hz. The backscatter profiles were subsequently block averaged, to increase the statistical reliability of the results, to produce backscatter profiles at 4Hz. Finally, samples of the suspended sediment laden water were taken at 3 heights above the bed by the method of pump-sampling [8] using a pump-sampling system mounted on the instrument frame [9].
3 DATA ANALYSIS AND ABS INVERSION

The irregular surface waves were generated conforming to a JONSWAP spectrum [9]. For the experiment investigated here, $H_s = 0.64\text{m}$ and $T_p = 6.11\text{s}$, as calculated from the power spectrum of the 25 minute water elevation time series using the method of Soulsby [10]. The spectrum of the water surface elevation has been compared with a theoretical JONSWAP spectrum [11] based on $H_s$ and $T_p$ through a parameterisation [12]. The theoretical and observed power spectra compare well in both amplitude and width. Each time series component of the ADV record was despiked using a Phase-Space Thresholding Method [13] and rotated to correct for any misalignment of the instruments to the main flow directions [14]. Turbulent fluctuations have been removed from the horizontal along-flume ADV velocity time series by applying a rectangular low-pass filter. In order to phase-lock the along-flume ADV velocity time series to the ABS suspended sediment concentration time series, each $\frac{1}{4}$ wave cycle has been identified using turning point and zero crossing analysis on the ADV time series. The rippled bed was found to be non-migrating and in a steady state for the duration of the experiment and 5 minutes into the experiment the instruments on the sliding platform were positioned over the crest of a steep sided ripple and held stationary for 10 minutes. Each ARP transect during this 10 minute period was low-pass filtered with a Gaussian filtering window to remove high frequency fluctuations. A turning point analysis was performed to extract the spatial distribution of ripple wavelengths ($\lambda$), heights ($\eta$) and steepness’ ($\eta/\lambda$) for each profile.

The ABS backscatter time series from the 3 ABS transducers has been examined and the bed level determined. The backscattered voltage from the ABS can be expressed as

$$V_{\text{rms}} = \frac{k_s k_t}{\psi r} M^{1/2} e^{-2r(\alpha_s+\alpha_w)}$$

[15] where $\psi$ accounts for the departure from spherical spreading in the near field of the transducers [16], $r$ is the range from the transducers, $M$ is the mass concentration of the sediment, $k_t$ is the ABS system constant obtained by calibration [17], $k_s$ describes the backscattering characteristics of the sediments and $\alpha_w$ and $\alpha_s$ are the attenuation coefficients of the water and sediment respectively. The attenuation due to sediment can be described by

$$\alpha_s = \frac{1}{r} \int_0^r \xi(r) M(r) dr$$

[15] where $\xi$ is known as the sediment attenuation coefficient and is a function of the total backscattering cross sectional area and the arithmetic mean sediment radius each averaged over the particle grain size distribution. Therefore, solving equation 1 for $M$ is non-trivial and an inversion is required. The inversions performed here are based on methods [15, 18] which depend on describing the acoustic scattering properties of the suspended sediment through the average form function $<f>$ and total backscattering cross sectional area $<\chi>$, where $<\cdots>$ represents an average over the particle grain size distribution. $<f>$ and $<\chi>$ therefore depend on the type and grain size distribution of sediment in suspension. The average grain size distribution of sediment in suspension has been obtained using a laser diffraction analysis of the pumped-samples (PS). Because the backscattering characteristics of the sediments are based on a probability density function (PDF) describing the distribution of number of particles in suspension, it is important to convert the % volume distribution of the PS to such a PDF. The standard deviation to mean ratio of this log-normal PDF was found to be $\sigma/\mu = 0.33$ which is typical of natural sand sediments. Heuristic expressions of $f$ and $\chi$ based on
many measurements of natural sands [19] have therefore been averaged over this PDF producing $<f>$ and $<q>$ which are used in the inversions here. An initial implicit [15] inversion has been performed on the backscattered voltage root mean square averaged over the 10 minute time period where the ripple crest was directly below the instruments, yielding an average grain size and concentration profiles. Subsequently, an explicit inversion was performed on the time series of backscattered voltage using the grain size profile obtained above to ensure that the solution to equation 1 remained stable. The result was a time series of suspended sediment concentration profiles at 4Hz. Each wave cycle in this result has been identified through a process of phase-locking to the along-flume ADV velocity. An ensemble average of the suspended sediment concentration though a wave cycle over the 10 minute period considered was taken. Finally, the intra-wave variation in the suspended sediment reference concentration ($C_0$), taken at the bed level, under each wave has been calculated by extrapolation down to the bed level.

4 RESULTS

Under the irregular wave forcing there was a distribution of near bed orbital velocity amplitudes. The significant near bed orbital velocity amplitude $U_{0s}$ has been calculated [20] from both the non-filtered and low-pass filtered spectrum of the 25 minute ADV time series, yielding $U_{0s} = 0.31$ m/s to 2 significant figures in each case. This verifies the veracity of the filtering method. Fig. 1 shows the results from the SSS and ARP. The ARP was mounted adjacent to the ABS such that the ripple, the crest of which was directly below the ARP was therefore also inline with the ABS. The mean temporal average dimensions of this ripple from the ARP results are $\bar{\lambda}_0 = 0.33$m, $\bar{\eta}_0 = 4.68$cm and $\bar{\eta}_0/\bar{\lambda}_0 = 0.14$ with standard deviations of 0.01m, 0.32cm and 0.01 respectively.

Fig. 2 shows the result of the ensemble averaged intra-wave suspended sediment concentration (a), near bed along-flume velocity (b) and reference concentration, $C_0$, (c). There are two clear peaks in the sediment concentration close to the bed around 90° and 270° through the wave cycle, corresponding to times of flow reversal (Fig. 2b) peaks in $C_0$ (Fig. 2c). The sediment concentration decays with distance from the bed such that 20cm away from the bed the concentration has dropped an order of magnitude.
Fig. 2: (a) Intra-wave and height variation of the suspended sediment concentration, \( C \), ensemble averaged over a burst where the ripple crest was nominally below the ABS. This result was obtained from the 1MHz ABS transducer and is shown up to \( z = 20\)cm, where \( z \) is the height above the ripple crest. (b) and (c) show the intra-wave variation in the near bed velocity from the ADV and reference concentration \( (C_0) \) respectively.

Whether or not vortex shedding occurs under each wave has been assessed by examining the intra-wave variation in \( C_0 \). Those waves where vortex shedding occurred showed clear peaks in \( C_0 \) around flow reversal and this occurred during approximately 60\% of the wave-half cycles.

5 CONCLUSIONS

High frequency ABS, ADV and ARP instruments have been used to study the intra-wave sediment suspension processes over a steep ripple crest under irregular waves. Peaks in the suspended sediment concentration around the times of flow reversal have been observed which is consistent with a vortex shed over the crest of a ripple as the flow changes direction. These results are significant as they indicate vortex shedding does not occur under every wave during an irregular wave sequence but rather only during 60\% of wave-half cycles.

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7 REFERENCES

COHERENT DOPPLER SONAR PROFILING IN THE BOTTOM BOUNDARY LAYER: PROBLEMS AND POTENTIAL

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Abstract: Acoustic systems provide great potential for non-invasive sampling of near bottom sediment transport processes: acoustic backscatter can be inverted to infer the concentration of suspended particles, and Doppler sonar can be used to determine flow velocities. As with any acoustic system there are many performance characteristics that must be taken into consideration and any practical system represents a compromise between conflicting requirements. The problems that we explore include: (1) what maximum speed can be measured and what limits accuracy, (2) what spatial resolution can be achieved and (3) how close to the bottom can measurements be made. We utilize a computer model of coherent acoustic backscatter to predict the performance of various system geometries. Model simulations indicate that horizontal velocities as high as 4 m s\(^{-1}\) can be resolved. Our working prototype system can generate three component velocity profiles with 0.3 cm range bins over a 40 cm range interval at a rate of about 100 s\(^{-1}\). Examples of prototype performance are provided through laboratory observations of a turbulent jet.

Keywords: Coherent sonar, Doppler, sediment transport
1. INTRODUCTION

Pulse-to-pulse coherent sonar or Doppler can provide high resolution velocity profiles without causing flow interference. Laser Doppler and PIV (particle image velocimetry) provide alternative high-resolution non-invasive measurement methods but only acoustic systems can operate reliably in turbid waters: the acoustic systems are therefore particularly attractive for field measurements. These systems exist as commercial instruments in the form of Acoustic Doppler Velocimeters which provide high resolution point measurements [1]. Profiling systems are also possible ([2] and [3]) but the occurrence of range and speed ambiguities have kept these systems from realizing widespread application. In this paper, performance limitations on a practical pulse-to-pulse coherent Doppler profiler are explored using laboratory observations made with a system prototype and with data simulations made using a model of coherent acoustic backscatter [4].

2. PULSE-TO-PULSE COHERENT SONAR

Coherent sonar estimates velocity by considering backscatter from two successive transmit pulses. If the targets or acoustic scattering responsible for the backscatter remain coherent between the two returns, then changes in phase in the backscatter can be related to the flow speed of the water. For a monostatic system considering direct backscatter,

\[ v = \frac{\delta \phi \lambda}{4\pi \tau} \]  

(1)

where, \( \delta \phi \) is the change in phase, \( \lambda \) is the acoustic wavelength, and \( \tau \) is the time interval between pulse transmissions. Position of the sample is determined from the directivity of the acoustic beam pattern and range given by,

\[ r = \frac{Ct}{2} \]  

(2)

where \( C \) is the speed of sound, and \( t \) is the time since the pulse was transmitted. Range ambiguities occur when \( t \) becomes larger than \( \tau \) so that backscatter is received from two ranges at the same time. Equation (2) gives the nominal range ambiguity when \( t = \tau \).

The limitation of measuring phase to within \( \pm \pi \) radians ultimately imposes a maximum unambiguous speed in Equation (1) of

\[ \delta_v = \frac{\lambda}{4\tau}. \]  

(3)

The characteristics of any functioning system are largely determined by how a balance is struck between the range and speed ambiguities imposed by Equations (2) and (3).

We have developed a system designed for sediment transport studies in a near-shore environment: here we expect horizontal velocities of several meters per second with vertical velocities close to the bottom boundary limited to 10’s of cm s\(^{-1}\). For this purpose we would like to observe profiles that extend over a 40 cm interval with the acoustic transducers remaining about 1 m away from the bottom. Observations of three independent velocity components must be made at the same point because structure in the flow is small compared to the overall profile scale. The bistatic transducer geometry required to achieve these operating goals is shown in Fig. 1. An acoustic frequency of about 2 MHz is appropriate in
this environment as it is sensitive to the suspended particle sizes that occur and absorption at this frequency does not restrict the range to less than the 1 m objective.

One of the most critical parameters in determining system performance is the pulse interval ($\tau$) because it determines the ambiguity velocity. With the instrument height at about 1 m above the bottom, the time for a pulse to reach the bottom essentially sets the minimum pulse interval at 1 ms, and with the presence of the bottom, the range ambiguity is not a (major) concern. Operating at a nominal frequency of 2 MHz, these operating parameters in Equation (3) give an ambiguity velocity of 20 cm s$^{-1}$. This value is comparatively low but the beam geometry assures that only a small component ($\sin \theta$) of the large horizontal velocities are measured: for $\theta = 14^\circ$ in Fig. 1, the maximum (unambiguous) horizontal velocity is 1.6 m s$^{-1}$.

The transducer geometry is selected to mitigate the problem with ambiguity velocities but they still pose a practical problem. In the present prototype system, this problem is dealt with by using a broad-band transmit pulse so that velocities can be sampled at multiple frequencies simultaneously. This approach allows the actual velocity to be extracted by requiring the velocity estimates from the various frequencies to agree [5]. For two frequencies, velocity can be estimated as

$$v = \frac{(\delta \phi_2 - \delta \phi_1)}{4\pi(f_2 - f_1)\tau} C$$

(4)

where $f_1$ and $f_2$ are the two frequencies being used and $\delta \phi_1$ and $\delta \phi_2$ are the respective phase changes. In Equation (4), the ambiguity velocity can be made arbitrarily large by making $f_2 - f_1$ small. However, as the velocity increases, the uncertainty in phase estimates increases and eventually, when that uncertainty is comparable to the phase differences themselves, Equation (4) can no longer extract meaningful velocities.

3. SYSTEM PERFORMANCE
System performance at high flow speeds was simulated by creating coherent acoustic backscatter with a computer model [4]. The transducer geometry was taken as that shown in Fig. 1 and operating frequencies 1.4, 1.7, 2.0, and 2.4 MHz were selected. For this example, a 4 m s\(^{-1}\) oscillating horizontal velocity was used. The reconstructed velocities (Fig. 2a) become unreliable at just below that maximum speed. At that speed some of the constituent velocities have wrapped several times (for example, the 2.0 MHz velocity shown in Fig. 2b has wrapped twice). The degradation in phase estimate accuracy accounts for the increased velocity variance and is a result of the reduced target residence time associated with higher speed flow ([2], and [6]). Essentially, it is the de-correlation of the signal that limits the maximum observable speed as is indicated by the correlations in Fig. 2c.

![Fig. 2 Modelled results extracting velocity from 4 m s\(^{-1}\) amplitude oscillating horizontal velocity flow: a) resolved x-component velocity based on 1.4, 1.7, 2.0, and 2.4 MHz simulated data, b) component measurement at 2.0 MHz frequency with ambiguity velocity of 0.12 m s\(^{-1}\), and c) correlation for the 2.0 MHz signal.](image)

High velocities are not the only flow condition that can lead to reduced data quality, de-correlation from any process will increase velocity estimate uncertainty. Turbulence can also cause de-correlation as is demonstrated by a coherent Doppler profile across a turbulent jet shown in Fig. 3 (see [7] for a detailed description of this experimental configuration). The data presented in Fig. 3 were collected using the same instrument geometry shown schematically in Fig. 1 with operating parameters similar to those used in the simulation presented in Fig. 2. Figures 3a and 3b show the axial (x-component velocity) and jet radial velocity (represented by the y-component velocity). Axial velocity dominates the flow but there is a radial component associated with the outward spreading of the jet. In this case, axial velocities are no greater than 15 cm s\(^{-1}\) but correlations in the jet axis are reduced to between 0.4 and 0.6 for 2.3 and 1.4 MHz data respectively. Reduced correlations beyond cross jet positions of ±8 cm are associated with low acoustic signal levels and are not related to flow conditions. These correlations are much lower than the correlations realised at much
higher speeds in the uniform flow simulations presented in Fig. 2. The difference here is that the jet flow is turbulent and the random motions associated with turbulence lead to de-correlations that inhibit the ability of coherent Doppler to measure velocities. The impact of this de-correlation on velocity estimates is minimised by using the shortest possible transmit pulse interval.

![Velocity Profiles](image)

*Fig. 3: Coherent Doppler profile of average a) axial, and b) radial velocities and c) correlations at 1.4 MHz (solid line) and 2.3 MHz (dashed line) made across a turbulent jet.*

The range resolution of profiling systems is typically determined by the transmit pulse length as

\[
\Delta r = \frac{Cr}{2}
\]

which is ultimately limited by the system bandwidth. The 250 kHz bandwidth (for a given frequency band), employed here allows a pulse length as short as 4 μs and gives a range resolution of 3 mm. However, the 3 dB width of the intersecting bistatic beams is as large as 2 cm, and, given the sloping geometry of the resulting sample domains, it is not clear what vertical resolution is being achieved.

The coherent backscatter model was used to explore depth resolution by simulating a flow with a constant velocity of 50 cm s\(^{-1}\) above 70 cm depth and a velocity of 25 cm s\(^{-1}\) beyond this depth: the bottom is placed at a depth of 80 cm. The transducer geometry for these trials was configured with transducer spacings of 15, 25, and 35 cm (\(\Delta s\) in Fig. 1). In all cases the intersection depth of the bistatic beams was maintained at 70 cm so that the angle \(\theta\) increased as \(\Delta s\) increased. Velocity profiles generated for this geometry are shown in Fig. 4a where the velocity discontinuity is clearly visible at 70 cm and is larger than the 3 mm range resolution of the system. Velocity shear is used to determine the depth interval affected by the discontinuity: shear should be zero everywhere with a discontinuity at 70 cm depth.
Shear is plotted in Fig. 4b for depths between 69 and 71 cm (indicated by dotted lines in Fig. 4a). Fig. 4b shows non-zero shear for all three geometries in a band of about 1.5 cm in depth with a slight reduction visible in the 15 cm spacing. The high degree of similarity indicates that depth resolution is not strongly dependent on transducer spacing.

Also shown in Fig. 4c for the 69 to 71 cm depth interval (the same interval as shown in Fig. 4b) are profiles of correlations for the 1.6 MHz simulation. The presence of the velocity discontinuity is clearly visible in these profiles and demonstrates that the presence of multiple velocities within one depth sample leads to signal de-correlation.

The issue of range resolution is particularly critical at a solid boundary because of the large discontinuity in signal amplitude that occurs and this gives rise to two forms of velocity contamination. First, for the bistatic system being considered, there is contamination of the near bottom profile data by side-lobes of the acoustic beams. This effect is visible in Fig. 4a where near bottom velocities are biased to zero as far as 3 cm above the bottom (for the case of a 35-cm transducer spacing). The second problem arises because of the use of digital demodulation filters. Exceptional frequency response characteristics are achieved by using high order filters with coefficients that extend 10’s of μs forward in time. When these forward reaching coefficients encounter the large amplitude signal from the bottom echo they can lead to data contamination several centimetres above the bottom. Figure 5 demonstrates this contamination using backscatter model simulations of a uniform horizontal flow of 10 cm s⁻¹ above a flat bottom (the model does not simulate a boundary layer). Fig. 5a shows the velocity profile for a 63-coefficient and a 7-coefficient filter when the bottom backscatter is
14 dB stronger than the volume backscatter above the bottom. In this case, contamination extends about 2 cm above the bottom with little difference between the two filters. Fig. 5b shows velocity profiles simulated with a 26 dB signal strength difference and in this case, the 63-coefficient filter contaminates data up to 4 cm above the bottom, while use of a 7-coefficient filter reduces that to 2 cm (equivalent to the zone of contamination expected from the transducer side-lobes). This contamination is reduced if the bottom echo saturates the digitiser because the difference in backscatter level is then limited: the least contamination occurs when comparatively high signal levels are measured in the backscatter just above the bottom.

![Graph showing velocity profiles](image)

**Fig. 5:** Sampled velocity profile for a modelled uniform flow of 10 cm s$^{-1}$ over a bottom using 63-coefficient (solid line) and 7-coefficient (dashed line) demodulation filters and bottom backscatter strengths of a) -14 dB, and b) -26 dB relative to the bottom echo.

### 4. CONCLUSIONS

We have evaluated the practical constraints for pulse-to-pulse coherent sonar profiling velocity in a bottom boundary layer. The system we consider operates at a frequency of about 2 MHz with transducers positioned 1 m above the bottom. Performance has been evaluated using a model of coherent acoustic backscatter and by laboratory observations of a turbulent jet. Profiles can be generated at a rate of up to 100 per second in 3 mm range bins over a sample depth interval of 40 cm. Model results show that the effective depth resolution of the system is about 1.5 cm constrained by transducer beam geometry effects. Depth resolution is also affected by the presence of strong backscatter gradients both because of transducer side-lobes and digital filter characteristics. For a given transducer geometry, sample proximity to the bottom is optimised by adjusting gain levels so that backscatter levels above the bottom are high and by using comparatively short digital filters with a consequent compromise of
filter performance. Model results also show that the simultaneous use of multiple frequencies allows resolution of \((\text{horizontal})\) velocities up to \(4 \text{ m s}^{-1}\). The maximum velocity is not constrained by the ambiguity effects but rather by de-correlation caused by scatterers moving rapidly through the sample volume. De-correlation is also caused when a range of velocities are present within a sample volume such as can occur due to velocity shear or turbulence. Observations of flow in a turbulent jet with axial speed of only \(15 \text{ cm s}^{-1}\) show mean correlations reduced to below 0.6, much lower than expected for a uniform flow of this speed.

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ACOUSTIC MEASUREMENTS OF BOUNDARY LAYER FLUX PROFILES OVER A SANDY RIPPLED BED UNDER REGULAR WAVES

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Abstract: The study of boundary layer sediment transport processes requires contemporaneous measurements of the bedforms, the flow and the sediment movement. Obtaining these three parameters, at the required temporal-spatial resolutions, has been traditionally difficult, especially within a few centimetres of the bed. To circumvent some of these difficulties acoustic techniques have been and are being developed. Here we look at the deployment of an acoustic backscatter system, ABS, an acoustic ripple profiler, ARP, and an acoustic Doppler velocity profiler, ADVP, to measure sediment entrainment processes above a rippled bed under regular waves. High resolution acoustic observations of the suspend sediment concentration, flow and bedforms have been collected. Here we report on some of the initial results obtained from this study.

Keywords: Acoustic, backscattering, Doppler, velocity, sediments, ripples, bedforms
1. INTRODUCTION

Wave induced ripples are a common occurrence on sandy beds in coastal waters. The ripples are formed by the oscillatory action of the fluid due to the waves, which through an initial bed deformation, develops into a rippled bed due to feedback between the flow, the mobile sediments and the evolving bedforms. In many cases they have dimensions directly related to the near bed oscillatory orbital displacement of the fluid particles and as such are known as orbital ripples. In many cases the dimensions of such ripples so formed are of the order of decimetres in length and centimetres in height. Once formed the ripples have a profound impact on the boundary layer physical processes of the hydrodynamics and the sediment mobility. For ripples of low slope, $\eta_r/\lambda_r \leq 0.1$, where $\eta_r$ is the ripple height and $\lambda_r$ is the ripple wavelength, the ripples enhance the near bed mixing relative to a plane bed, although the fundamental dynamics are usually considered to be comparable to that of a plane bed. However, if the ripples are relatively steep[1] with $\eta_r/\lambda_r \geq 0.1$, then the dynamics change completely and mixing close to the bed is considered to be dominated by a coherent process involving boundary layer separation and vortex formation. Direct measurements of the hydrodynamics and sediment structures, within this wave boundary layer, above a rippled bed, at field scales are relatively few and it is only recently that detailed observations of the combined hydrodynamic-sediment processes have begun to become available. In this paper we present some measurements of sediment dynamics over a rippled bed collected using recently developed acoustic technologies. The study was carried out in a large scale flume facility in Barcelona, Spain. Combining an acoustic multi-frequency backscatter system, ABS, with an acoustic ripple profiler, ARP, and an dual frequency acoustic Doppler velocity profiler, ADVP, simultaneous collocated measurements of the suspended sediments, bedforms, and flow were collected above a rippled bed. Here we describe the analysis of the data and some provisional results.

2. EXPERIMENTAL ARRANGEMENT AND MEASUREMENTS

The study was carried out in late Autumn 2008, in a large-scale wave flume facility located at the Maritime Engineering Laboratory (LIM) at the Catalonia University of Technology (UPC) CIEM, Barcelona, Spain. The flume dimensions are 100 m long, 3 m wide and 5 m deep. The size of the flume allows hydrodynamic and sediment studies to be carried out at near full scale. Illustrations of the flume are shown in Fig. 1. For the study medium sand was placed in the bottom of the flume to a thickness of 0.5 m. The water depth above the sand was between 1.6 m-1.9 m. The waves used for the work ranged in height, H, between 0.30 m-0.55 m and had a fixed period, T, of 6.5 s. Calculations prior to the experiment indicated that waves with these parameters, for the water depths used, would have sediment processes varying from dynamically plane through to the vortex ripple regime.
Fig.1: Photographs of the Barcelona flume facility.

Fig.2: Schematic showing the layout of the Acoustic Ripple Profiler, ARP, Acoustic Backscatter System, ABS and the two Acoustic Doppler Velocity Profilers, ADVP.
The experimental set up is shown above in Fig. 2. Suspended sediment concentration profiles were obtained using the ABS. The system operated at 0.5, 1.0, 2.0 and 4.0 MHz. During the study the pulse repetition frequency was set at 64 Hz. The pulse length and range sampling were selected to provide 0.005 m vertical range resolution with 200 samples, thereby covering a range of 1.0 m. To measure the bedforms the ARP, based on a rotary narrow beam scanner, was used to measure the height of the sand surface over a transect of 4 m. Transects were collected each minute during the data collection periods, which were typically between 15 min – 30 min. To obtain the vertical and horizontal velocity components dual frequency ADVP’s were used. These operated at 1.25 MHz and 2.0 MHz. From the phase and magnitude of the backscattered signal these respectively provided 12.5 Hz velocity and concentration profiles, at two locations above the bed, with 0.003 m vertical resolution.

3. DATA ANALYSIS

Suspended sediment measurements. The objective of the study was to measure collocated velocity and suspended concentration above known bedforms to assess the processes of sediment entrainment. To this end the ABS was calibrated using a suspensions of spheres[2] and suspended concentration profiles obtained based on an inversion[3] using a median suspended particle size of $d_s=200 \mu m$, which was 80% of the median bed size. The results from the acoustic inversion are shown in Fig. 3. The suspended sediment concentration is seen to steadily increase in magnitude as $H$ is increased, with typical profiles[4,5] which rapidly reduce close to the bed followed by slower reduction with increasing height above the bed.

![Fig.3](image_url)  
*ABS measurements of suspended sediments concentration with height above the bed for increasing values of wave height.*

To convert the ADVP backscattered signals to suspended sediment concentration, the same inversion was performed on their backscatter data and the results matched to the ABS concentration profiles. Comparison of the ABS and ADVP results are shown in Fig. 4. The
results from the ABS and the ADVP are deemed sufficiently comparable that the veracity of the ADVP concentration measurements is considered high.

Fig. 4: a) b) Comparision of suspended sediment concentration profiles from the ABS and the ADVP c) d) Comparison of the intra-wave suspended sediment concentrations at the two ADVP frequencies

Bedform measurements. As mentioned earlier, bedforms have a profound impact on sediment entrainment and transport[6], due the changing flow and sediment dynamics as the bedforms develop. Examples of the bedforms from the study are shown in Fig. 5. This shows the development of a 4 m transect over time. For low wave conditions, H=0.3 m, the average ripple height and wavelength were respectively \( \eta_r=0.009 \) m and \( \lambda_r=0.11 \) m, this results in a slope of \( \eta_r/\lambda_r=0.08 \). Ripples of these dimensions and slope are considered to increase the bed roughness, i.e. enhance near bed turbulence, however, the dynamics is considered to be nominally similar to those of a plane bed. For the second case shown in Fig. 5b, H=0.55 m, the values for the ripples were \( \eta_r=0.03 \) m, \( \lambda_r=0.26 \) m and \( \eta_r/\lambda_r=0.12 \). The
expectation for ripples with these dimensions and slope is that flow separation would occur and the sediment dynamics be related to the process of vortex entrainment.

**Fig.5: Measurements of bedforms for a) H=0.3 m and b) H=0.55 m.**

**Combined measurements of the flow and suspended sediments.** As shown in the ‘Suspended sediments’ section the amplitude of the ADVP backscattered signal can be used to obtain suspended concentration. However, the rate of change of phase of the same signal can also be utilised for flow measurements [7-10]. This technique can provide high spatial-temporal detailed velocity profiles in the bottom boundary layer. Therefore using the ADVP, collocated combined profile measurements of the velocity components and the suspended sediment concentration were obtained. Provisional results are shown in Fig. 6. In Fig. 6a, the horizontal flow component, u, is superimposed on the instantaneous flow vectors in the uw plane, where w is the vertical velocity component. In Fig. 6b, the corresponding concentration field is given; this was simultaneously obtained from same ADVP signal. It can be seen that the event of the near bottom maximum in suspended sediment concentration coincides with a strong upward surge in the velocity field. The data are presently being analysed in order to understand the physical processes which gave rise to the type of results shown in Fig. 6, although it is already evident that such detailed combined velocity and concentration observations will provide new insight into bottom boundary sediment dynamics.
Fig. 6: Collocated simultaneous intra-wave measurements of; a) the velocity field, b) the concentration field, obtained using an ADVP.

4. CONCLUSIONS

The aim of the paper has been to provide a description of the application of acoustics to the study of sediment entrainment over sandy rippled beds. An overview of the facility, instrumentation and data collected has been presented. Detailed analysis of the boundary layer measurements, aimed at furthering understanding of the mechanisms of sediment entrainment is ongoing. Here we have presented some early results from this analysis. It is expected that the detailed contemporaneous observations of flow and suspended sediments collected with the ADVP, coupled with the bedform measurements, will provide a data set which will both assess and aid the development of theoretical models describing sediment transport over complex bedforms.

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On using acoustic profiling to study bottom boundary layer dynamics in unsteady sediment-laden open-channel flow

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Abstract: Sediment transport dynamics in the bottom boundary layer of unsteady open-channel flows directly impacts on processes in the water column. To study the bottom boundary layer interactions requires observations of the bed, flow and sediment movement. Recently developed acoustic techniques can provide measurements of these three parameters at high temporal-spatial resolution. Here we employ acoustics to provide quasi-instantaneous simultaneous and collocated profile measurements of (i) the three orthogonal components of flow using phase coherent techniques, and (ii) suspended sediment particles using acoustic backscatter. This allows for an in depth investigation of bottom boundary layer physics in unsteady sediment-laden flow.

Keywords: unsteady flow, fine sediment, resuspension, ADVP.

1. INTRODUCTION

Suspended sediment transport in rivers directly impacts on the physical and biogeochemical processes in the water column. While some progress in understanding sediment fluxes has been made under uniform flow conditions [1], [2], much less is known about unsteady flows, even though flows in rivers and open channels are unsteady most of the time. Unsteady flows may lead to initiation of sediment resuspension, generate bedforms and bed topography changes. Therefore, in order to capture the essential dynamics of unsteady flow, instrumentation is required which can simultaneously measure hydrodynamics, sediment concentration in the whole water column and bed morphology with sufficient spatial and temporal resolution to resolve turbulent scales. Acoustic methods are well suited to fulfilling these requirements.
Acoustic Backscattering Systems (ABS) allow capturing the Doppler phase angle and the intensity of the backscattered signal. The phase angle has been used in Acoustic Doppler Velocity Profilers (ADVP; [3]) and was extended to full 3 velocity component instruments [4], capable of resolving turbulence scales in space and time. They were further improved in the hardware [5] and software [6], [7], [8] domain and are today reliable instruments.

Backscattered intensity can be inverted into particle size and concentration after calibration [9], [10]. An iterative inversion method has been proposed [11] and an explicit inversion method is also available [12]. However both methods suffer from errors propagating through the profile.

This problem was overcome by using backscattered and forward scattered profile signals, thus providing attenuation compensation even in high particle concentrations as long as multiple scattering is avoided. Integrating this approach into the existing ADVP, a particle flux profiler was developed [13] which simultaneously determined the 3-axis velocity field and the suspended particle concentration field in the same scattering volumes of the profile.

The inconvenience of this solution (requiring forward scattering measurements ) for field applications of the system was overcome by a new approach based on the exploitation of backscattering intensity at two (or more) emitted frequencies [14], [15]. The advantage of this solution is that two relatively close frequencies (such as 1.25 Mhz and 2 Mhz) completely resolve the concentration field of fine particles typically found in hydraulic applications. This frequency range can easily be handled by a single transducer. Therefore, a two frequency backscattering intensity profiler can be integrated in the existing ADVP and provides a particle flux profiler which is unlimited in its application in laboratory and field studies in rivers, open channels, oceans and lakes. Again, 3-axis velocity and particle concentration profile information can be obtained simultaneously, co-located in the same scattering volumes within the profile, resolving turbulence scales in time and space.

The location of the bed is easily extracted from ADVP or particle flux profilers by the strong echo of the backscattering intensity or from the zero velocity in the Doppler phase.

In the present study, the acoustic two-frequency particle flux profiler has been applied to unsteady turbulent open-channel flow. The objective is to determine whether the instrument is capable of detecting the onset of fine sediment resuspension.

2. EXPERIMENTAL SET-UP AND PROCEDURE

The measurements were carried out in a glass-walled open-channel which is 17 m long and has a rectangular cross section 0.6 m wide and 0.8 m deep. The bottom was covered with a 0.1 m thick gravel layer (D$_{50} = 5.5$ mm). ADVP profiling was carried out on the centerline of the channel about 15 m. A bifrequency (1.25 Mhz and 2 Mhz) system was used.

The hydrograph for the experiment consisted of a base discharge, followed by the rising stage of the unsteady flow where the discharge was linearly increased over a period of 30 s. Table 1 gives the hydraulic parameters at the beginning and the end of the hydrograph. No sediment resuspension occurred during the initial phase of the unsteady flow.

In order to investigate resuspension of fine sediments, the coarse bed was covered with an about 2 mm thick layer of sand with D$_{50} = 0.16$ mm on a surface area extending about 1m upstream from the location of the ADVP. The acoustic measurements were complemented by simultaneously collected high-speed videos in the center of the channel, just upstream of the ADVP location. Only a narrow slot (about 1 cm in transversal depth) of the flow in the center of the channel was illuminated.
3. RESULTS

The water level change in the unsteady range is not linear, even though the discharge increased linearly. This indicates a deformation of the wave progressing through the channel. The unsteady flow range was divided into a sequence of 100 mean profiles. During the unsteady flow, all profiles followed a logarithmic law in the inner layer, documented by examples in Fig. 1 where data values up to the velocity maximum are included. Therefore mean flow dynamics of unsteady flow are comparable to steady flow conditions.

![Fig. 1 Examples of mean longitudinal velocity profiles for the unsteady range of the hydrograph.](image)

Backscattered intensity was extracted from the same data from which the velocity components discussed above were obtained. Results are plotted in Fig. 2 for the unsteady flow range. Initially, no particle transport occurred. Video images indicate that saltation starts around profile 30, followed by ejection events around profile 50 and more general resuspension after profile 70. The ADVP detects the different resuspension phases and shows that during the unsteady flow range, particles are progressively resuspended to greater height which reaches up to about $0.3h$ ($h =$ waterdepth). Note that resuspension occurs in individual events even during the final phase of the unsteady flow. This agrees with the video images (Fig. 3). From Fig. 3 it can be seen that even at the end of the unsteady flow range, the particle concentration in suspension is low. The ADVP can track these low concentrations.

Total velocity vectors, combining the horizontal, $u$, and the vertical, $w$, velocity vectors for the range of profiles 80 to 90 (Fig. 2) were plotted in Fig. 4. The range of profiles 80 to 90 is characterized by two events of strong backscattering intensity. In the near bottom boundary layer, velocity vectors are mainly upwards oriented which can explain the resuspension of fine sediment during this period.
4. CONCLUSIONS

In this study, unsteady open-channel flow over a coarse bed with a fine sediment layer was investigated. Acoustic techniques were successfully applied to determine detailed current velocity and backscatter intensity profiles. Particle resuspension progressively intensified during the unsteady flow range. Even though the concentration of suspended particles was too low to reliably invert the backscattered intensity signal into particle concentration, the ADVP is sensitive enough to capture clean signals for the time history of the initial phase of sediment resuspension.
The combination of acoustical and optical methods provides for an ideal approach in studying resuspension in unsteady flow. An event structure in resuspension is seen by both methods. These results and further experiments which will be carried out refining the approach outlined in this paper provide valuable insight into the dynamics of fine sediment resuspension under unsteady flow conditions.

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Abstract: Multi-frequency acoustics can be used to measure profiles of suspended sediment with a wide range of sediment types and concentrations. Commercially available instruments generally employ the technique of transmitting multi-frequency, coincident acoustic beams into the water column and measuring the backscattered signal. By exploiting the particle size dependency of the backscatter response for each frequency, the mean particle size for a given range cell can be calculated. Once the particle size uncertainty has been resolved, sediment load may also be derived for each cell.

Typical instrumentation operates within the frequency range of 500 kHz to 5 MHz. At these frequencies, signal attenuation can become significant, especially with finer particle sizes and higher concentrations, and this imposes limits on the profiling range. Range limitations are also found for low concentrations, when there is insufficient backscatter to be detected by the instrument.

Using a mathematical model of the AQUAscat 1000 acoustic sediment profiler, we quantify key limitations of particle size and concentration that can be practically tested in a laboratory environment, using a specially constructed sediment tower, and discuss the results of the laboratory studies. We also present selected data sets from field deployments of the instrument to compare its performance in a practical setting with model predictions.

Keywords: Suspended sediment, acoustic backscatter, particle size
1. BACKGROUND

Hay [1] was among the first to describe monitoring instruments that measured acoustic backscatter from suspended sediment by transmitting a pulse of high frequency, directional, acoustic energy into the suspension, and analyzing the received signal to obtain a time series of scattered signal strength. Scattered signal strength is related to the amount of material in suspension. Thorne and Hanes [2] summarised the relationship between a measured backscatter signal $P$ and mass concentration $M$ as follows:

$$M = \left( \frac{P r \psi}{K K_t} \right)^2 \cdot e^{4r\alpha}$$

where $K_t$ is a system constant derived during calibration, $F_m$ is the sediment form function, $a_s$ the mean sediment radius and $\rho_s$ the sediment density.

Hay and Sheng [3] described a method of particle size determination using multi-frequency acoustic backscatter by using the dependence of the backscatter form function for suspended particles on $ka_s$, where $k$ is the wave number equal to $2\pi f/c$, and $c$ is speed of sound. Using frequencies $f$ of 1 MHz, 2.25 MHz and 5 MHz, they could estimate particle sizes in the range 50 $\mu$m to 170 $\mu$m to between 10% and 20% accuracy.

Aquatec began manufacturing multi-frequency acoustic backscatter systems in 1992. The AQUAscat 1000, launched in 2006 is the latest evolution of this technology. Up to four acoustic transducers are used at frequencies in the range 0.5 MHz to 5 MHz. Signal acquisition and processing is engineered to provide a stable, measurable gain and low noise, both of which aid in calibration. Now that such instrumentation is commercially available for the profiling of mean suspended sediment particle size and load, an inevitable question from potential users is 'Will it work in the suspended sediment regime that we are studying?' We now describe the practical limitations of acoustic signal measurement in this context, and the development and testing of a predictive tool to establish practical operating range limits for the AQUAscat 1000 acoustic suspended sediment profiler.

2. INSTRUMENT LIMITATIONS

Caine and Smerdon [4] outline factors affecting the performance of multi-frequency acoustic backscatter instruments. To establish system operating range limits, we need to account for measurement dynamic range, which limits measurable signal.

The amplitude of the measured backscatter pressure from a suspension of particles fits a Rayleigh distribution if $ka_s < 1$. For a practical measurement system, signal saturation determines the maximum signal amplitude that can be detected. From the Rayleigh distribution we know that we need to measure higher voltages than our final mean value. The saturation of these signals will cause an artificially lower mean value to be calculated. We therefore should aim to operate the system at a level that limits the effect of this saturation to within our required accuracy. Fig 1 illustrates how the Rayleigh distribution changes as the overall backscatter intensity increases. The amplitude axis has been scaled such that the range zero to one indicates the system’s operating range. Saturation occurs for amplitudes exceeding a value of one.
The lowest portion of the Rayleigh distribution will be lost due to the discrete sampling interval of the analogue to digital converter (ADC). Once the signal is less than one bit, it is effectively discarded and therefore can be modeled by setting the lower limit equal to the minimum detectable signal.

From [4] equation (2a) gives the mean of the ideal distribution and equation (2b) takes into account the effect of the saturation and the minimum signal. Fig 2 is a plot illustrating how the mean measured by the instrument is lower than the theoretical value. A measurement system using a 16 bit ADC will generate count values from 0 to 65535. The minimum detectable signal is 1 and the saturation point is 65535.

Fig 2 shows the error in the actual mean (2b) compared to the ideal situation given by equation (2a) for the small and larger signal ranges. It can be seen that operating the system with a mean value between 8 and 32000 results in less than 1% error in the mean of the distribution. This gives a 72 dB usable dynamic range.

\[
\mu_{\text{ideal}} = \xi \cdot \frac{\sqrt{\pi}}{2} \quad \text{where } \mu \text{ is the mean of the distribution} \tag{2a}
\]

\[
\mu_{\text{actual}} = \int_{A_{\text{sat}}}^{A_{\text{MAX}}} \frac{A}{\xi^2} e^{-\frac{A^2}{2\xi^2}} dA + A_{\text{SAT}} \cdot \int_{A_{\text{sat}}}^{0} \frac{A}{\xi^2} e^{-\frac{A^2}{2\xi^2}} dA \tag{2b}
\]

Fig 2: Error in the calculated mean signal due to system min and max limits

3. DEVELOPMENT OF A MODEL

To assess the performance of a practical system, a model was developed whose structure is shown in Fig 3.
The model outputs usable operating range contours for each frequency as a function of mean particle size and concentration, as well as multi-frequency operating range contours for different numbers of frequencies. Inputs are as follows:

- The instrument is characterised by frequencies $f$, effective transducer radii (allowing calculation of $\psi$), and calibration constants $K_t$, for each channel.
- Environmental parameters of temperature, depth, salinity and pH affect acoustic absorption and the speed of sound, which is used to measure range.
- The sediment suspension is currently described by material density and a standard deviation that assumes a normal or log-normal size distribution. A range of mean particle sizes and a range of concentrations is input.
- For the purpose of simulation, it is possible to set the size of range bin, and the number of range bins to be evaluated.

The model uses simplified Form Function and Scattering Cross Section estimates and size distribution modifiers obtained on a heuristic basis by Thorne & Meral [5], which were found to be sufficiently accurate to describe both normal and log-normal particle size distributions. Holdaway et al [6] summarised sediment attenuation equations taking account of both viscous absorption and rigid particle scattering, which are also used in the model.

**4. MODEL RESULTS**

The model was run with parameters of a previously calibrated instrument to establish theoretical minimum and maximum range limits at various range cell sizes, with typical results below.

Fig 4 shows a plot for range limitation due to saturation. The contours, in metres, show that saturation is rarely likely to be a problem with this instrument, as it would only occur very close to the transducer in very high concentrations of sand.

**Fig 3: Structure of AQUAscat model**

Fig 4: Typical saturation range contours for above parameters
4.1. Minimum Signal Limits

Fig 5 shows a typical set of plots for range limitation due to lack of signal. In these plots, only ranges to 256 cm (the maximum instrument range with 1 cm range cells) were evaluated. In this case, the plots for each frequency show that low concentrations of fine particles are not detected at lower frequencies, such as the 0.5 MHz channel, while the maximum range at 4 MHz is limited to less than the instrument’s maximum range. The former is explained by the lower scattering cross-section of finer particles at lower frequencies, while the latter is due to the increased absorption of higher frequency sound.

![Fig 5: Typical minimum signal range contour plot for each frequency](image)

![Fig 6: Typical minimum multi-frequency range contour plot](image)
By analysing the maximum achievable range for any 1, 2, 3 or all frequencies, a similar set of contours can be derived showing multi-frequency operating limits. Fig 6 shows a set of plots for multi-frequency range limits. In all these plots, ranges to 1024 cm (the maximum instrument range with 4 cm range cells) were evaluated. In this case, 4-frequency operation is limited to around 2 m range for sand, primarily due to the attenuation of the 4 MHz signal, while 2-frequency operation is possible over a range of concentrations and particle sizes at up to 10 m range.

5. LABORATORY EXPERIMENTS

Data were collected in Aquatec’s calibration tank, which is cylindrical with a diameter of 0.40 m and a depth of 2.3 m, giving a total volume of 280 l. A peristaltic pump provides a recirculation mechanism for the water volume and turbulence is generated near the inlet point by an impeller to aid the mixing process. The outlet is connected to a funnel on the bottom of the tank and water is injected through two inlet pipes near the top of the water volume. The inlet pipes direct the flow upwards onto a horizontal deflection surface through which the transducers are mounted. The AQUAscat was deployed with four transducers at 0.5, 1.0, 4.0 and 5.0 MHz and the bin size was set to a nominal value of 1 cm (based on a sound speed of 1500 m/s). The ping rate was set to 4 Hz and the instrument averaged data (arithmetic mean) over 4 pings before being written to disk, giving a stored profile rate of 1 Hz. Ballotini glass beads used for the experiments were analysed using a Malvern Mastersizer 2000, giving a D50 value of 172 µm with a standard deviation of 0.205 times the D50 value. The beads were introduced into the tank to give concentrations of 0.01 0.05, 0.2, 0.5, 0.75 and 1.00 g/l, based on the volume of water in the tank and the mass of the beads introduced. After each new quantity of beads had been mixed through the volume, recordings were made with the AQUAscat over a period of 30 minutes, giving 1800 profiles (corresponding to 7200 pings). Direct samples were siphoned from 0.6 m below the transducers at the end of the experiment.

Fig 7 shows the results from both the model and the tank experiment. The four solid lines represent the range at which the modelled received voltage at the transducers drops below a threshold of 20 counts. The blue line represents an estimate of the maximum range achievable with all four transducers deployed. Below approximately 0.13 g/l the range is limited by the low level of the modelled backscatter received by the 0.5 MHz transducer, where the wavelength is much greater than the size of the average scatterer and the backscatter is within the Rayleigh regime. Beyond approximately 0.13 g/l the range is limited further by the attenuation and absorption of the 5 MHz signal. The red line represents the estimate of the maximum range that can be achieved with different combinations of three of the four available transducers: Below approximately 0.13 g/l the maximum range is achieved with by choosing the three higher frequency transducers and is limited by the attenuation and absorption of the 5.0 MHz signal; Above approximately 0.13 g/l the maximum range is achieved by choosing the three lower frequency transducers and is limited below approximately 0.3 g/l by the low backscatter strength at 0.5 MHz and above approximately 0.3 g/l by the attenuation and absorption of the 4 MHz signal. The green line represents the maximum range achievable using combinations of two transducers and the yellow line represents the maximum achievable with just one transducer (where concentration can be derived from a direct sample using the explicit type of inversion).
Three 2 litre samples were siphoned from a point 0.6 m below the transducers after the last recording was made with the nominal concentration of 1.0 g/l. A mean concentration of 0.87 g/l (standard deviation=0.0244 g/l) was derived after the samples were filtered, dried and weighed. This is lower than the 1.0 g/l expected due to the mass of beads introduced to the volume of water. This is consistent with other experiments undertaken in the calibration tank and is caused by an accumulation of material in the recirculation system owing to the high settling velocities in comparison to the flow rate of the pump. The estimated mass concentration for each of the different quantities of beads was therefore derived by multiplying the nominal value by a factor of 0.87. Recordings of the background noise in the tank were made with the transmit power turned off, giving a mean value of 7.6 counts. The maximum range for each transducer (corresponding to the model results) was therefore determined by the range at which the time-averaged voltage dropped below 27.6 counts (as the model does not include noise). The results are shown in Fig 7. The dashed lines were derived from the maximum range achievable with combinations of four, three, two and one of the four transducers (The colours correspond to the plots derived from the model with the same number of transducers).

6. FIELD DATA

Fig 8 (inset) shows the predicted maximum range of the AQUAescat for a deployment using three transducers at 1MHz, 2MHz and 4MHz. The model assumes the sediment is sand with standard deviation 0.3 times the mean particle size. Range is determined as the point where the modelled voltage at the transducers has fallen to a level of 30 counts. The point highlighted on the plot represents the data collected in an estuary with a suspended load of medium silt (mostly 4-8 µm particle radius with typical concentrations of around 200-250 mg/l). The model predicts a maximum range of 2.37m at the point shown in Fig 9 and the measured range of the time-averaged voltage received at the field site is 2.32m (range determined by a threshold of 30 counts above the mean noise level).

Fig 8 (main plot) shows the levels of the transducer voltages (corrected for spreading, and attenuation) recorded over a 30 minute period at the field site. Data below the bed has been removed as has data below a threshold set just above the noise level. The range at which the 4 MHz signal is truncated can clearly be seen in the third plot.
7. CONCLUSIONS AND FURTHER WORK

We have described a tool for the theoretical prediction of performance of a commercially available multi-frequency acoustic sediment profiler. Comparisons with laboratory and field data show promising performance, and highlight the main areas of applicability for measurement of suspended sediment using high frequency acoustics.

The results of various model runs have provided insight into the performance of the instrument’s acoustic calibration process, and will be used to optimise the characteristics of the calibration tank and calibration procedures. They will also provide useful direction for improving the instrument’s main operating characteristics. Further work is required to model other suspensions including flocculation, bimodal distributions, and mud interfaces.

REFERENCES

Suspended Particles and Acoustic Backscattering in Flow and Sediment Measurements using Acoustic Doppler Velocimeters.

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Abstract: Acoustic Doppler velocimeters (ADV) have been extensively used to measure flow dynamics in the benthic and surface boundary layers in both oceanic and coastal environments. By definition the quality of the measurements depends on the amount of scattering material in the water column. Excessive amount of scattering material attenuates the signal and deteriorates the quality of flow measurements. On the other hand too little scattering material and the acoustic backscattering become insufficient to provide reliable flow measurements. At the same time, a number of investigators (e.g., Fugate and Friedrichs, 2002; Voulgaris and Meyers, 2004) have been combining the intensity of the backscattering signal with the acoustic turbulence flow measurements to estimate particle fluxes using the Reynolds flux approach and thus determine the settling velocity of the sediment. This approach assumes that the flow and sediment measurements are independent something that is not really true. This contribution consists of two parts. In the first part, I will use theoretical estimates of acoustic propagation and attenuation by particles to quantitative define the maximum concentrations that ADVs can be used to carry out flow measurements. Various concentrations of single particle sizes as well as distributions of particles are used. In the second part the accuracy of the Reynolds flux method for estimating particle settling velocities will be evaluated as function of acoustic frequency and particle size. The results provide useful applicable limits of maximum concentration per particle size and per ADV frequency that ADVs can be used as well as error estimates in sediment settling velocities.
Abstract: We propose an acoustic sediment model designed to predict velocity and attenuation of compressional- and shear waves as a function of frequency and depth in a wide range of soft marine sediments. The model uses the concept of a simplified sediment structure, modeled as a binary grain-size sphere pack. The acoustic response is formulated using Biot’s poroelastic theory as the general framework that we have extended by two viscoelastic model components. These extensions describe the mechanisms that we consider to have the most significant influence on wave propagation through soft sediment. An effective-grain model considers a viscoelastic response arising from local fluid flow if expandable clay minerals are present in the sediment, resulting in complex and frequency-dependent bulk- and shear moduli of the grain material. A heuristically modified Hertz-Mindlin/Walton based viscoelastic-contact model describes local fluid flow at the grain contacts, resulting in complex and frequency-dependent bulk- and shear moduli of the sediment frame. Porosity, density and the structural Biot parameters (permeability, pore size, structure factor) as a function of pressure follow from the binary grain-size sphere-pack model. Therefore, the remaining input parameters to the seismic/acoustic model consist solely of the effective pressure (or depth) in the sediment column, the mass fractions and the known mechanical properties of each mineral constituent (e.g., carbonates, sand, clay, expandable clay), and the environmental parameters (water depth, salinity, temperature). Compressional- and shear-wave velocities and attenuation predicted by our acoustic model are in good agreement with experimental data on coarse-grained and fine-grained unconsolidated marine sediment.
Theoretical developments on sound wave and shear wave propagation in marine sediments

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Abstract: An unconsolidated marine sediment supports the propagation of longitudinal (sound) waves and transverse (shear) waves. For each of these wave types, the phase speed tends to increase with the grain size, typically spanning the range 20 to 150 m/s (shear) and 1450 to 1800 m/s (sound). In both cases, the attenuation scales essentially as the first power of frequency over an extended spectral range. As the classical theory of Biot does not accommodate this first-power dependence, an alternative approach was developed recently, designated the Grain-Shearing or GS theory, in which the primary dissipation mechanism is associated with grain-to-grain interactions and, in particular, a non-linear phenomenon known as strain hardening. The GS theory yields two sets of dispersion relations, one for sound and the other for shear, which match the available data on phase speed and attenuation as functions of frequency, at least for frequencies above about 10 kHz. In addition, the GS theoretical expressions predict correctly the variation of the phase speeds and attenuations with material parameters such as porosity, density and grain size. At frequencies below 10 kHz, some evidence, notably from the ONR-supported SAX99 and SAX04 experiments in the Gulf of Mexico, indicates that the GS dispersion relations show minor departures from the experimental data. In an effort to ameliorate this problem a modification to the GS theory was introduced recently: known as the Viscous Grain Shearing (VGS) theory, it allows for viscous saturation of the molecularly thin layer of pore fluid between grain contacts, and yields dispersion curves that fit the SAX data at lower frequencies, below 10 kHz, as well as essentially all the available data above 10 kHz. In both the GS and VGS theories, the relationship between porosity and grain size plays a role. Observational evidence indicates that the porosity tends to increase with decreasing grain size. Such behavior does not occur with smooth, uniform spheres when they form either regular or random packing structures. In general, sand grains are not smooth spheres, leading to the suggestion that their irregular shape may be responsible for the higher porosity of finer-grain material. An analysis of grain shapes has recently been performed, using a computer-controlled optical microscope. Although this investigation has not yet
resolved the porosity issue, it has shown that sand grains of various origins, including beaches, shallow sediments and desert dunes, all possess essentially the same radial spectrum. Based on this observation, an algorithm has been developed for simulating random particle shapes which have the same statistical properties as the shapes of real sand grains. The next stage is to use this computational tool as a component of a numerical technique for packing together highly irregular particles, with a view to investigating porosity as a function of grain size through mathematical simulation. (Research supported by the Office of Naval Research).
A SENSITIVITY ANALYSIS OF PARAMETER ESTIMATES IN BUCKINGHAM'S GRAIN SHEARING MODEL

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Abstract: A compilation of 54 sets of geoacoustic measurements of marine sediments was reviewed in order to bound estimates of parameters for use in Buckingham's grain shearing (GS) theory of acoustic propagation. These data, for unconsolidated sands (siliciclastic and carbonate) with grain sizes between 0.0156 mm and 0.57 mm, were all made in shallow water (maximum depth of 60 m) sites at diverse locales. In each data set, measurements of the speed and attenuation of the compressional wave at high frequency, the shear wave speed at 1 kHz, porosity, and bulk density are sufficient to calculate the three free parameters in GS theory (a material exponent, a compressional coefficient and a shear coefficient). The spread of the values calculated for GS parameters, combined with a sensitivity analysis, do not support the use of a single material exponent value for all sediments. Nor do they support the notion that the material exponent be 1, which would be the case if elastic and viscous forces at grain contacts are equal. Finally, it is suggested that, given the sensitivity to spot measurements, the GS parameters be estimated simultaneously from a full range of compressional and shear wave measurements.

Keywords: geoacoustics
1. INTRODUCTION

Buckingham's Grain Shearing (GS) theory was introduced[1] as an alternative to poroelastic theories, such as Biot's[2], in an attempt to better match observed dispersion curves. The foundation of this alternative theory, that rigidity in the sediment is provided by grain-to-grain shearing, removes the necessity of describing an elastic frame, an artificial construct which must be described by unobservable elastic parameters. However, GS theory introduces three parameters, which are not in themselves observable, but are derived from other physical observations.

GS theory describes stick-slip events at grain contacts, which, rather than being truly impulsive, have, on average, some exponential decay, parameterized by a material exponent $n$. Compressional and shear waves propagate via these grain contacts with relative intensity parameterized by two elastic moduli, $\gamma_p$ and $\gamma_s$ respectively. These are not elastic moduli relating stress and strain rates, but are related to the rate and intensity of radial and translational shear events at grain contacts. Not directly measurable, they are derived from observed quantities.

Given a known porosity $N$, GS theory requires estimation of three unknown parameters ($n$, $\gamma_p$ and $\gamma_s$). Buckingham has described a method of calculating these three values from measurements of compressional and shear wave speeds and attenuations[3]. First compressional sound speed and attenuation at a high frequency is used to calculate the material exponent $n$. Then the shear speed at a low frequency is used to calculate both moduli, $\gamma_p$ and $\gamma_s$. This method avoids use of the shear wave attenuation, a usually imprecise measurement seldom available.

2. GEOACOUSTIC DATA SET

The geoaoustic measurements were all made in situ using various versions of the In Situ Sediment geoAcoustic Measurement System (ISSAMS). The measurement system is described in Chapter 5 of Jackson and Richardson [4] and most of the geoaoustic and physical property data used in these analyses can also be found in that reference. The data base can be traced back to the original sources using references in [4]. All in situ geoaoustic measurements were made at 20-30 cm below the sediment-water interface over path lengths ranging from 30-100 cm. Compressional wave speed and attenuation were measure at either 38 or 58 kHz using time-of-flight and amplitude of a 5- to 10-cycle pulsed sine waves propagating between identical radially-poled ceramic cylinders through sediment and a reference of seawater just about the sediment water interface. Shear wave speed was measured at 1 kHz using time-of-flight between bimorph ceramic benders mounted on the same diver-deployed or remotely-operated hydraulic systems. Shear wave attenuation was measured at selected sites using a 4-transducer transposition technique which calculates shear wave attenuation from waveform amplitudes measured using two transmit and two receiver transducers. This technique eliminates the need to measure transducer sensitivity or measure variable insertion losses. Multiple sediment cores were collected from each site to measure high frequency (400 kHz) compressional wave speed and attenuation and to provide data of sediment physical properties such as grain size distribution, sediment bulk density and porosity. All values of wave speed and attenuation as well as bulk sediment physical properties used in these analyses are averages from multiple deployments and multiple sediment cores collected at the same location (a roughly 25-m$^2$ area).
The variability among measured wave speeds and attenuation at a single location is generally thought to be equal or greater than the actual measurement error which is less than 1% for wave speeds and less than 10% for attenuation.

For this study we restrict ourselves to data from the 54 sandy sites where the microscopic stress relaxation mechanisms that are part of the Buckingham theory are most likely to be applicable. It seems unlikely that this particle-to-particle stress relaxation mechanism can be applied to muddy sediment where electrostatic repulsive and attractive forces and the adhesion of organic matter control particle-to-particle interactions and the flexure of clay particles may provide a dissipation mechanism.

3. SENSITIVITY ANALYSIS

Errors in estimates of Buckingham’s parameters are driven by uncertainty in the measurements of the physical parameters from which they are calculated. Given the high correlation between porosity and grain size, only one need be included, and here it is porosity. Buckingham describes a process of calculating all parameters by first determining $c_0$ as a function of porosity $N$ from Wood’s equation[5], calculating the spectral exponent $n$ from $c_p$ and $\alpha_p$, and finally evaluating the GS moduli from the shear wave speed $c_s$. Hence, the error in estimating $n$ involves errors in $c_p$, $\alpha_p$ and $N$. Estimates of the GS moduli additionally involve errors in measuring $c_s$. Assuming the errors in measurements are Gaussian and uncorrelated, the overall error in each estimate is the sum of the contributions from individual sources.

$$\sigma^2_n = \left(\frac{\partial n}{\partial c_p}\right)^2 \sigma^2_{c_p} + \left(\frac{\partial n}{\partial \alpha_p}\right)^2 \sigma^2_{\alpha_p} + \left(\frac{\partial n}{\partial N}\right)^2 \sigma^2_N$$

$$\sigma^2_{c_p} = \left(\frac{\partial c_p}{\partial c_s}\right)^2 \sigma^2_{c_s} + \left(\frac{\partial c_p}{\partial \alpha_p}\right)^2 \sigma^2_{\alpha_p} + \left(\frac{\partial c_p}{\partial N}\right)^2 \sigma^2_N$$

$$\sigma^2_{\alpha_p} = \left(\frac{\partial \alpha_p}{\partial c_s}\right)^2 \sigma^2_{c_s} + \left(\frac{\partial \alpha_p}{\partial c_p}\right)^2 \sigma^2_{c_p} + \left(\frac{\partial \alpha_p}{\partial N}\right)^2 \sigma^2_N$$

Here $\sigma^2_\theta$ is the variance of the estimate of the parameter $\theta$. The partial derivatives given in equation (1) are given in the annex. Although the variance of the errors in the measurement of physical and geoacoustic parameters vary for each data set, for this analysis, it was assumed that the coefficient of variation in each measurement was the same for all data sets. Hence the following variances were assumed.
\[ \sigma_N = 0.02N \]
\[ \sigma_{c_p}^2 = 0.01c_p \]
\[ \sigma_{a_p}^2 = 0.30\alpha_p \]
\[ \sigma_{c_s}^2 = 0.10c_s \]  

(2)

4. RESULTS

The values calculated for the material exponent \( n \), compressional modulus \( \gamma_p \) and shear modulus \( \gamma_s \) are shown in Figs. 1 to 3, respectively. Fifteen data sets yielded negative values for \( n \), an unrealistic, yet mathematically possible result, and are hereafter omitted from the analysis. For each value calculated from the remaining 39 data sets, error bars span plus or minus one standard deviation about the estimate. Each parameter is plotted as a function of porosity. Carbonate sands are plotted in cyan, siliciclastic in blue, and individual SAX99 data in red. Values derived from SAX99 used by Buckingham are green[6].

![Fig. 1 – Material exponent versus porosity.](image-url)
Although there is a wide spread in uncertainty in the material exponent shown in Fig. 1, there appears to be a dependence on porosity. The bulk of the values are of magnitude much less than one. This contradicts the notion that elastic and viscous forces at the grain contacts should be considered of equal importance, at least within context of the GS theory.
The apparent increase in value for \( n \) with increasing porosity contradicts Buckingham's assertion that one value for \( n \) can be used for all sediments[7]. Porosity is the determining factor in evaluating the low frequency limit of compressional sound speed \( c_0 \), given by the Wood equation. \( c_0 \), in turn, is used in calculating \( n \). But there is obviously further dependence of \( n \) on porosity.

Assuming that estimates of porosity and compressional sound speed are accurate, and that inaccuracies in measuring compressional attenuation prevent accurate determination of \( n \), an attempt was made to find a single value of \( n \) that resulted in values of \( \alpha_p \) that is consistent with the entire data set. Specifically, if \( \alpha_p^h \) is the attenuation required to give a hypothetical material exponent \( n^h \) and \( \alpha_p^i \) is the measured attenuation from data set \( i \), then a residual error can be computed.

\[
\epsilon^i = \alpha_p^i - \alpha_p^h
\]

(3)

The value of \( n^h \) that minimized the mean square error for all data sets is \( n^h = .14 \). Adopting a lower value of \( n^h \) for all sediments (\( n^h = .0851 \), as Buckingham has suggested[3], implies that compressional attenuation used in the GS model is significantly lower than that observed in these data sets. However, this is consistent with his assertion that GS theory tends to give lower bounds on attenuation values, in that it accounts only for intrinsic attenuation[3]. Other sources, such as scattering of sound by large-scale inhomogeneities are not accounted for in this theory.

Both grain-shear moduli, plotted in Figs. 2 and 3, also show strong dependence on porosity. Note the errors are relatively greater for carbonate sands than siliciclastic. Also the error in the estimate of \( \gamma_s \) is higher than that for \( \gamma_p \). Generally, the largest source of error in the estimate of \( \gamma_p \) is due to errors in measurement of \( c_s \), while that for the estimate of \( \gamma_s \) is due to errors in measurement of \( \alpha_p \).

The uncertainties given in the above figures are calculated for the method of evaluating the three GS parameters specified by Buckingham. However, it is postulated that alternative methods may yield more accurate estimates. For instance, rather than sequentially determining \( n \), then the GS moduli, they may all be fit to observations simultaneously. The possible reduction in uncertainties in these estimates is discussed next.

5. BOUNDS ON ALTERNATIVE ESTIMATES OF BUCKINGHAM’S PARAMETERS

In a mapping \( G \) of a set of \( N_m \) model parameters to a set of observations \( d \),

\[
d = G(m)
\]

(4)

the covariance of the A Posteriori errors, \( C^*_{m} \), in the estimates of \( m \) can be calculated if all errors are assumed to be Gaussian and the mapping can be linearized about some estimate \( m_0 \). In this case[8],
\[ C_m' = \left( C_m + G_1^T (C_T + C_D)^{-1} G_1 \right)^{-1} \]  

Here, \( C_m \) is the covariance of the A Priori errors, \( C_D \) is the covariance of the data (measurement errors), \( C_T \) is the covariance of theoretical errors (due to mismatch between the forward model and reality), and \( G_1 \) is given by the linearization

\[ G_{ij} = \left. \frac{\partial g_i}{\partial m_j} \right| \]

In this case, \( m \) is the set of GS parameters \((n, \gamma_p, \gamma_s)\), \( d \) is the set of observations \((N, c_p, \alpha_p, c_s)\), the A Priori errors are given by the previous sensitivity analysis and the linearization provided by the partial derivatives previously derived. Measurement errors are assumed to be dominant, and theoretical errors are neglected.

In the case in which the three GS parameters are estimated simultaneously, the standard deviation of the error in estimating \( n \) (for the composite SAX99 data set) is only slightly reduced (by a factor of .95), while those for the two GS moduli are roughly halved. This is understandable, as Buckingham's sequential method uses the most reliable information to first calculate \( n \), and the GS moduli are based on a more error prone measurement of the shear wave speed.

Another alternative was investigated, to see if uncertainties can be reduced more significantly. A second set of compressional wave measurements from the SAX99 experiment, provided by the diver deployed "attenuation array" gives \( c_p \) and \( \alpha_p \) at 100 kHz[9]. Assuming similar variances in these estimates as with the 58 kHz data, the above error analysis again resulted in an insignificant decrease in the error in estimating \( n \), about a halving of the error in estimating \( \gamma_s \), but a reduction in the error in \( \gamma_p \) by a factor of .07.

### 6. CONCLUSIONS

For the GS theory to be validated, accurate and meaningful estimates of the three GS parameters must be achieved. It is not likely that this can be accomplished with field data. Laboratory conditions will likely provide better control over sediment properties and more accurate measurements. But this analysis of a large set of field observations, coupled with the sensitivity analysis, provides a first step in bounding the range of values that should be investigated.

The estimate of \( n \) is a critical first step and relies on accurate measurement of porosity, compressional sound speed and attenuation, all of which contribute significantly to the error. It is clear from this data set that a single value of \( n \) cannot be assumed for all sediments. But more accurate measurements are required in order to determine whether these GS parameters can be specified so that GS theory can be thoroughly validated.
7. ACKNOWLEDGEMENTS

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REFERENCES


ANNEXE

A1. PARTIAL DERIVATIVES

List of symbols

- $c_0$: compressional sound speed in equivalent suspension
- $c_p$: compressional wave speed in sediment
- $c_s$: shear wave speed in sediment
- $K_0$: Bulk modulus of equivalent suspension
- $K_g$: Bulk modulus of grain
- $K_w$: Bulk modulus of pore fluid
- $n$: Strain-hardening index
- $N$: Porosity
- $T$: Arbitrary time constant
- $\alpha_p$: Compressional attenuation coefficient
- $\chi$: Dimensionless grain-shearing coefficient
The partial derivatives of the Buckingham model parameters with respect to measured values are given below. For the sake of clarity, some derived parameters \((X, \chi)\) are given first. Then relevant expressions are given in terms of those derived parameters.

\[
\frac{\partial n}{\partial \kappa_p} = \frac{1}{\pi\kappa_p} \left[ \sin(n\pi) + 2X\sin^2\left(\frac{n\pi}{2}\right) \left[ 1 - \left(\frac{c_p}{c_o}\right)^2 \left(1 - X^2\right) X^2 \right] \right]
\]

\[
\frac{\partial n}{\partial \kappa_s} = \frac{1}{\pi\kappa_s} \left[ \sin(n\pi) + 2X\sin^2\left(\frac{n\pi}{2}\right) \left[ 1 + 2\left(\frac{c_p}{c_o}\right)^2 \left(1 + X^2\right) \right] \right]
\]

\[
\frac{\partial n}{\partial N} = \frac{1}{nX} \left[ \frac{\left(\frac{c_o}{c_p}\right)^2}{\sin(n\pi) + 2X\sin^2\left(\frac{n\pi}{2}\right)} \left[ 1 + 2\left(\frac{c_p}{c_o}\right)^2 \left(1 + X^2\right) \right] \right]
\]

\[
\frac{\partial \chi}{\partial n} = -\chi \log(\omega T) + \frac{n}{2} \cot\left(\frac{n\pi}{2}\right) \left[ 1 - \frac{\left(\frac{c_p}{c_o}\right)^2}{\sin(n\pi) + 2X\sin^2\left(\frac{n\pi}{2}\right)} \right] \left[ \frac{\left(\frac{c_p}{c_o}\right)^2}{\sin(n\pi) + 2X\sin^2\left(\frac{n\pi}{2}\right)} \left(\omega T\right)^n \csc\left(\frac{n\pi}{2}\right) \right] \frac{\partial \chi}{\partial n}
\]

\[
\frac{\partial \gamma_s}{\partial n} = -\frac{3}{4} \rho_0 c_s^2 \left(\omega T\right)^n \sin\left(\frac{n\pi}{2}\right) - \gamma_s \log(\omega T)
\]

\[
\frac{\partial \gamma_p}{\partial n} = \rho_0 c_o^2 \frac{\partial \gamma_{X}}{\partial n} - \frac{4}{3} \frac{\partial \gamma_s}{\partial n}
\]

\[
\frac{\partial \gamma_{X}}{\partial n} = 2 \frac{\gamma_{X}}{c_s}
\]

\[
\frac{\partial \gamma_p}{\partial \chi} = \frac{4}{3} \frac{\partial \gamma_s}{\partial \chi}
\]
Structured Session 14

Underwater Networks

Organizers: Wolfgang Jans & Kim McCoy
A ROADMAP TO UBIQUITOUS UNDERWATER ACOUSTIC COMMUNICATIONS AND NETWORKING

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Abstract: Underwater acoustic communication systems are of increasing importance to all nations and organization performing maritime/littoral activities. Currently, few standards exist for underwater communications, and modems from different vendors are in general incompatible. This leads to a lack in interoperability.

This paper points out a possible roadmap to standardization and flexible modems and protocols, arguing that in order to be successful in standardization of underwater communications, it is essential that standardized simulation models and methods are agreed upon first. The paper also discusses various efforts known to the authors, and their relation to the proposed roadmap.

Keywords: Underwater acoustic communications, underwater networks, standardization

1. INTRODUCTION

Underwater acoustic communication systems are of increasing importance to all nations and organizations performing maritime/littoral activities. As the situation is now, few standards exist for underwater communications, and modems from different vendors are in general incompatible. On the other hand, underwater communication technology is maturing to such a level that standardization should be possible within some years, and it should be possible to build flexible modems capable of handling different communication schemes.
Underwater acoustic communications is challenging due to a number of reasons: The bandwidth is limited due to strong absorption of high frequency sound, and the low speed of sound leads to significant multipath and Doppler effects as well as propagation delays. Techniques developed for radio therefore need significant modifications to be adapted to the underwater channel.

This paper proposes a possible roadmap to standardization and flexible modems and protocols, which are prerequisites to reaching a state of ubiquitous underwater acoustic communications and networking. While research in radio communications is mostly based on simulations using established models, research in underwater acoustic communications is mostly based on sea trials, which are expensive and time-consuming. In this paper we argue that in order to be successful in standardization of underwater communications, it is essential that standardized simulation models and methods are agreed upon first.

2. THE PROPOSED ROADMAP

A schematic overview of the proposed roadmap is given in Fig. 1. We wish to stress the point that in the middle column, the lower blocks should be standardized before the higher blocks. E.g., in order to be able to standardize physical layer waveforms, one should first have standard channel models to be able to compare different waveform proposals. In the following sections we describe each building block in more detail.

Fig.1: Schematic view of the proposed roadmap towards standardization of underwater acoustic communications.

2.1. Standardized channel models and channel simulators

The foundation should be standardized channel models and channel simulators. Methods for channel simulation should be defined in an unambiguous manner such that different
implementations will give the same results. Several test channels should be defined, representative of a wide range of scenarios which can then be used to define standards.

To be able to investigate and compare modem performances, the standardized channel simulator should model delay spread as well as Doppler spread in a realistic manner. Such a channel simulator can be implemented as a time-varying linear filter, but the question is how to define and generate the time-varying tap gains. Several different principles could be applied: (1) Methods based on replaying actually measured time-varying impulse responses, (2) methods based on generating channels with the same statistical properties as measured time-varying impulse responses, (3) methods based on a parameterized scattering function (e.g., a finite number of paths with defined delay and Doppler spreading of each path), (4) methods based on building a parameterized scattering function from raytracing simulations.

A number of test channels should be defined, representative of a wide range of scenarios: Vertical or horizontal links, shallow or deep water, surface interaction or not, short or long range, low or high frequency. Each test channel should be defined such that it can be implemented in the channel simulator. One possible inspiration could be [1], which defines a number of standard test channels for ionospheric radio communications (one commonality between underwater acoustic and ionospheric radio communications is that in both cases a wide range of channel conditions can be experienced due to different physical environments).

Selection of test channels and their model parameters should be based on a thorough study of available measurement results, complemented by new measurements.

One should also decide whether to include modem motion in the standard channel simulator. If so, this has to modelled on top of the acoustic channel model, since including modem motion in the channel model will severely distort the scattering function and violates the WSSUS (wide sense stationary uncorrelated scattering) assumption [2].

2.2. Standardized physical layer waveforms

The physical layer (modem waveforms) should be standardized in such a manner that modems from different vendors supporting the standard can communicate with each other. The physical layer includes modulation, forward error correction, interleaving, and frame format. The standard should specify a variety of parameter combinations, for different data rates and different scenarios as described above. The frame format should contain a header specifying the parameter combination used, such that it can be automatically determined by the receiver.

Different institutions and vendors should be given the opportunity to propose candidate waveforms for standardization, and the different proposals should then be compared on a fair basis using the standardized channel simulator and test channels (e.g., varying the SNR of the test channel to see which proposal requires the lowest SNR for a certain bit error rate).

Benefits of successful standardization of physical layer waveforms would include: (1) A user could buy modems from different vendors and use them together. (2) Different users operating in the same area could communicate acoustically with each other if they use the same frequency range. (3) Increased competition would force the vendors to make their modems as good, cheap and power-efficient as possible. (4) Modems from different vendors could operate in the same frequency band if they all used standardized medium access control.
2.3. Flexible modems

Modem vendors should ideally produce modems that are as flexible as possible, capable of implementing a wide range of physical layer waveforms. Flexible modems range from reconfigurable modems, able to choose among a finite set of modulation types with defined parameter sets, to fully reprogrammable modems able to implement any modulation and demodulation scheme by simple means of importing new software (based on open systems accessible to the user).

A vendor taking this approach would in principle ensure that he is able to implement whatever waveform is standardized at a later point, or if standardization is unsuccessful be able to implement waveforms used by other vendors such that the systems would be interoperable anyway (by agreement between vendors, i.e., de facto standardization).

In terms of standardization, common and flexible software interfaces should be defined. Here, one may look to ideas from “software radio” [3].

2.4. Standardized simulation models and tools for protocols (MAC and above)

Once physical layer waveforms are standardized, the next step is to standardize higher layer protocols for systems with multiple modems in use at the same time. Also at the higher layers, standardized simulation methods and models are important to be able to compare different ideas and proposals.

Protocol and network simulations are in general based on discrete event network simulators. Example network simulator frameworks are [4]: ns-2, J-sim, and OMNet++, which are all open-source, and OPNET, which is a commercial product.

When a communication network running a particular protocol is to be simulated, one would typically model a deployment scenario and implement the protocol into the simulator framework. A set of standard deployment scenarios should be defined.

In the setup of the network simulator, one would need to invoke some model of the modem and channel for each link in the network to determine the bit error rate for that link. Here, one option would be simulate the modem waveform over a relevant test channel (preferably standardized) to find its bit error rate vs SNR curve. Then, SNR vs range can be computed based on a path loss model and a background noise model. The path loss model could be a simplified attenuation and spreading model [5], or a simulation of path loss for the particular modem depths used, range, and sound speed profile, using e.g. Bellhop [6]. The background noise model could be based on the Wenz curves.

Another issue that needs to be addressed, is how to model collisions, i.e., what happens when two incoming packets arrive at a receiver so close in time that they interfere.

2.5. Standardized and flexible protocols

In the long term, protocols (medium access control, link control, routing, etc) should also be standardized based on a fair comparison of proposals using standardized simulators. Ideally, it should be possible to use similar software code both for simulation and in modem implementation, but this may prove hard in practice. As for the physical layer, a number of protocol variants should be standardized, suitable for different conditions and scenarios.
We would like to stress that trying to standardize protocols without first standardizing the building blocks below it in Fig. 1 would probably lead to much frustration.

2.6. Frequency allocations and band plans

One of the many differences between radio and underwater acoustic communications is that the underwater acoustic spectrum is unregulated. As use of the spectrum increases, it would be of great value if band plans and emission spectrum masks were standardized also for the underwater communication environment. This important topic has not been considered further for this paper.

3. SELECTED ONGOING EFFORTS

In this section we discuss various efforts known to the authors, and their relation to the proposed roadmap. Note that this list is not intended to cover everything happening in this field, and it is probably flavoured by the authors’ position in a Defence Research Establishment in Europe. For a comprehensive review of current research in underwater acoustic communications and networks, we refer to the survey papers [7, 8].

3.1. The UCAC project

The UCAC (Underwater Covert Acoustic Communications) project was running in 2005-08, with partners from Denmark, Finland, Germany, Italy, the Netherlands, Norway, and Sweden. The main goal was to develop covert methods for long-range communication, which in essence means communicating at very low SNR. One important outcome of the project for the topic of this paper, is the development of a Matlab-based channel simulator covering at least 2 of the 4 channel simulation methods mentioned in Section 2.1 [2].

If made available to the international community, the UCAC channel simulator could serve as a starting point for a standard channel simulator.

3.2. The 80 bps WHOI standard

WHOI (Woods Hole Oceanographic Institution) has defined a standard physical layer waveform operating at an information data rate of 40 or 80 bps [9]. The modulation format used is binary FSK (frequency shift keying) combined with frequency hopping, and a rate-1/2 error-correcting code with interleaver is applied. This standard is supported by the WHOI micromodem and by modems from Teledyne Benthos. This is one of the few standards already existing for underwater acoustic communications, but the data rate is relatively low.

3.3. The JANUS initiative

NURC (NATO Undersea Research Center) has taken an initiative for standardization called JANUS [10]. Two workshops have been arranged in 2008 and 2009, with
participation from industry, government institutions, and academia. The outcome this far is a proposed standard “first-contact” waveform and header format, which has been described as a “VHF channel 16” for digital underwater acoustic communications, and which also has been suggested for use as a beacon. The physical layer waveform is in many respects inspired by the WHOI standard, but is defined such that the data rate is proportional to the center frequency (such that the fraction bandwidth/center frequency is independent of center frequency). The header is among other things intended to contain information on which capabilities the transmitting modem has, in addition to supporting JANUS.

The JANUS waveform has low data rate, and has proven itself robust in several sea trials carried out in 2008. However, the waveform has not been systematically tested on a channel simulator, and thus the process has bypassed the first step in the roadmap proposed in this paper. A NURC report describing the JANUS waveform is intended to be released in 2009.

3.4. Activities at MIT

MIT (Massachusetts Institute of Technology) is developing a flexible modem called “r-modem” (reconfigurable modem) [11]. The modem is intended to be used in research to develop and test different physical layer methods using rapid prototyping with tools from The Mathworks (Simulink and Real Time Workshop). Such a modem will be a valuable tool in the development of physical layer waveforms and scores high on flexibility.

MIT has also developed an underwater acoustic network simulator called AUVNetSim [12]. It is written in Python and makes use of the SimPy discrete event simulation package. AUVNetSim is distributed as open source, and is one of several possible starting points for standardized protocol simulators.

3.5. The PLUSNet program and CCL

The PLUSNet (Persistent Littoral Undersea Surveillance Network) program [13] is a US initiative funded by ONR (Office of Naval Research). One long-term goal is to provide an autonomous adaptive communications network. To have the means to achieve this goal, an extensible and open architecture is applied [13]. This may serve as a good foundation for future standardization. The PLUSNet system currently uses WHOI Micromodems at the physical layer, and has developed a message format called CCL (Compact Control Language) at the application layer [14]. CCL comprises a large and well-defined set of 32-byte messages designed to control AUVs (Autonomous Underwater Vehicles).

3.6. The UNA initiative

Further efforts to provide an architecture framework for underwater acoustic networks have been made in the UNA (Underwater Network Architecture) initiative [15], with participation by ARL (Acoustic Research Laboratory at National University of Singapore), WHOI, and MIT. UNA is loosely based on the OSI model, and may provide a good foundation for future standardization of underwater communication systems and networks.
3.7. The Seaweb program

Another ONR-funded program for underwater communications networks is Seaweb [16], which has been run by SSC/PAC (Spawar Systems Center Pacific) and NPS (Naval Postgraduate School) with Teledyne Benthos as the main contractor. Link and networking protocols have been developed, and fielded in numerous sea trials over the years. It is therefore well proven that the Seaweb system is working in several real-life scenarios.

If the Seaweb specification or parts of it was released and made available to the international community, it could serve as a good starting point for standardization of protocols.

3.8. The UAN project

UAN (Underwater Acoustics Network) is a project funded by the European Community's Seventh Framework Programme (FP7), with participation from Italy, Norway, Portugal, and Sweden. Objectives include “to implement a generic underwater ad-hoc mobile acoustic network (MANET) composed by fixed and mobile nodes”, which “must be reconfigurable and scalable”. Outcomes from this project may be relevant to the standardization of communication protocols. The project is still in an early phase.

3.9. NARCISSUS-2005

NARCISSUS-2005 [17] is a channel simulator developed by Thales Underwater Systems. It is a model based on raytracing, wave motion, and modem motion, and produces time-varying impulse responses. If made available to the international community, it could serve as a starting point for a standard channel simulator.

3.10. The planned RACUN project

RACUN (Robust Acoustic Communications in Underwater Networks) is a planned project under EDA (European Defence Agency). The project is likely to have partners from 5 European countries, and will start no earlier than in 2010. The overall goal is to develop and demonstrate the capability to establish an underwater ad hoc robust acoustic network for multiple purposes with moving and stationary nodes. The draft project description includes several secondary goals which are relevant to the framework in this paper: Improve models of the underwater acoustic communication channel, model the link and network layers, evaluate network protocols based on simulations, make channel and network models generic such that they can be used as a test bench for different systems, and define common and flexible software interfaces for modem(s) and models.

4. DISCUSSION AND CONCLUSIONS

The roadmap presented in this paper is a proposal on how one should proceed in standardization of underwater communications. The process is going to take time, but
bypassing some step in the process is not likely to be cost-efficient in the long term. Many relevant efforts are being made, but it seems clear that a large and combined effort of several interested parties and vendors is required, and coordination is essential. It should, however, be possible to split the work into several groups. The question remains who should be in charge of this process, e.g., whether it should be driven from the military side or from the civilian side. On the civilian side the market is larger, but on the military side one might find it easier to focus efforts.

In radio communications, one major contributing factor to successful standardization has been the existence of standardized simulation models, which can be used for fair comparison of different standardization proposals. To be successful also in standardization of underwater acoustic communications, it is of great importance to first focus on standardizing simulation models, since sea tests are not suitable for comparing different standardization proposals.

REFERENCES

STANDARDISED TESTING OF ACOUSTIC COMMUNICATIONS MODEMS USING A HIGH FIDELITY SIMULATOR

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Abstract: The performance of underwater Acoustic Communications (AComms) systems is highly dependent on the properties of the sound channel and is affected by many factors including reflections from the sea surface and seabed, scattering and Doppler. This paper describes a high fidelity Acoustic Communications Simulator (ACS) which has been developed by the UK Ministry of Defence (MoD) and used to evaluate the performance of underwater AComms systems in multiple global environments. The model simulates the transmission of a broadband or narrowband AComms signal through a dynamic underwater sound channel. Doppler shifts due to platform motion, moving sea surfaces and internal currents are modelled using a time dependent representation of the environment. The received signal can be calculated at individual hydrophones within an array or at the output of a beam steered towards the transmitter. Ambient noise and self-noise are modelled as Gaussian random processes. Modem hardware can be utilised to modulate the message data and to demodulate the signals received after transmission through the simulated channel. This allows close-loop testing to be conducted to evaluate modem performance.

Keywords: Acoustic communications prediction and performance
1. INTRODUCTION

Digital Acoustic Communications (AComms) technology is being developed to provide a reliable method for transmitting data between underwater systems to support a variety of military and commercial applications.

The UK MoD has developed a high fidelity acoustic communications simulator in order to assess the performance of different underwater communications systems and signalling schemes under realistic operating and environmental conditions. The simulator has been used to carry out extensive, standardised performance assessments of different AComms systems in multiple global environments. Standardised testing has been conducted to evaluate performance for different transmit frequencies and bandwidths, transmitter/receiver options, platform configurations and types of encryption. Using the simulator for performance assessment of AComms systems has allowed the UK to greatly de-risk and optimise subsequent trials activity.

This paper provides an overview of the simulator and defines the characteristics of two standardised underwater sound channels. Example results are presented which show the performance of a spread spectrum signalling scheme in the defined channels.

2. ACOUSTIC COMMUNICATIONS SIMULATOR

The Acoustic Communications Simulator (ACS) has been developed by QinetiQ for the UK Defence Technology and Innovation Centre (DTIC) Research programme to evaluate the performance of acoustic modems under a wide range of operating conditions and environments. The system simulates the transmission and reception of an acoustic signal through a dynamic underwater sound channel. The model is designed to provide a detailed representation of the multipath, Doppler shift and scattering processes that occur during propagation of an acoustic signal. In particular, the model includes a time-dependent representation of the channel that allows Doppler shifts due to platform motion and moving sea surfaces to be modelled.

The principal functionality of the simulation system is shown in Fig. 1. An important feature of the simulator is that signals can be modulated and demodulated using modem hardware thereby allowing closed-loop testing to be conducted. ACS is linked to a number of historical databases which provide global and seasonal data for sound speed profiles and wind speeds. ACS is also linked to global databases for bathymetry and seabed parameters. The transmitter and receiver can follow arbitrary tracks in position and depth. The propagation model generates eigen-rays between the transmitter and receiver at regular intervals throughout the transmission. The eigen-rays are subsequently interpolated at each time sample to determine the time variant channel transfer function. This allows Doppler shifts due to platform motion and moving sea surfaces to be inherently included within the model.

The propagation model does not attempt to intrinsically model the multiple scattered reflections that occur from a moving sea surface. Instead signals reflected from a moving sea surface are represented as a complex valued Gaussian random process with zero mean. Alternatively the sea surface can be modelled as a flat stationary surface or a flat surface that moves vertically with sinusoidal motion. Attenuation due to scattering of energy at the sea surface is represented using the Beckman and Spizzichino model [1]. The ocean bottom is modelled as a three layer system comprising a thin sediment layer, a fluid sediment layer with depth dependent sound speed and attenuation but constant density, and a basement.
represented solely by a reflection coefficient. Spreading loss is modelled by calculating the cross sectional area of an infinitesimal ray tube. Absorption is calculated in the water column according to the Francois and Garrison model [2].

The received time series signals can be calculated at a single hydrophone receiver or at an arbitrary configuration of array in the presence of noise. Gaussian noise, simulated ambient noise or measured noise can be added to the received signal at different signal-to-noise ratios (SNRs) before attempting to demodulate the signal. The error rate is calculated by comparing the demodulated message data to the known input data. Multiple messages are transmitted through the channel in order to calculate the error rate to a statistically significant level. The simulator enables a variety of performance metrics to be calculated for each signal type including the SNR at the receiver, the bit error probability, the probability of decoding the message without error and the signal excess.

![Diagram of AComms simulation system]

**Fig. 1: Principal functionality of AComms simulation system.**

The propagation model has been validated by comparing transmission loss predictions with other models including Hodgson. Model predictions are also in quantitative agreement with experimental results obtained during testing of a prototype AComms modem at AUTEC.

3. **STANDARDISED TESTING**

The simulator is designed to conduct standardized testing of AComms modems in which performance is evaluated under consistent conditions in a predetermined manner. Standardized testing provides a number of benefits including the ability to compare the performance of different modems under identical conditions and the ability to determine the effect of specific environmental and operating factors on performance such as receiver and transmitter locations and speeds, wind speeds and wave heights. Three example channels are considered in this paper:  

- **a)** an idealised Additive White Gaussian Noise (AWGN) channel in which Gaussian noise is added to an exact replica of the transmitted signal;  
- **b)** the Gulf of Oman in January, a deep water (3300m depth) environment characterised by a relatively simple sound channel;  
- **c)** the Gulf of Hormuz in January, a shallow water (90m depth) environment characterised by complex propagation

The deep water and shallow water environments are modelled as time variant channels in which fading is imposed by motion of the transmitter, receiver and sea surface. The principal parameters of the example sound channels are listed in table 1. The receiver moves directly towards the transmitter at a speed of 2.5m/s. The seabed is modelled as an
acoustic half-space with parameters extracted from the WADER database produced by Ocean Acoustics Developments Ltd. The environmental parameters for the example channels are assumed to be range independent.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Deep water channel</th>
<th>Shallow water channel</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Environment</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Location</td>
<td>Gulf of Oman</td>
<td>Strait of Hormuz</td>
</tr>
<tr>
<td>Time of year</td>
<td>January</td>
<td>January</td>
</tr>
<tr>
<td>Water depth [m]</td>
<td>3300</td>
<td>90</td>
</tr>
<tr>
<td>Sea state</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Wind speed [m/s]</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>Wave height peak-peak [m]</td>
<td>0.5</td>
<td>0.5</td>
</tr>
<tr>
<td>Wave period [s]</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>Temperature at sea surface [°C]</td>
<td>24</td>
<td>24</td>
</tr>
<tr>
<td><strong>Transmitter</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Type</td>
<td>Omnidirectional</td>
<td>Omnidirectional</td>
</tr>
<tr>
<td>Depth [m]</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Speed [m/s]</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td><strong>Receiver</strong></td>
<td></td>
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<tr>
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<td>Omnidirectional</td>
</tr>
<tr>
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<td>40</td>
</tr>
<tr>
<td>Speed [m/s]</td>
<td>2.5</td>
<td>2.5</td>
</tr>
<tr>
<td>Direction</td>
<td>Towards Tx</td>
<td>Towards Tx</td>
</tr>
</tbody>
</table>

Table 1: Environmental parameters and transmitter and receiver geometries for example deep water and shallow water sound channels.

Sound speed profiles were interpolated from ASRAP temperature a data supplied by the Hydrographic Office. The deep water environment is characterised by a shallow surface layer at depths down to 100m in January. The profile is strongly downwardly refracting from 100m to 2000m and is upwardly refracting below 2000m, where the sound speed is dominated by pressure. The sound speed profile in the shallow water environment is close to iso-velocity and is mildly upwardly refracting.

4. CHANNEL CHARACTERISATION

The performance of an AComms modem is dependent on many factors including the time spread and Doppler spread of the channel, and whether energy is received via a direct Line-of-Sight (LOS) path or via surface and / or bottom interactions. The simulator enables the properties of an underwater sound channel to be rapidly assessed. The tool can be used to advise on the selection of an appropriate signalling scheme and bandwidth for a specific environment.

Table 2 summarises the characteristics of the example sound channels. The deep water environment represents a relatively simple sound channel in which a direct LOS and a surface reflected path are present at 2km range. Energy is received predominantly via the surface reflected path leading to Rayleigh fading. Doppler spreading of the signal is estimated to be less than 0.3Hz at 10kHz. The time spread of the deep water channel is low (less than 2ms) at short ranges since bottom reflected energy is heavily attenuated at high incidence angles. The channel introduces negligible inter-symbol interference (ISI) when the symbol duration is much greater than the time spread of the channel. The low time spread of the channel indicates that a short symbol duration can be employed in the deep water channel.

The shallow water environment represents a more complex channel in which energy propagates by many different paths. The time spread is 42ms and the Doppler spread around 1Hz in the shallow water environment.
Table 2: Summary of channel characteristics at 2km range and 10kHz for parameters listed in Table 1.

<table>
<thead>
<tr>
<th></th>
<th>Deep Water</th>
<th>Shallow Water</th>
</tr>
</thead>
<tbody>
<tr>
<td>RMS wave height [m]</td>
<td>0.12</td>
<td>0.12</td>
</tr>
<tr>
<td>Sea state</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Transmission loss at 10kHz [dB]</td>
<td>69</td>
<td>63</td>
</tr>
<tr>
<td>Number of significant eigen-rays</td>
<td>2</td>
<td>9</td>
</tr>
<tr>
<td>Dominant arrival path</td>
<td>Surface path</td>
<td>Multiple paths</td>
</tr>
<tr>
<td>Multipath time spread [ms]</td>
<td>1.2</td>
<td>42</td>
</tr>
<tr>
<td>Coherence bandwidth of channel [Hz]</td>
<td>830</td>
<td>24</td>
</tr>
<tr>
<td>Doppler spread due to 5m/s Rx motion at 10kHz [Hz]</td>
<td>0.04</td>
<td>1.1</td>
</tr>
<tr>
<td>Doppler spread due to surface waves at 10kHz [Hz]</td>
<td>0.28</td>
<td>0.74</td>
</tr>
</tbody>
</table>

5. BIT ERROR PROBABILITY

The reliability of a communications system is governed by the bit error probability that can be achieved at a given SNR. The simulator can be used to calculate the error performance in any environment to determine the effect of the sound channel on communications performance.

Fig. 2 shows the bit error probability (\(\xi_b\)) as a function of the SNR per bit for a Direct Sequence Spread Spectrum (DSSS) signalling scheme in the three example channels: the idealised AWGN channel, and the deep and shallow water sound channels at 2km range. The SNR per bit is equivalent to the energy per bit (\(E_b\)) divided by the noise spectral density (\(N_0\)) in the signal band. The signalling scheme is based on coherent detection of an M-ary multiple shift keyed (MPSK) signal and is implemented without error correction. The symbol duration of the DSSS signalling scheme is 16ms. The results show that the measured bit error probability of the modem in the AWGN channel is comparable to the theoretical error performance for the given signalling scheme. This indicates that the performance of the modem used during testing is close to optimal.

The simulations show that the bit error probability is highly dependent on the properties of the sound channel. The error rate in the deep water channel is slightly higher than in the AWGN channel due to motion of the sea surface and Rayleigh fading. The SNR required to achieve a given bit error probability is around 2dB higher in the deep water channel than the AWGN channel.

A much higher SNR is required to achieve the same error rate in complex shallow water channels. For example, the SNR required to achieve a bit error probability of 0.005 varies by more than 10dB between the deep and shallow water channels. The large time spread of the channel (42ms) in comparison to the symbol duration (16ms) results in frequency selective fading in the shallow water channel. Fading in the shallow water channel results in ISI which leads to an irreducible error rate at high SNR. Doppler spreading of the signal due to motion of the transmitter, receiver and sea surface, may also introduce errors. Increasing the SNR beyond a certain level has little effect on the bit error performance in a communications channel governed by ISI. In such cases, error rates can be reduced by utilising an error correction algorithm or by increasing the symbol duration.
6. COMMUNICATIONS RANGES

The signal processing implemented within many underwater communications systems can be represented by a matched filter process. In such cases, the SNR at the receiver is related to the SNR at the transmitter by the propagation loss. The average SNR at the output of the receiver processing system can be related to the SNR per bit by introducing the signal bandwidth $B$:

$$\frac{S_R}{N} = \left(\frac{S_T}{N}\right) \frac{1}{P_L(r)} = \left(\frac{E_b}{N_0}\right) \frac{R_b}{B}$$

where the signal power at the receiver $S_R$, the signal power at the transmitter $S_T$, and the total noise power $N$ are defined across the full bandwidth of the signal ($N = N_0 B$), $P_L$ is the propagation loss at range $r$, and $R_b$ is the bit rate. In the case where the signals are received by an array of sensors, the noise level $N$ is defined at the output of the beamformer.

Fig. 3a shows the bit error probability as a function of range in the two example environments for a DSSS signal transmitted at a source level of 170 dB re 1µPa @ 1m. The bit error probability is dependent on the SNR at the receiver and increases with range. The bit error probability is also dependent on the complexity of the channel and is much greater in the shallow water channel than the deep water channel.
Fig. 3: a) Bit error probability and b) probability of receiving a 100 bit message without error, for a DSSS signalling scheme transmitted at 170 dB re 1µPa @ 1m, in the deep water (solid) and shallow water (dashed) channels.

Fig. 3b shows the impact of error performance on the ability to receive a 100 bit length message without error when transmitted at a source level of 170 dB re 1µPa @ 1m. The DSSS signalling scheme is demodulated to a probability in excess of 95% at ranges up to 10km in the deep water channel. However, the maximum range at which the same signal can be demodulated to a high probability is less than 7km in the shallow water environment. Furthermore, the probability of correctly demodulating a continuous 100 bit message is limited to around 75% in the shallow water environment and decreases rapidly with increasing message length due to ISI.

Several methods can be adopted to improve the reliability of the communications system in the shallow water channel. Error correction algorithms can be utilised to reduce the error rate when the underlying error rate of the communications bearer is low. Increasing the symbol duration of the signalling scheme can mitigate the effects of ISI. In addition, frequency selective impairments of the sound channel can be overcome by reducing the signal bandwidth.

The simulator has also been used to establish the maximum communications ranges for a prototype spread spectrum signalling scheme in a low frequency band across a range of global environments. Modulated signals were transmitted through simulated underwater channels to determine the maximum range at which a bit error probability of $10^{-3}$ is achieved at an omnidirectional receiver in the presence of ambient noise.

The simulator has also been used to assess the performance of the Deep Siren tactical paging system, which is designed to establish long range communications with submarines at speed and depth. Predicted communications ranges are in close agreement with recent experimental measurements obtained in the Mediterranean Sea during RN exercise Taurus 09. The simulator was used to de-risk and plan the trial and generated performance results for the device in many other global locations.
7. CONCLUSIONS

A high fidelity simulation system has been developed specifically to evaluate the performance of underwater AComms modems. The system allows closed-loop, standardised testing to be conducted in a controlled, reproducible environment using modem hardware to modulate and demodulate the signals. The simulator is designed to rapidly assess the properties of underwater sound channels in order to identify limitations imposed by the channel on communications. Detailed modelling can be conducted in order to select an optimum signalling scheme for a particular application and environment.

Simulations show that the properties of the sound channel have a significant effect on the performance of an AComms modem. Complex sound channels, such as the example shallow water channel, typically require a much higher SNR to demodulate a given signalling scheme than benign channels, such as the example deep water channel.

The UK MoD has utilised the high fidelity simulation system reported here to carry out extensive, standardised performance assessments of AComms systems in multiple global environments. Extensive standardised testing of different frequencies and bandwidths, transmitter and receiver options, encryptions and platform configurations has been completed and the results used to de-risk and optimise trials activity.

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MULTI-MODE ADAPTIVE MAC PROTOCOL SUITE AND
STANDARDIZATION PROPOSAL FOR HETEROGENEOUS
UNDERWATER ACOUSTIC NETWORKS

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\textbf{Abstract:} This paper investigates a multi-mode MAC protocol suite for use in a heterogeneous underwater network. Modems from different manufacturers may employ different physical layers and MAC protocols. If multiple modems operating in the same or overlapping frequency bands are deployed in the same geographic area, they cannot function correctly due to interference. We propose a mechanism to achieve co-existence and communication among such heterogeneous assets. Co-existence is defined as the state where modems of the same type can communicate with each other while minimizing interference with other types. The Communications state is when all modems in one area can communicate with each other. Alien signals are defined as non-decipherable communication signals for any one modem type. The Co-existence mode requires all modems to implement an alien signal detection feature. In this mode, a CSMA based scheme will be used to communicate among modems of the same type, if alien signals are detected, and to allow non-interfering co-existence among different modem types. In order to achieve Communications mode, modems will be required to implement a standardized Physical Layer (along with their proprietary Physical Layer or as the only system and on top of the alien signal detector). Such suitable Physical Layer standards are currently being addressed under the JANUS initiative at NURC. For this mode, we propose using a MACA based protocol suite for the MAC layer. It has a distributed MACA-based mode and a centrally controlled polled mode that nodes dynamically adapt to, depending on deployment and node configurations. Features include autonomous selection of distributed or controlled MAC modes, inter-cell interference mitigation strategies, and provisions for both reliable and unreliable modes. This paper presents some simulation results and mathematical analysis for the two MAC protocol modes.

\textbf{Keywords:} Underwater, acoustic networks, adhoc, MAC protocols, standards, MACA
1. INTRODUCTION

Underwater modems currently utilize diverse physical layer standards and MAC protocols in overlapping frequency bands and do not co-exist or communicate with each other over standardized protocols. Physical Layer protocol standardization attempts are in progress, such as the JANUS initiative at NURC [1]. In this paper, we propose a potential candidate for a standardized MAC protocol suite. We use the term JANUS to refer to protocol standardization aimed at both Physical and MAC layers for underwater networks.

For MAC protocol standardization to take place, underlying physical layer and frequency band allocation standards have to be in place. An initiative that is currently looking at this issue is JANUS. In this paper we assume that such standards are in place and take the next step of addressing the MAC Layer. We assume that the packet structure defined by the Physical Layer standard has a type field that determines the structure of the packet that follows, as well as a checksum field. The MAC protocol could use control packets or data packets and will indicate this using the type field. Based on real implementation trials using ARL’s OFDM modems, we recommend that 3 or more bytes should be available in the control packet definition in the Physical Layer standard (excluding type field and checksum) in order to be able to convey sufficient information. Current JANUS proposal for example has 6 bytes (excluding type and checksum fields) in the control packet and hence is sufficient.

We aim for two levels of compliance for the standard, the first is to achieve co-existence and the second level is to achieve communications among heterogeneous assets (sections 2.1 - 2.4). We also look at how nodes can dynamically adapt their physical layer and MAC protocols based on node capability and environment sensing (section 2.5). Performance analysis is presented in section 3. We also look at some key aspects such as FEC, power control etc (section 4) and how to integrate proposals for broadcast beacons such as the JANUS Beacon into a MAC framework (section 4.3). The proposal attempts to provide flexibility for modem manufacturers to choose the level of compliance they are willing to adopt in phases, and to be able to use, innovate and improve their indigenous technologies while conforming to accepted standards.

2. JANUS MAC PROTOCOL SUITE PROPOSAL

2.1. Level-1 MAC – Co-existence, CSMA based

In the co-existence mode, we propose that all modems adopt the same detection preamble scheme for a given frequency band. If that is not possible, modems can implement an alien signal detection feature, i.e., to be able to detect a signal in its frequency band that is not of its Physical Layer type. This could be based on energy detection, for example. There could also be wakeup tones that are also present as part of the preamble structure, such as those being proposed in the JANUS initiative. Energy detection can also be used alongside the detection preamble to monitor the signal following the preamble, to determine end of the packet, for example.

With such a minimal compliance at the Physical Layer, we propose that a suitable variation of the Carrier Sense Multiple Access (CSMA) based MAC scheme be used to communicate among modems of the same type. The carrier sense comes from the ability to detect alien packets as stated above. In this scheme, nodes use a random back-off before
transmitting a packet. The details of the back-off procedure are the same as in section 2.3 for RTS packet in MACA-D. In fact this scheme can be viewed as similar to the Basic Access Scheme in 802.11 [2]. CSMA MAC protocols have lower performance compared to many other options, but at this level of minimal Physical Layer compliance, it offers perhaps one of the best solutions to avoid interference between different modem types.

2.2. Level-2 MAC –Communications, MACA based

In order to achieve communications mode, modems will be required to implement a standardized Physical Layer (along with their proprietary Physical Layer or as the only system and on top of the alien signal detector or detection preamble). Such suitable Physical Layer standards are currently being addressed under the JANUS initiative at NURC.

For this mode we propose using a MACA based protocol suite for the MAC layer. It has a distributed MACA-based mode (MACA-D) and a centrally controlled polled mode (MACA-C) that nodes dynamically adapt to, depending on deployment and node configurations as explained later on in section 2.5. In the centralized scheme, we define a cell to consist of a MAC Controller (MC) and the nodes within its control. Many of the centrally controlled MAC protocols use a polling scheme, where the MC polls the client nodes [3]. Some of the distributed protocols are ALOHA, CSMA, MACA [4], FAMA [5] etc. Among distributed protocols, some protocols such as MACA and FAMA involve handshaking using control packets before data transmission. Centrally controlled modes typically perform better than distributed modes by eliminating contention. However, in a generic network environment with heterogeneous nodes, a centrally controlled protocol alone might not be usable and distributed modes may be needed. Prior work addressing such large scale ad hoc dynamic underwater networks includes the Seaweb project [6]. The terrestrial IEEE 802.11 family of protocols also use such combined topology suites in the form of Point Coordination Function (PCF) and Distributed Coordination Function (DCF). Level 2 MAC protocol suite is the key focus in this paper and the next two sections discuss this in greater detail.

2.3. Distributed Mode (MACA-D) of Level-2 MAC

Here we look at an enhanced version of MACA that shall form the basis of the distributed mode in the Level-2 MAC protocol suite. This protocol uses RTS-CTS-DATA\_TRAIN-ACK sequence similar to those used in other MACA based schemes (note that some protocols use selective ARQ instead of ACK). We shall use DATA to indicate DATA\_TRAIN for brevity.

In the RTS contention phase, a node starts off with a uniformly selected back-off time in the range of 0 to \( W \). When the back off timer expires, an RTS is transmitted. Once the RTS is sent off, the CTS timer \( t_A \) starts. If the timer expires before reception of the CTS, the RTS back-off procedure starts again. Once the CTS is received, the DATA train is sent off followed by a wait for ACK. If an ACK is not received within the timeout period, the RTS cycle repeats. Reception of RTS/CTS packets and a possible DATA frame while waiting to send RTS triggers Virtual Carrier Sense (VCS), i.e. nodes refrain from transmissions for an appropriate time depending on the control packet received (i.e. waits until a potential CTS can be sent if an RTS is received, or waits long enough for a DATA batch to be sent if a CTS is received). Successful DATA transmission for any one node restarts RTS contention cycle for all.

To handle some nodes missing the winning CTS and interfering with the DATA phase, all nodes monitor for DATA packets. DATA packets have information on packets remaining in
the batch. This helps nodes that missed the RTS/CTS to regain VCS with a probability close to 1 after a few DATA packets are sent. Every node overhearing any packet continuously updates its own NAV (Network Allocation Vector) just as in 802.11 [2]. Another option is not to fully rely on decoding DATA packets for DATA based VCS. We can assume that the preamble detection for DATA is the same as the control packets and these packets can thus be detected (though information is not decodable) prompting a back-off or wait. Since the delay between DATA packets in a train is small and fixed, receiving nodes will continuously keep receiving these packets and they wait till the train finishes. The protocol has an additional Early ACK enhancement – a node sends an ACK instead of CTS, if the RTS is repeated for the duplicate packet train. This happens when the previous ACK is lost.

For 802.11 DCF mode, it can be shown that the optimum back-off window is dependent only on the number of neighbours and is directly proportional to it [2]. The authors have verified using independent analysis and confirmed the same finding for MACA-D, but the analysis is omitted here for brevity. Thus the back-off window size is set according to number of neighbours as known by the node over a period of time. For power control, the RTS contains the transmit power used and the CTS suggests power level based on the received SNR.

2.4. Centralized Mode (MACA-C) of Level-2 MAC

In this mode of the protocol suite, an MC controller controls the collision domain or “cell”. An RTR (Request-to-receive) initiates all communication sequences for the uplink (towards MC or between nodes in the same cell). All nodes monitor for MC control packets to detect presence of a controlling MC and then switch to the controlled MAC mode. Channels not mentioned in RTR can be assumed to be uncontrolled by the MC and nodes may make use of them as they wish (e.g. using MACA-D). Configuration determines which nodes may operate as a MC (e.g. radio buoys).

For uplink, MACA-C operates in few modes as follows:

RTR-DATA-ACK: The intended node responds with DATA in control channel modulation if it’s meant for the MC and uses power control information inferred from RTR’s received power, assuming bi-directional validity of power information. MC then closes with ACK. Multiple ACKs may be used to increase receive probability. ACK may include earliest next RTR timing and helps reduce uncertainty.

RTR-RTS-CTS-DATA-ACK: If the destination is another node (not the MC), or if a node wishes to use another FEC scheme, a node sends out RTS once RTR is received. That is followed by CTS-DATA-ACK just as in MACA-D. This mode is quite similar to the protocol discussed in [3]. CTS indicate FEC and power control information as described earlier.

For downlink, MC starts with RTS and uses RTS-CTS-DATA-ACK sequence just as in MACA-D. RTS-CTS-PILOT_DATA-ACK scheme may be used as in MACA-D for downlink channel measurements.

2.5. Dynamic MAC

We propose that a modem assess its neighborhood and switch to an appropriate MAC scheme from the above choices. When modems don’t sense dissimilar modems, they could use any physical layer and MAC schemes. This allows the usage of proprietary technologies and protocols in isolated environments. For modems that have only Level-1 compliance, if they detect alien signals (hear a standardized preamble followed by indecipherable packet or based
on energy detection) they should automatically switch to Level-1 MAC (co-existence MAC) that uses random back-off.

Level-2 compliance is possible in two ways – modems implement only the standardized Physical layer or they implement the standardized Physical layer alongside any proprietary scheme and have mechanisms to switch between them. For nodes using the compliant Physical Layer and MAC protocol, there is no change in behavior required. For nodes using compliant Physical Layer and non-compliant MAC protocol when in isolation, if they hear packets belonging to the standardized MAC protocol (as identified by the type field), they need to switch to the standardized Level-2 MAC protocol for compliance. Nodes with multiple Physical Layers (one of which is compliant), operating in non-standard Physical Layer in isolation, need to switch to compliant mode upon detecting alien signals. In level-2 MAC, there are two modes MACA-C and MACA-D. The nodes determine the presence of an MC through RTR messages. If they do hear RTR, they use MACA-C. If they do not hear RTR messages, they use MACA-D.

3. PERFORMANCE ANALYSIS OF LEVEL-2 MAC

3.1. MACA-D

Simulation results are shown in Fig. 1 (basic MACA scheme and one with DATA_VCS and Early ACK enhancements) and depict how the performance improves with packet train length (no FEC or power control is used in simulations).

![Throughput vs Batch Size](image)

Fig. 1: Performance of MACA-D and MACA-C
(Packet duration $L = 0.5$ s, detection and decoding probability $P = 0.81$, one-way latency $D = 0.5$ s, number of nodes $n = 4$)

With appropriate batch size selection, the throughput can be over 60% for reliable delivery as seen in Fig. 1. Mathematical analysis also confirms these findings but is not included in this paper for brevity. Throughput behaviour is independent of neighbour nodes as shown for 802.11 DCF [2]. Together with the need for dynamic FEC and power control, such a handshaking scheme is very suited to underwater networks.
3.2. MACA-C

Let \( p \) be the combined probability of detection and decoding for a control packet, \( D \) is average one way propagation delay, \( L \) is packet time duration, \( B \) is batch size (number of packets in a train). Let the average time period till a successful RTR reception (due to detection and decoding losses) be \( W_{RTR} \). RTR may be lost and the MC will resend RTR (potentially to other recipient nodes). This can be viewed as a geometric distribution with expected number of retries \( 1/p \). Considering \( 1/p - 1 \) failures, where time spent includes RTR duration, round trip delay and time for first DATA packet, the successful reception of RTR is given by,

\[
W_{RTR} = \left( \frac{1}{p} - 1 \right) \left( L + 2D + L \right) + L + 2D
\]

(1)

The total time for batch transmission \( s \) including time for packet train \( BL \) and time for \( n \) ACKs is,

\[
s = W_{RTR} + BL + D + nL
\]

(2)

The average throughput \( T \) is as follows and is also plotted in Fig. 1. As expected, MACA-C performs better due to lack of contention and associated collisions.

\[
T = \frac{BL}{s}
\]

(3)

4. OTHER ASPECTS OF THE PROTOCOL SUITE

4.1. Dynamic FEC and power control

The packet format is as shown in Fig. 2 and has two parts – a detection preamble and an information part. A series of such packets constitute a packet train. The information signal could be control packets using a predetermined modulation and Forward Error Correction (FEC) scheme or user DATA in variable modulation and FEC schemes. Adapting the FEC scheme for the DATA train is critical in ensuring optimum data rates. The decoding of control packet RTS could help estimate the BER and the CTS can specify the FEC scheme to use. If such a mechanism to derive BER estimates from control packets is not possible, probe pilot packets (with a known data pattern) may be sent in order to receive feedback from receiver on channel characteristics, for e.g. RTS-CTS-PILOT_DATA-ACK scheme. The receiver of PILOT_DATA indicates channel performance measures in the ACK.

![Packet Train](image)

**Fig. 2: Packet Train**
Power control may be needed to adapt the range required for routing and node connectivity. For such purposes of dynamic FEC and power control, handshaking based MAC protocols such as MACA are well suited [6].

4.2. Inter-Cell or Inter-MC Interference

When there are multiple cells and controlling MCs in a neighbourhood using MACA-C, adjacent cell interference could take place. In scenario 1, as depicted in Fig. 3, the MCs are not able to hear each other or neighbouring cell nodes’ transmissions. We assume here that all nodes only have single hop range to reach MC using power control.

Fig. 3: Neighbouring cells Scenario 1

So it will continue to use MACA-C. But as we can see, nodes from adjacent cells interfere with each other’s transmission. They observe VCS as mentioned in section 2.3 and will not reply to MC’s RTR. MC will proceed to do communications between nodes such as 1 and 2 away from such interference. Thus parallel communications takes place in neighbouring cells correctly using MACA-C in the face of inter-cell interference that does not involve the MCs.

If MCs discover that their cells are near enough to cause interference, option A is to give up RTR based control and let nodes revert to MACA-D. Option B for MCs is to continue use MACA-C with a back-off for RTR, just as in RTS back-off in MACA-D. The neighbour MCs can hear either the RTRs or the reply RTS, INFO or DATA packets. MCs obey VCS rules for allowing neighbours to complete one communication sequence (till ACK). In option B, the RTR contention will only be between MCs of neighbouring cells unlike in RTS contention involving all nodes in option A. It’s possible that since the optimum contention window is directly proportional to participating neighbours [2], the MC based RTR contention needs a shorter contention window and the effective contention period could be lower. Since under normal circumstances, MACA-C gives better performance than MACA-D and RTR back-off based method could solve the problem of neighbouring cell interference, we propose that MCs do not relinquish their roles in favour of MACA-D upon discovery of inter-cell interference, and instead use RTR back-off (option B). This idea needs to be further validated by simulations as part of future research.

4.3. Unreliable Messaging, Broadcasts and JANUS Beacon

In the Level-1 MAC, the random back-off method applies to all packets equally, including unreliable messages and broadcasts. In Level-2 MAC-D and downlink MACA-C, if unreliable short messaging (no ACKs) is required, the same contention logic as RTS can be used to send single short DATA packets using control packet FEC. In other words, DATA is sent in the place of RTS. In MACA-C uplink, RTR-DATA format can be used for unreliable short messaging (no ACKs). Broadcasts are done via these unreliable modes.
Unreliable broadcast mode can be used for beacons such as those proposed in JANUS. JANUS beacon and similar concepts attempts to allow nodes to broadcast useful information about itself to neighbouring nodes. Such broadcast packets need to come under the control of a MAC protocol to avoid interference in a given acoustic frequency band. It’s easy to accomplish such a beacon in the proposal as mentioned above using the general purpose broadcasts.

5. CONCLUSION

We have presented a comprehensive MAC protocol suite to address a diverse and heterogeneous underwater network with multiple levels of compliance which also allows for proprietary schemes to be used in isolation. The MAC protocol in the communications mode has both distributed and centralized operating modes. Some indicative performance results were also shown. The authors have begun sea-trials using the MACA-D component of the protocol suite. Preliminary results are quite promising and are expected to be published in the near future. Nodes self regulate the operation modes and topology according to self capability and capabilities of neighbouring nodes. The key vision is a self-organizing network, with nodes able to dynamically adapt to any scenario through environment discovery. We note that many of the details of the protocol are still open. Here we only attempt to provide a good direction and once there is acceptance for the fundamental ideas, then the next step of detailed standardization needs to take place.

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REFERENCES

MAC PROTOCOLS FOR MONITORING AND EVENT DETECTION IN UNDERWATER NETWORKS EMPLOYING A FH-BFSK PHYSICAL LAYER

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Abstract: Underwater acoustic networks of fixed and autonomous nodes can be a very valuable tool in a number of situations, from environmental monitoring to emergency scenarios (e.g., ships in distress). In this paper, we compare the performance of some MAC protocols for underwater networks in typical scenarios. We consider random access protocols, which provide sufficiently high performance in case of low traffic, and then compare random access with handshake-based access, which achieves better coordination among nodes, at the price of greater control overhead. We consider both periodic traffic and event-driven traffic, and provide insight about which scheme achieves the best performance in terms of relevant network metrics such as throughput, error rate and overhead. In our evaluation, we assume the network protocols to work over a low-rate FH-BFSK-based physical layer, a simple technique that can be easily implemented, e.g., to work as a common PHY for different modem hardware.

Keywords: Underwater acoustic networks, FH-BFSK, MAC protocols, random access, handshaking, performance evaluation, simulation, periodic traffic, event-driven traffic.
1. INTRODUCTION AND RELATED WORK

The growing interest in underwater acoustic networking covers many aspects, from channel access, to routing and topology control [1]-[4] and is well understandable in light of the wealth of applications that could be supported by autonomous networks of underwater fixed and mobile nodes. However, while research has mostly focused on medium to large-sized network simulations so far, these scenarios are difficult to realize, mainly due to the very high cost of underwater nodes. Another impairment to large deployments comes from the substantial differences in the hardware sold by different manufacturers, who generally employ proprietary modulation/coding formats and undisclosed receive algorithms.

In order to harmonize the communication format of underwater nodes, at least for a restricted number of functions, recently simple modulation and coding schemes have been created that could be easily included in standard off-the-shelf hardware. Such schemes may rely, e.g., on Frequency-Hopping Binary Frequency Shift Keying (FH-BFSK), a signalling pattern that offers some resilience to interference, thanks to the different frequency hopping patterns of different transmitted signals, which reduce the probability that two transmissions by different nodes collide, and therefore both packets are lost. Moreover, the FH-BFSK signal is relatively easy to receive using non-coherent detection. Thanks to its simplicity, this transmission format can be supported by a number of hardware implementations. Its main drawback, however, is the very low transmit bit rate, on the order of the tens of bits per second, which limits both its applications and the higher-level protocols that can operate on top of it.

In the present study, we compare three Medium Access Control (MAC) protocols [1], [2], [5], featuring different transmission coordination schemes, from no coordination to full-fledged handshakes. We study the behaviour of these protocol, on top of the previously overviewed physical layer, in the presence of two different applications which must handle periodic and event-driven traffic, respectively; the performance evaluation we carry out aims at providing insight on the access scheme yielding the best performance in terms of throughput, transmit error rate and other relevant network metrics.

2. CONSIDERED PROTOCOLS

In this section, we briefly review the protocols we have chosen for our comparison. The first, ALOHA, is a basic random access protocol, and represents a typical uncoordinated scheme; the second, T-Lohi, is a contention-based access scheme that employs tones to manage contentions; the last one, DACAP is a 4-way handshake-based protocol, where nodes exploit the transmission timings in order to detect possible collisions, and thereby warn the transmitter, or directly refrain from sending packets. We highlight that this form of handshaking yields better coordination but also more overhead than T-Lohi’s.

2.1. ALOHA

ALOHA [5] is a simple random access algorithm, whereby a node immediately transmits whenever it has data to send. This means that collisions may take place if two mes-
Fig. 1: An example of contention for channel access in T-Lohi.

messages arrive at the receiver at the same time. Standard contention resolution techniques (e.g., random backoff times) are to be applied in this case. ALOHA comes with no specifications as to sending acknowledgment (ACK) messages to confirm correct data reception or not. In case ACKs are used, a backoff policy can be implemented, whereby a terminal refrains from transmitting for a random amount of time if an expected ACK is not received. ALOHA, in general offers poor throughput performance and is very prone to congestion. Nevertheless, it can be a feasible option in the presence of light traffic [3].

2.2. Tone-Lohi

Tone Lohi (T-Lohi) [1] is a reservation based MAC protocol. In T-Lohi, nodes detect and count the number of neighbours simultaneously accessing the channel, and contend through a traffic-adaptive backoff algorithm driven by the estimate of the number of contenders. The slow underwater sound propagation favours this mechanism, as long as the signalling packets have a much lower duration than the propagation delay. Contenders are detected through the use of wakeup tones, which allow the nodes to stay asleep most of the time, thus providing substantial energy savings during the reservation phase. On the other hand, the use of tones requires a specific wake-up tone detector, which should listen to the tone with minimal energy consumption.

To conceptually describe the protocol, it is useful to consider its synchronized version, ST-Lohi. In ST-Lohi, all nodes in the network are aligned to contention slots whose duration is equal to the maximum propagation delay plus the tone length. Any nodes seeking channel access must contend by sending their reservation tones exactly at the beginning of these slots, if they are not restricted by backoff. After sending the tone, a node waits and listens to detect the arrival of other tones for the rest of the contention slot. If it does not hear any other tone, it wins the reservation, and immediately transmits its data. Otherwise, a contention is arbitrated among the multiple nodes trying to reserve the medium. More specifically, the nodes back off for a random number of slots which depends on the number of contenders, and retry at a later time. In order to obtain the number of channel access competitors, the nodes count the received tones, and use this number as their backoff window size. A winner is elected whenever a node is the only one to transmit a tone in its contention slot. An example of such a contention involving two nodes can be seen in Fig. 1.

Note that the long underwater delays are exploited, in that different propagation delays allow tones to be more separated in time at the receivers, and thus help count the number of contenders. When no synchronization can be assumed (e.g., to save signalling efforts or
because of the difficulty of synchronizing a multihop network), the timing of tone transmissions can be slightly modified to obtain a conservative (cT-Lohi) and an aggressive (aT-Lohi) version (where contention slots are asynchronous among nodes, and the length of a slot is respectively longer or shorter [1]).

### 2.3. DACAP

Distance-Aware Collision Avoidance Protocol (DACAP) [2] is a non-synchronized handshake-based access scheme that aims at minimizing the average handshake duration by allowing a node to use different handshake lengths for different receivers. The protocol is specified as follows. The transmitter and receiver notify their intention to set up a link through an RTS/CTS exchange. If, after sending the CTS, the receiver overhears a packet threatening the pending reception, the node sends a very short warning packet to its transmitter. To exploit the advantage granted by this further signalling the sender waits some time before transmitting the data packet. If it overhears a packet meant for some other node or receives a warning from its partner, the sender defers its transmission. These situations are depicted in Fig. 2. In some cases the warning arrives while the node is transmitting the data, and hence is lost because modems are half-duplex. The length of the waiting period is chosen so as to guarantee absence of collisions, and depends on the distance between the nodes, which the sender can learn by measuring the RTS/CTS round-trip time. We note that handshakes only need to avoid collisions from nodes closer than a certain distance, as farther nodes would create little interference, and thus allow the packet to be received in any case. Hence, handshakes between close neighbours can be made short, while those between far apart nodes need to be longer. To achieve a trade-off that maximizes the throughput of a given network, a minimum handshake length $t_{\text{min}}$ is predefined for all the nodes. For a network in which most links are close to the transmission range, $t_{\text{min}}$ needs to be nearly twice the maximum propagation delay. When some links are shorter, it can be reduced. Two versions of the protocol can be envisioned: with acknowledgments (ACKs) sent right after receiving a correct data packet, and without ACKs. In the first case, the protocol requires slightly different timings with respect to the second case, in order to accommodate the ACK message [2]. The DACAP protocol (with or without ACK) has been designed to provide a controlled collision environment where multiple data communications can coexist without harming one another.
3. SIMULATION RESULTS

3.1. Network settings and parameters

Our main purpose in the following comparison is to test the applicability of the protocols described before to underwater communications performed using a FH-BFSK modulation with 13 subcarriers. In our implementation, a convolutional code of rate 1/2 is also employed to yield further resilience to bit errors, and all packets are preceded by an 8-byte header, which is also encoded. We recall that the transmission format is designed to be easily supportable by diverse acoustic modem implementations. To this end, transmissions are performed in the 9-14 kHz band. The very low bit rate of the resulting system, 160 bps including PHY-level coding, restricts the effective transmit rate to at most 80 information bits per second. Hence, protocols must be very effective at employing such a scarce resource, and should impose little overhead, as a general rule. This explains why we chose to compare a protocol with no overhead (ALOHA) with another bearing light overhead (T-Lohi) and a third imposing greater overhead (DACAP). For each protocol, we considered both an ACK and a no ACK version: in the latter case, we set the maximum number of retransmissions of any packet to 5.

We arrange nodes in a rectangular grid topology that covers an area of 5 km × 2 km. We deploy either 4 or 10 nodes, by dividing the area in 4 (respectively, 10) rectangles and placing a node at the centre of each rectangle. These nodes must communicate to a sink placed at the centre of the network area. We reproduce two different traffic scenarios: in the first one, nodes periodically report environmental data. The corresponding traffic is generated according to a Poisson process of rate $\lambda$ packets per second per node. In the second scenario, nodes must detect moving objects that traverse the network area. To emulate the corresponding bursty traffic pattern, we assume that an object crosses the network and triggers packet generation events at the rate of 1 packet every 10 seconds whenever it comes within the detection range of a node. For simplicity, we assumed that the detection range of a node is 1.5 km, and consider only a 10-node topology, which adequately covers the area.

We have implemented the protocols described in Sec. 2 using ns2 [7] and the ns2-MIRACLE extensions [8]. In order to reproduce acoustic propagation, we have fixed a location for our experiments at 49.25°N 10.125°E, close to the Pianosa island, off the northeastern coast of Italy. Bathymetry data have been taken from the General Bathymetric Chart of the Oceans [9], a public database offering 30-arcsecond spaced samples; bottom sediment parameters are taken from the National Geophysical Data Center’s Deck41 database [10]. Sound speed profiles are computed as averages of the profiles measured during the GLINT’08 sea trials [11] or from the World Ocean Database [12] if the GLINT data set did not cover the wanted location and month of the year. The Bellhop ray tracer [13] is finally used to simulate signal propagation among all nodes, including the sink.

3.2. Results for periodic traffic

We start our performance evaluation by considering throughput, defined as the fraction of the offered traffic per node (indicated in the abscissa) that the protocol can manage. In
particular, Figs. 3 and 4 show throughput for a network with 4 nodes and 10 nodes, respectively. With 4 nodes, the overall traffic load is very light, therefore light handshakes (such as in aT-Lohi) or no handshakes (ALOHA) are preferred with respect to more complicated handshakes (DACAP) and cT-Lohi, which forces the nodes to stay silent for longer times before transmitting. From Fig. 3 we also infer that, in general, using ACKs tends to decrease the throughput, due to both the longer handshake duration and the probability that an ACK message collides with data packets. We observe similar trends in Fig. 4, where the presence of 10 nodes yields greater traffic. In this case, handshake-based protocols are put under heavier stress: at low to medium traffic aT-Lohi (both with and without ACK) and DACAP (no ACK) offer good throughput, but are eventually outperformed by ALOHA (no ACK), which is much lighter for greater $\lambda$.

The drawback of this version of ALOHA can be seen in Fig. 5, which depicts the fraction of erroneous transmissions over all transmissions. This figure shows that ALOHA bears a high error rate (roughly 1/3), yet lower than the failure rate of coordinated protocols, which undergo a greater probability that signalling transmissions interfere with data transmissions. On the other hand, the benefit of coordination among nodes is observed in particular at low rate, where the error probability of ALOHA is outperformed by all versions of T-Lohi and DACAP. To conclude this first part of our comparison, we show in Fig. 6 the protocol overhead, defined as the ratio of non-data bits sent over all transmitted bits (purged of convolutional coding overhead). Hence, this figure accounts for both signalling overhead and erroneous packets (which have to be retransmitted). We see that the lowest overhead is achieved by
the no ACK versions of cT-Lohi and aT-Lohi, whereas ALOHA with no ACK has slightly greater overhead due to the larger number of collisions between data packets that take place as traffic increases. We recall in fact that collisions are the only source of overhead for no ACK ALOHA, whereas for other protocols the number of signalling messages increases, and consequently collisions of data and signalling packets increase as well. This also shows that DACAP has a worse overhead performance, due to its longer handshakes.

### 3.3. Results for event-driven traffic

In this scenario, we recall that an object moves through the network and is detected by the nodes. Unlike in the previous evaluation, here we focus on the arrival time of the first packet triggered by the event detection (a measure of the readiness of a protocol) and on the arrival time of the last packet (a measure of the ability of a protocol to handle bursty traffic). Again we consider ACK and no ACK versions of all protocols, and assume that ACK versions are fully reliable (infinite number of retransmissions). For the no ACK versions, we also measure the packet error rate. We report in Table 1 the packet arrival times and the error rate for ALOHA, DACAP, aT-Lohi and cT-Lohi. Unlike in the previous set of results, here we observe that ALOHA with ACK yields the lowest arrival times of all fully reliable protocol versions. This is due to the bursty traffic pattern, which causes packet generation events to eventually take place away of the first nodes that sense the moving object. A reduction of local interference follows, so that the FH-BFSK modulation format and the convolutional code together can sufficiently protect transmissions from bit errors. In addition to the simplicity of the ALOHA scheme, this also explains why ALOHA performs better.

A second observation is in order here: while ALOHA without ACK bears an unacceptable error rate, the no ACK versions of the other handshake-based protocols trade off error rate for the overall duration of transmissions. In particular, cT-Lohi yields almost zero errors even without ACK, but takes 183 s to complete transmissions. aT-Lohi’s errors are on the order of 0.3%, but the protocol takes 116 s to complete, which is very close to no ACK ALOHA. Finally, DACAP yields a 3.7% error rate, but completes in a shorter time than fully reliable ALOHA (98 s against 105 s). Therefore, if such error rates can be withstood by the application running on top of the protocols, it might be worthy to consider non-reliable handshake-based schemes instead of fully reliable ALOHA. Furthermore, we recall that, unlike DACAP and ALOHA, aT-Lohi and cT-Lohi keep nodes in a low-power

<table>
<thead>
<tr>
<th>Protocol</th>
<th>First packet arrival (s)</th>
<th>Last packet arrival (s)</th>
<th>Error rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>ALOHA (ACK)</td>
<td>1.15</td>
<td>105</td>
<td>---</td>
</tr>
<tr>
<td>DACAP (ACK)</td>
<td>1.92</td>
<td>178</td>
<td>---</td>
</tr>
<tr>
<td>aT-Lohi (ACK)</td>
<td>1.43</td>
<td>127</td>
<td>---</td>
</tr>
<tr>
<td>cT-Lohi (ACK)</td>
<td>1.72</td>
<td>202</td>
<td>---</td>
</tr>
<tr>
<td>ALOHA (no ACK)</td>
<td>1.15</td>
<td>29.6</td>
<td>0.43</td>
</tr>
<tr>
<td>DACAP (no ACK)</td>
<td>1.80</td>
<td>98.0</td>
<td>3.7e-2</td>
</tr>
<tr>
<td>aT-Lohi (no ACK)</td>
<td>1.43</td>
<td>116</td>
<td>1.9e-3</td>
</tr>
<tr>
<td>cT-Lohi (no ACK)</td>
<td>1.72</td>
<td>183</td>
<td>3.4e-4</td>
</tr>
</tbody>
</table>

*Table 1: Packet arrival times and error rate for the event-driven traffic scenario.*
state while not transmitting: this offers the opportunity to save energy, which must also be accounted for when choosing the network protocol to be employed in the network.

4. CONCLUSIONS

In this paper, we have presented a comparison among three different protocols that are suitable for use in underwater networks employing a low rate FH-BFSK physical layer with convolutional coding. Each protocol bears a different level of coordination and a correspondingly different handshake complexity: ALOHA (no coordination), T-Lohi (light coordination) and DACAP (strong coordination). For each protocol, we have tested both a reliable and a non-reliable version, under both periodic and event-driven traffic. Our study highlights that while the aggressive version of T-Lohi yields the best performance under periodic traffic, there is no clear winner in the event-driven traffic case, as the best protocol choice actually depends on the capability of the application to withstand a certain fraction of packet errors.

5. ACKNOWLEDGEMENTS

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JANUS: From Primitive Signal to Orthodox Networks

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Abstract

JANUS is an underwater acoustic communications method. It is a robust, primarily uni-directional and unsolicited public transmission. It is composed of three tones followed by a hyperbolic frequency modulated sweep (HFM) and 64 bits of information relaying the sender’s methods, purpose and intentions. The composite signal enables: acoustic channel estimation, time synchronization, information exchange and provides some knowledge of the propagation paths related to ocean structure.

JANUS transmissions are evaluated for message decoding success over distances up to 6000 meters and depths from 4 to 100 meters. The effects of relative platform motion (Doppler) are assessed up to 5.3 meters/sec for mobile surface and autonomous underwater vehicles (AUVs). Field experiments were conducted in the Mediterranean Sea with varying stratification, including strong reverberant and multipath periods. The sound speed variations were up to ~30 m/sec from sea surface to sea floor. The data transmission bit error rates (BERs) were evaluated as functions of depth, time and the internal wave climate. Ray tracing methods were employed to evaluate optimal locations for fixed and mobile assets to optimize communications, media access control (MAC) and network routing methods.

Keywords: underwater communications, channel estimation, orthogonal frequency hopping, FH-BFSK, bit error rates, internal waves, ray tracing, underwater networks

I Introduction

The intent of JANUS is to provide a message containing information about the transmitter and allow the receiver(s) to make decisions based upon additional knowledge. Such communications, frequently called beacons, are common, have been used for millennia, are intentionally conspicuous and robust. Examples of beacons include light houses, instrument landing systems (ILS), wireless local area network access points (Wi-Fi 802.11) and many other methods. Usually a beacon includes some information about the transmitter’s location or an “identity packet” of known structure. This document provides the understanding, experimental results and rationale for using JANUS for initial contact between dissimilar underwater assets. JANUS is unique in its open and public nature such that academia, industry and governments may all benefit from its use. The tools necessary to create a JANUS signal, encode it in a desired frequency band and decode a received signal are publically available at: http://nrcsp.zftp.com/users/janus-tmp/.

Ocean data acquisition, telemetery and networking services are disciplines that utilize underwater communications. The scientific, technical and engineering efforts over the last few decades have resulted in many products with commercial success. Unfortunately, the great diversity of proprietary modulation schemes, acoustic frequencies and bandwidth usage have prohibited underwater devices from being interoperable. The basic definitions for data packet, time base, MAC, routing and control frame standards do not yet exist for heterogeneous underwater networking. Additionally, the ocean is in a constant state of change with multi-path, reverberant and Doppler shifted layers of complexity and remains severely under sampled.
The NATO Undersea Research Centre (NURC) organized a workshop in March of 2008. Attendees from academia, industry and government created the international framework to address a public standard for underwater communications. It was with the cooperation and guidance of this group that JANUS has evolved beyond the initial concept to actual sea deployments, to create the foundation of an open-standard method of communicating underwater.

The initial concept for JANUS evolved from discussions in California with Walter Munk in 2005 and solidified onboard the German RV Planet with Ivor Nissen and Wolfgang Jans in 2007. The first short distance transmissions were made within the confines of the small harbour (darsena) bordering the NURC compound (La Spezia, Italy) using M-FSK modulation. As a result of the first JANUS Workshop in March 2008, the initial signal was modified to satisfy additional ‘group requirements’. Representatives from eleven countries have helped build JANUS.

II JANUS Signal Composition

Several modulation techniques were considered as candidate methods for JANUS. Selection criteria included the ability to provide relatively simple encoding, decoding, electronic hardware requirements, low firmware complexity and minimize the power requirements.

The chosen encoding method is “Frequency-Hopping Binary Frequency Shift Keying” (FH-BFSK) and has been described in detail. It is easily created and has been demonstrated to be robust as discussed in the next sections of this document. The data presented below is the result of the version 0.0 JANUS signal composed of: a) 3 wake-up tones with each tone having a duration of four chip-lengths b) a hyperbolic frequency modulated (HFM) sweep of the entire band and c) 64-bits of information (resulting in 144 bits after convolutional encoding). Optionally, a ‘data payload’ can be appended to the 144 bits to enable the transmission of optional data such as an AUV status report, environmental data, etc. The transmitted data is interleaved to help distribute possible errors that may result from environmental fluctuations and ocean physics. Below (fig. 1) is a spectrogram of a typical JANUS transmission. The received signal decoding is discussed later in this document.

The purpose of the three wake-up tones is to allow underwater modem hardware adequate time (~0.400 seconds) to come to life and start data acquisition. The HFM sweep, which is more
‘immune’ to Doppler shift influences, aids in time synchronization alignment. The three tones and the HFM sweep together provide a unique and robust target for matched filter detection purposes. The HFM may also be used to investigate time variant oceanographic phenomena including sea state, internal waves and water column properties. The central frequency of the source transducer establishes the bandwidth of the transmission and the chip rate of the JANUS signal. Thus, a JANUS signal can be viewed as a “ratio of times and frequencies” as specified by its central frequency. As an example, in the 9-14 kHz band, with 11520 Hz as the central frequency, there is 160 Hz of separation between ‘bits’, the chip rate duration is 6.25 milliseconds and the bit rate is 160 bits per second. A higher central frequency will result in a wider frequency band usage, a shorter chip rate duration and a higher bit rate. The bandwidth efficiency is low, only ~0.03 bits/sec/Hz bandwidth.

III Experimentation at Sea: Confined and Shallow Water tests (~4 to 40 meters)

In late 2007 and early 2008 extensive harbour testing was completed over distances from 30 to 6000 meters. The shallow ~4 meter depths and the stone structured boundary of the harbour provided an adequate site for robustness testing. Transmissions were in several bands from 1 to 30 kHz. Most tests were completed with 11520 Hz as the central frequency resulting in about 5000 Hz of bandwidth. After promising low error rate (<1%) results, additional testing was conducted in the waters to the west of Porto Venere near La Spezia. The CRV Leonardo towed an acoustic source at different speeds, sound pressure levels and depths.

FIG. 2 is the ship’s position vs. the transmitted power level and the received s/n ratio. The dotted line represents where the s/n ratio should be adequate to result in zero errors.

Littoral Waters (100 meters)

Extensive experimentation occurred during the GLINT08 trials in July and August of 2008 in the waters surrounding Italy’s Pianosa Island in the Tuscan Archipelago. Pianosa is an excellent location from which to conduct experiments. The oceanographic environment in the summer is characterized by strong stratification (fig. 6b) and an active internal wave climate. Over 8,000 JANUS transmissions were made during a four week period. The acoustic transmissions were made using several types of acoustic sources that ensonified in six bands, from 1 to 70 kHz, and
were received by diverse fixed and mobile assets. The fixed platforms consisted of bottom-mounted acoustic modems, transmitters (sources) and receivers (hydrophones) that were cabled to shore. The mobile platforms included multiple ships with towed sources, AUVs with modems, autonomous surface vehicles with modems, towed acoustic arrays, gateway buoys and sonobuoys. The mobile platforms had relative velocities up to 5.3 meter/sec. Deployment durations lasted from hours to several weeks. The acoustic data was collected (24 bits at 192 kHz per hydrophone channel) using a fibre-optic cable to shore. The collective cable length was over four kilometres and allowed the ~10 Gbytes/ hour of acoustic data to be written directly to a hard disk on land.

IV  Signal Propagation, Decoding Errors and Robustness

Decoding must occur after a signal has been encoded and transmitted. Decoding difficulties are manifold in origin and vary in time and severity. After temporal alignment (using a matched filter), each chip period is evaluated for the presence of one of two tones representing either logical “0” or logical “1”.

Multipath, ducting and shadow zones adversely affect decoding efforts. Figure 3 ray tracing plot depicts a signal propagating over a range of ~600 meters in ~30 meters water depth. Low signal levels exist in the surface layer ranging from 100 to 270 meters. Figure 4 depicts the same signal propagating over a range of ~6000 meters with most of the energy propagating above the mixed-layer depth. Although the raw BER increased with range (i.e. s/n in fig. 2), successful decoding occurred as far as 6000 meters from the source. A strong sound speed profile (ssp) gradient existed between 4 and 15 meters (fig. 5). However, a seasonal change of sound speed profile and source level depth can result in the signal propagating primarily below the mixed-layer as shown in figure 6a.

FIG. 3. Ray tracing 600 m          FIG. 4. Ray tracing 6000 m          FIG. 5. Winter ssp profile

FIG. 6a. Path and strength of a 2000 Hz signal propagating into shallow water in summer          FIG. 6b. Summer ssp profile
V Time and Space Variant Decoding Errors

Fundamental understanding of nature frequently requires devout long-term observations. During GLINT08 several long duration acoustic communications experiments lasting several hours, were completed. In this section the environmental influences on BERs is discussed. The BERs are associated with time spreads as depicted in figure 7 and are correlated with the water mass changes as shown in figures 8 through 10 below.

Internal waves are caused by differences in hydrostatic pressures in a stratified liquid. The maximum amplitude of an internal wave occurs at the strongest density gradient. Internal waves influence acoustic signal propagation in time and space. The internal wave frequency was calculated during GLINT08 using the Brunt–Väisälä frequency $N^2$, where $N = \sqrt{\frac{g}{\rho} \int \frac{dp}{dz}}$ and $\rho$, is the potential density which depends on both temperature and salinity. The water column density was determined with several CTD profiles and resulted in an $N^2$ value of near 4 cycles per hour (i.e. 15 min. periods) at the bottom of the mixed layer. The maximum estimated internal wave amplitude is ~15 m when the ensemble of isotherm excursions from CTD casts is used as a proxy.

The raw BER was calculated for the signals from two fixed transmitters (9-14 kHz band, 1.5 m above the bottom) transmitting to a vertical array of three hydrophones. Each transmitter made a JANUS transmission every 30 seconds for over four hours. The array was located ~500 m from the transmitters. The first hydrophone (H1) was at 20 m, the second (H2) at 40 m and the third (H3) at 60 m. The water depth was ~80 m. Below is the raw signal and the decoded (Viterbi decoding) BER at each hydrophone plotted as a time series. H1 hydrophone was at the base of the mixed layer and exhibited the highest BER as is to be expected. The mid-water H2 hydrophone,
which was less subject to multi-path arrivals and fading, had the lowest long-term BER. An FFT was used to inspect each hydrophone’s raw BER. A secular water column spectral peak is apparent at ~15 minutes. The N^2 Brunt–Väisälä frequency coincides with the 15 minute BER however the water measured column data is too sparse to show causality.

FIG. 8a and 8b.

H1 Hydrophone
Depth = 20 meters
BER and FFT
(at mixed layer)

FIG. 9a and 9b.

H2 Hydrophone
Depth = 40 meters
BER and FFT

FIG. 10a and 10b

H3 Hydrophone
Depth = 60 meters
BER and FFT

FIGS. 8a-10a show the raw (in red) BERs for a signal received at three different depths. A signal transmission was made every 30 seconds for over four hours. FIGS. 8b-10b show the FFT of the raw BERs with a spectral peak at 15 minutes (4 cycles/hr).

As stated, it is not possible to discern if the variation in BER results from internal wave driven vertical water motions or if the BER is resultant of advective water flow. The water velocity was measured using an ADCP and yielded typical velocities under 10 cm/sec which decreased with depth. If the forcing function of the BER is advective, then the characteristic length scale would be of the order of ~90 m for a ~15 minute period. A thermistor string and ADCP will be used in future experiments to help resolve the higher frequency internal waves.

Regardless of BER origin, an AUV or other platform could exploit the BER diversity and optimize the location (or time) from which to transmit or receive information. In the example above, the optimal water column location would be near ~40 where the BERs are lowest.

Additional problems must be overcome when transmitters or receivers are mobile. Any relative motion between the transmitter and the receiver will influence the apparent frequency of
a transmitted signal (Doppler). JANUS signals were transmitted and successfully decoded with up to 5.3 m/sec relative motion. The Doppler at 5.3 m/sec is equivalent to a shift of ~35 Hz, remaining well below the 160 Hz separation between tones a 10 kHz central frequency.

Ambient noise levels fluctuate due to many sources including ship and small boat traffic. When a small boat is close to a receiver the engine, gear box, propulsion, and flow noise can extend to over 10 kHz. Such anthropogenic noise can cause significant increases in BER. Several measurement periods included small boats passing close to the vertical hydrophone array which was receiving JANUS signals. The raw BER increased during the boat passage periods but almost none (<1%) remained after Viterbi decoding (similar to fig. 10a above).

VI Medium Access Control (MAC), Routing and Networking

The JANUS workshop held in 2009 (http://nrcsp.zftp.com/users/janus-tmp/) addressed the MAC issues and how to mitigate acoustic interferences. Several MAC protocols are currently under consideration including: CSMA “ALOHA” (carrier/preamble sense), Tone-Lohi and MACA (RTS/CTS). These MAC techniques are well described in other documents, help improve the frequency band usage and help minimize the signal interference from other transmitters.

Since JANUS has a known structure and repetition interval, a transmitted JANUS signal will allow the receiver to have knowledge of the multi-path environment, signal strength and BER. Optimizations may be subsequently be made for future message timing and routing. Successful routing may include multiple hops at higher frequencies (short distances) or fewer hops at lower frequencies (longer distances). An AUV may recognize a ‘shadow zone’ (as in fig. 6 above), be incapable of receiving any signal or may ‘decide’ to change its behaviour, heading, depth or frequency to minimize the BER. Optimizing location is a three dimensional task, varying in time, and as complex as the ocean itself.

VII Conclusions

A public standard must be robust enough to remain functional during adverse environmental conditions. JANUS transmissions have been made in Europe, North America and Asia, in six frequency bands (from 1 to 70 kHz). It has been demonstrated to be robust (i.e. BER <1%) under multi-path conditions, resistant to acoustic interference and is Doppler tolerant. JANUS robustness is at the expense of bandwidth efficiency (< 0.1 bits/sec/Hz bandwidth).

The success of future systems will rely upon the establishment of underwater standards. Underwater networked systems must interact on multiple network layers to disseminate and harvest information. A public standard will generate a common metric by which to quantify future system performance. Radio transmissions are regulated by international standards. Those standards have created the framework for terrestrial network designers to create interoperable products. There is a myriad of applications for JANUS underwater including network discovery, submerged vessel collision avoidance and the public announcement of surface vessels. There is no public international standard for physical layer underwater communications and networking. JANUS is an attempt to provide a public basis for such a standard to emerge.

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References

THE USE OF UNDERWATER COMMUNICATION NETWORKS IN FIXED AND MOBILE SENSING SYSTEMS

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Abstract: During the past decade Defence Research and Development Canada – Atlantic has been investigating the use of underwater deployable systems and autonomous vehicles. Both of these fixed and mobile systems are dependent on reliable underwater communications and, ideally, on the establishment of efficient and robust communication networks. This paper describes some of the project work that has been undertaken by DRDC and how underwater networking has been an enabling technology. In particular, the NetALS networked sensor concept is described and how we are attempting to implement this concept.

The NetALS concept involves the use of two independent communication networks. A short-range, low-power, network collects data from a field of deployed sensor nodes at a larger, more capable, data fusion node called a clusterhead. The clusterhead nodes also contain long-range network hardware that provides connection with users through gateway nodes. At present, we have investigated optical, electromagnetic, and acoustic means of providing reliable short-range physical layer mechanisms. The long-range network capability is currently provided by using Teledyne telesonar acoustic modems utilizing the US Seaweb networking code.

The development of practical underwater networks is a difficult task that requires a broad range of skills. Not only must the physical layer provide reliable links in all environmental conditions, but there are a host of protocols that are required to support the network discovery and maintenance in addition to interoperability, message formation, and system security. There are also extreme limitations in achievable bandwidth and link range, and there are strong limitations in power usage and processing. These difficulties make underwater networking a challenging and rewarding endeavour that is showing constant progress.

Keywords: networks, modem, acoustics, optics, electromagnetic
1. INTRODUCTION

The underwater environment is a uniquely difficult one for communications with issues resulting from the prospect of large acoustic propagation losses, extended and variable propagation delays, strong multi-path signals, and limited bandwidth capacity and channel stability. In addition, underwater non-acoustic signalling methods also experience large propagation losses and scattering issues. Communication channel limitations together with deployed sensor node energy limits combine to produce a very challenging problem in the development of underwater sensor networks.

This paper describes some of the efforts that have been carried out by Defence R&D Canada (DRDC) in the conduct of a number of projects that have investigated the potential for deployable surveillance sensor networks. Acoustic, electromagnetic, and optical methods of underwater communication have been investigated.

2. DRDC RAPIDLY DEPLOYABLE SYSTEMS

A few years ago DRDC successfully completed the Rapidly Deployable Systems (RDS) technology demonstration project (TDP) [1]. The RDS concept was to create a pair of tripwire barriers from a number of sea-bottom arrays linked together and with an operator by an underwater acoustic network. The concept is effective at providing choke point surveillance and contact redetection cueing.

Users communicate with the underwater network either by using a gateway buoy that provides an interface between a radio transceiver and the underwater modem or by a direct connection with a modem. The US Seaweb acoustic networking code [2,3] was provided under collaboration with SPAWAR Systems Center – San Diego and the Naval Postgraduate School in Monterey. This leading underwater network solution was an ideal fit to the project requirements.

Solid and reliable communications were established in a number of different littoral environments including the Arctic, a Norwegian fjord, North Atlantic coastal regions, and the warm waters of the Carolinas and Gulf of Mexico. Teledyne telesonar modems [4] with the Seaweb code reliably provide acoustic data links of moderate speed (800 bps) over horizontal ranges typically in excess of 3 km. A maximum shallow water range in excess of 10 km has been observed in several environments.

Figure 1 shows an RDS processor during a recovery operation. The computer has inflated a lift-bag under command of the on-board processor. The command was received via the telesonar Seaweb modem transducer that is visible on its own float above the main canister.

3. NETALS FIXED SURVEILLANCE SYSTEM CONCEPT

To complement the RDS concept of deployable surveillance a new sensor field concept was developed. This deployable sensor system has become known as the Networked Autonomous Littoral Surveillance (NetALS) concept [5]. Figure 2 illustrates the NetALS concept, which is intended to provide area coverage.

Figure 2a shows how a 1-km square area might be monitored by a deployed sensor network. The central hexagonal symbol represents a capable sensor node called a clusterhead that is equipped with two separate communication networks. One network is intended only for
short-range communication with hop distances typically shorter than 500 m. The other network is intended for longer ranges typically between 1 and 2 km.

Fig. 1: An RDS processor canister during a recovery using a computer controlled lift-bag. The telesonar modem transducer is visible above the canister at the end of a 2-m long tether.

The short-range network provides a control and message capability between the clusterhead and a field of distributed sensor nodes that might each provide anything from a simple detection to a target localization message at intervals of time. Most of the network activity is expected to occur on the short-range links between the distributed sensors and their associated clusterhead. The short range communication links generally require only relatively low-power sources for data transmission. Using the short-range network rather than a more powerful longer range capable modem can potentially conserve system energy. It can also serve to minimize the risk of traffic congestion for large area coverage. Additionally, by purposely limiting the transducer power for the short-range network, the majority of system communications can be made more difficult to counter-detect.

The clusterhead receives messages from the distributed sensor nodes over a period of time. The intention is for the clusterhead to fuse the individual target detections in order to reduce false alarms and to provide higher level information concerning the target detections such as target tracks, vessel course and speed, and possibly target classification.

Based on predetermined operational characteristics, the clusterhead makes use of the long-range network to communicate the detection and fused information to the operator. By communicating infrequently over the long-range network, system energy can be conserved. The long-range network capability is well matched to the Seaweb system. Figure 2b shows how the long-range network is intended to operate between clusterheads. Fig. 2a also shows an optional link to a gateway buoy, repeater, or Stealth Buoy [6] shown by the three devices located below the clusterhead symbol.

The network capabilities potentially allow for highly efficient system operation. The operator can provide an additional level of data fusion by combining clusterhead messages
from different areas: extended target tracks and classifications can be made. Additionally, the operator, or the clusterheads themselves, could reduce detection thresholds of the sensor nodes in order to increase detection probability of an important target. Similarly, detection messages could be suppressed when uninteresting targets are present with the result that system energy is conserved or managed.

4. SHORT-RANGE COMMUNICATIONS

A primary goal of employing the two networks in the NetALS concept is to reduce the system energy usage by employing an efficient short-range modem capability for the most frequent communications. It could be argued that using a single network with a controllable source level could provide sufficient power savings, but the NetALS concept allows for networks that use completely different physical layers for communications. Under limitations of hop distances it is possible to make use of very significant energy savings using optical or electromagnetic underwater communications. At the same time, physical layer and frequency diversity between the networks allows for more efficient communications over the throughput limited underwater channels. Our investigation has shown that the short-range links can require significantly less power than the existing longer-range modems. The potential exists to limit the short-range links to power levels between 0.1 and 4.0 W, depending on the type of physical medium used and the length of the hops. This compares favourably with the power required for existing acoustic modems where the power is typically 10-20 W.

In this section, we discuss the three physical layer media that have been investigated. Optical methods making use of high efficiency LEDs have been considered. The use of quasi-static electric and magnetic fields allow for low-throughput underwater communications and some suggestion has been made for using high-frequency radio waves for underwater communication [7]. Finally, various components of underwater acoustic communications are being looked at.

4.1. Optical Methods

Two different experiments have been conducted at DRDC using LED light sources for underwater communications. The first experiment used a cluster of low-power blue LEDs as a transmitter and an inexpensive video camera as a receiver. The objective of these experiments was to determine practical ranges of optical communication in various water conditions. The second experiment used high-power blue LEDs as transmitters and high-speed silicon photodiodes as receivers. The objective of the second set of experiments was to test an optical beacon that is intended to assist an autonomous underwater vehicle (AUV) in locating a deployed system node and provide a short-range 1 Mbps data transfer capability.

Figure 3 shows the results for the low-power LED light projector and video camera receiver. The video camera image from the LED projector is shown at various ranges. The camera is directed downwards and is at a depth of 35 m. The LED projector is lowered from 2-23 m deeper than the camera and is directed upward toward the video camera. The results shown in Fig. 3 are for extremely turbid water and represent some of the worst viewing conditions to be expected. The results show that optical communications are practical in bad conditions at ranges of between 10-15 m with inexpensive light sources and receivers. In good conditions, ranges of several hundred meters appear to be practical.
Figure 4 shows the apparatus used for testing the optical guidance beacon and the 1 Mbps data transfer capability. A very bright, modulated light source is generated using the high-power LED sources. This modulated light beam is received by four optical sensors that generate an error signal that is intended to provide guidance for an AUV as it approaches a deployed system. Once the AUV is sufficiently close to the deployed unit, data can be transmitted bi-directionally at 1 Mbps.

![Figure 4](image)

Fig.3: Low-power LED light source viewed by a black & white digital video camera in turbid water 35 m below the surface. (a) the view at 2-m range, suspended particles visible, (b) the view at 3 m range, attached cable visible, (c) the view at 5 m, cable still visible, (d) 10 m, (e) 15 m, and (f) 23 m.

4.2. Electromagnetic methods

Electromagnetic methods, other than optical, can be used for establishing underwater communications. It is possible to use low-power, quasi-static electric and magnetic fields to provide short-range underwater communications. Two companies, Magneto-Inductive Systems Ltd. (MISL) (now part of Ultra Electronics Maritime Systems) and Wireless Fibre Systems (WFS) both have products providing underwater communications using low-frequency magnetic fields. These magnetically coupled links are extremely promising and have obvious applications for non-contact data transfer. Unfortunately, power, range, and data-rate limitations make these solutions non-ideal for the NetALS concept short-range data link.

Recently Shaw et al. [7] have published results that suggest that high-frequency radio waves can penetrate highly conductive seawater. This suggestion is of great interest as it can be verified it could potentially provide a nearly ideal solution to underwater communications. Unfortunately, the results have not to our knowledge been independently verified as of yet and tests at our laboratory by Birsan [8] did not replicate the result.
4.3. Acoustic methods

Not surprisingly underwater acoustic methods show the greatest potential for meeting the requirements of the NetALS short-range network links. The issue with acoustic networks has generally been the power requirement of the acoustic modems. DRDC efforts in short-range acoustic communications have been directed toward transducer and receiver design, coherent high-rate communications, and robust low-rate communications.

In many conditions the combination of ambient noise, absorption, and propagation effects suggest that there could be a sweet spot on the frequency band for communications. That preferred band is generally located at about 30-60 kHz. The idea is to obtain a maximum SNR for a minimal emitted source level. At DRDC, Fleming [9] has developed a simple, efficient, robust, free-flooding transducer with an extended ultrasonic bandwidth that is ideal for use in the 35-55 kHz band. Figure 5 is a photograph of the multi-mode pipe projector (MMPP) transducer used in our tests. The projector requires only a 1 Vrms drive signal to produce a source level in excess of 140 dB/1µPa@1m sufficient for 300-m communication ranges.

Receiving hydrophones for ultrasonic frequencies have also been investigated at DRDC. Working with Geospectrum a modified version of an existing hydrophone has been developed with higher capacity for lower noise and improved directivity. An extremely low-power preamplifier circuit has been developed for use with these hydrophones. Small arrays of these hydrophones have been constructed.

Energy utilization is extremely important in deployable systems and work is on-going at present to develop a practical analogue heterodyning receiver that will allow for the use of slower and less power hungry processor electronics. When a receiving array with 4 to 6 hydrophones is used, considerable energy is consumed in the front-end of the digital receiver. By employing analogue techniques we hope to reduce the clock rate and power requirements of a multi-channel receiver.

In a collaboration between the Naval Research Laboratory and DRDC, Gendron and Heard have investigated the use of coherent signaling techniques with bottom mounted transducers[10]. This work has led to data being successfully transmitted in various environments from bottomed transducers at horizontal ranges from a few hundred metres to several kilometers using the MMPP source and hydrophone receiver array. At the present stage of development the code is implemented in Matlab and is used in post-processing only. Data rates from 1-20 kbps have been demonstrated.

In order to test both electronic and software components of potential modem devices a modem test-bed has been developed. The current test-bed supports an MMPP transducer and
up to four receiver hydrophones. The unit has a powerful TI6711 digital signal processor for uploading new program codes. Figure 6 is a photograph of the test-beds prior to deployment. In operation the processor and transducers sit on the sea floor and are connected to light weight surface buoys (background) that provide a Freewave radio link that is directly wired to the sea floor units. Using these test-beds, hardware and software components can be tested in actual conditions. At present, a low-rate acoustic communications code has been tested and the plan is to implement the coherent communications algorithm in the future.

Our next step in the acoustic short-range link investigation will be to make use of a new commercial product. Teledyne has released the Compact Modem, which is compatible with the telesonar modems, but does not support the Seaweb system. DRDC is acquiring several of the compact modems for testing as the NetALS short-range link. Other manufacturers, notably LinkQuest, are now also producing low-power modems.

5. NETWORK PROTOCOLS

Protocols for operating the networks are extremely important. Seaweb provides many built-in functions and capabilities to support data transfer, network discovery, node replacement, and energy management. A similar protocol will need to be developed for the short-range network. This protocol development will be a future work effort for the NetALS concept. In the NetALS concept we are considering using frequency division to support 14 or 15 independent channels for simultaneous access to the clusterhead. Each channel would then support low-bandwidth communications for detection messages. Control and network maintenance could be relegated to a reserved channel. The complexity of the receiver and the associated processing and energy requirements will play a large part in the system design.

Additional network protocols, such as the NATO Janus protocol [11] will need to be incorporated to support interoperability and transient network nodes.
6. CONCLUSION

This paper has described some of the recent research efforts carried out by DRDC in support of underwater communication networks for deployable surveillance devices. The RDS project use of the Seaweb network has been described and the need for a short-range, low power modem network to support the NetALS concept has been described.

Optical and electromagnetic underwater communications techniques do not appear to be ideal solutions for the NetALS short-range network; however, they are applicable for data transfer to AUVs and other transient network nodes. Range and power limitations are the main drawback for these communication methods.

Ultrasonic acoustic modems appear to offer the best hope for implementation of a practical short-range network. Recent developments by several manufacturers reflect improving modem technology by the appearance of special purpose low-power modems.

REFERENCES

UNDERWATER ACOUSTIC NETWORKING DEVELOPMENT AND DEMONSTRATIONS AT WHOI

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Abstract: Work at Woods Hole in underwater acoustic networks in focused on several aspects of development with the goal of creating a general-purpose framework that facilitates research for multiple applications. These applications include fixed sensor networks for environmental or naval use, as well as mobile ad-hoc networks and mixed systems that include both stationary nodes and mobile underwater vehicles. Common to research in all of these areas is a need for a hardware and software test-bed that facilitates prototyping of approaches to networking. Our work in this area includes a complete approach: modems for the physical layer, general-purpose processor for networking and sensor interface, and a modular software environment.

Keywords: acoustic communications, telemetry, networking, sensor networks
1. INTRODUCTION

Research and development work being done in ad-hoc networking at the Woods Hole Oceanographic Institution has three major components: compact hardware development, a multi-purpose network software stack, and demonstration of specific solutions for different applications, including sensor networks.

Our work in hardware development includes a National Science Foundation effort that is focusing on construction of ten small nodes. These small nodes include a Micro-Modem, amplifier, transducer and ARM-based network processor running Linux. The user simply plugs in a USB port to program the network processor with new software or to download log files after experiments. A major objective of the project is to provide easy access to students and researchers to hardware that is compact and readily adaptable to different protocols. Thus in addition to the hardware work we have also written an interface library and a basic modular network stack.

The culmination of these two efforts will be applied to specific sensor network problems and the demonstration of network-ready sensors. Under sponsorship from the state of Massachusetts, we are building a fixed network also with approximately ten nodes that will be deployed in shallow water in the Atlantic Ocean near the semi-permanent Martha's Vineyard Cabled Observatory. A base station will be wired into the observatory for power and Internet connectivity. Sensors suitable for this oceanographic environment and amenable to acoustic connection will be interfaced to the network controller and deployed in the area, along with test nodes creating traffic to represent an ad-hoc network with varying telemetry requirements. The project aims to bring together both academic research and oceanography, depicted graphically in Figure 1.

Figure 1 Network observatory project description.
2. PHYSICAL LAYER – THE MICRO-MODEM

The physical layer for the system is based on the WHOI Micro-Modem [1], which has been developed for use with underwater vehicles and sensors over the past ten years. The system has two modes of communication, frequency-shift keying (FSK), and phase-coherent keying (PSK). The FSK mode is slow (80 bps), but it is robust and is implemented on a fixed-point processor that only requires 0.2 W. Higher data rates use coherent modulation (QPSK), and can be equally robust, but require a more sophisticated receiver. The receiver, an adaptive decision-feedback equalizer that can process multiple hydrophone channels, is accommodated on a floating-point co-processor that is only turned on when required because it draws 3 W. In Figure 2 the modem boards are shown, both the DSPs and an integrated stack including the amplifier.

3. INTEGRATED COMMUNICATIONS NODE

The approach taken for the integrated node separates the two major tasks of the system into logical units: the physical layer modem and the network processor. While there are obvious advantages to a highly-integrated system where the modem incorporates networking features, the potential complexity of those protocols and routing algorithms are such that we decided to use a very general-purpose processor for networking, and not attempt to include them in the digital signal processors. The DSPs have real-time constraints to satisfy in order to keep up with signal processing, and so separating these functions keeps the two types of software from interfering with each other. Another advantage is that the development of networking capabilities can be done at other organizations and tested with hardware-in-the-loop on the bench prior to in-water testing. It is anticipated that after a particular protocol is tested and proven for a given application that the software that implements it could be ported to the fixed-point DSP.
The block diagram of the node is shown in Figure 3. A rechargeable lithium battery that can be charged without opening the housing provides power for all of the subsystems in the bottle. The Gumstix processor [2] controls the modem and also performs the power control function on the amplifier. The amplifier is a low-power linear unit that can source up to 2 W, sufficient for close-range links in shallow water (500-1000 m). Finally, a connector is available for an external sensor and to download new software over USB.

![Figure 3 Self-contained node block diagram.](image)

The physical system is depicted in Figure 4, where the electronics and battery are contained in a small pressure housing with the transducer mounted to one cap of the pressure case. The dimensions are 7.5 cm by 32 cm (3.25 by 14 inches).

![Figure 4 Test-bed node with integrated battery and transducer.](image)

3.1. **Network Software**

Recognizing that every application has different networking requirements has motivated creation of a modular software stack that allows use of different algorithms or protocols as required [3]. The interfaces between the different modules are defined to allow interchange of software libraries, for example different medium access control implementations. The goal is to be able to quickly create a specific blend of functions to meet a given need. While the software is written with a layer that interfaces to the Micro-Modem, connections to other types of modems is also possible.

The **physical layer** interface includes commands to send data to the modem, and to query modem parameters such as the minimum transmission unit (MTU), which is the size of a data frame that can be sent. The **medium access control layer** (MAC), interfaces between the
physical layer and the next higher layer and it provides channel adjudication using a number of simple schemes including ALOHA (no control, but acknowledgement), multiple-access with collision avoidance (MACA), and more recently, time-slotted access. Finally, the logical link control layer (LLC) provides networking functions, including neighbour management.

While there are myriad approaches to ad-hoc networking, our interest focuses on supporting networks that include a mixture of fixed and mobile nodes. The oceanographic applications that are being considered include groups of sensors around a gateway (cabled or buoyed), or lines of sensor-equipped network nodes that relay data along a path (similar to SEAWEB). Important networking features include discovery and routing.

Ad-hoc capability is desirable for a number of reasons, and it is useful for both fixed and mobile systems. While it might appear that fixed systems do not need dynamic capability, in reality acoustic conditions are subject to change, and links between nodes that exist at one point in time can disappear a few minutes later. For example, diurnal effects impact both propagation paths and noise, and an intermediate hop may be required occasionally, but not all the time. Other motivations for ad-hoc networks include sensor drop-in around a hub and to support AUVs working around a fixed network.

Motivating papers [4] [5] that describe a desirable philosophy for modularizing ad-hoc wireless networks was the basis for our approach in developing the software architecture, though our implementation is different. The most important concept is that of providing for different types of networks without having to make structural changes to software.

4. APPLICATION: COASTAL OBSERVATORY

Under sponsorship from the State of Massachusetts John Adams Mass Tech Collaborative a medium-sized acoustic network is being developed and will be deployed south of the island of Martha’s Vineyard in Atlantic Ocean. The system consists of a cabled base station, plus several gateway buoys and sub-sea acoustic nodes that transmit data back to the base station via radio (Figure 5). The objective of this project is to demonstrate a multi-node fixed network with multi-hop relay from sensors to multiple gateways. It will also adapt to the ambient acoustic conditions, which vary with sea-state.

While our past work has relied heavily on simple polling strategies and time windows, our goal for this project includes demonstration of demand-driven communication. This will take advantage of routing that is determined dynamically and will thus adapt to propagation conditions. Setting up routes is done on demand, but takes advantage of any network traffic that is received. If no routes are known when a node needs to send data, a beacon is transmitted to any potential neighbours. They respond with replies that include their neighbour information and the best route to the destination is then selected.

5. CONCLUSIONS

The use of ad-hoc protocols for undersea networks is highly desirable because it can automate some of the issues associated with setting up links in unreliable channels. The development of these systems is currently hampered by a lack of in-water testing facilities, slowing down feedback of the practical observations to protocol developers that often do not understand the differences between terrestrial wireless and acoustic channels. Our goal is to increase the accessibility of underwater networking hardware to the research community and to further our own development of specific oceanographic applications.
6. ACKNOWLEDGEMENTS

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Structured Session 15

Ocean Acoustic Tomography – Applications to Shallow Seas and Benthic and Terrestrial Waters

Organizers: Arata Kaneko & Jean-Pierre Hermand
RIVER ACOUSTIC TOMOGRAPHY FOR CONTINUOUS MEASUREMENT OF WATER DISCHARGE

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Abstract: River discharge is an important hydrological quantity on river and coastal planning/management, control of water resources, etc. In the present study, water discharge in a shallow tidal channel, which is 120 m wide and 0.3~3 m deep, was continuously measured using a new river acoustic tomography (RAT) system. A couple of transducers of central frequency 30 kHz were installed diagonally across the channel. The system has a few noticeable functions represented by the accurate measurement of travel time using the GPS clock and the attainment of high signal-to-noise ratio as a result of transmission signal modulation by the 10th order M-sequence. The RAT, operated at the shallow tidal channel with large changes of water depth and salinity, successfully measured the cross-sectional mean velocity over a long duration. The agreement between RAT and ADCP on water discharge was satisfactory. Thus, the RAT is promising method for continuous measurement of streamflow.

Keywords: Streamflow, acoustic tomography, water velocity, tidal channel
1. INTRODUCTION

River discharge is an important hydrological quantity on river and coastal planning/management, control of water resources, etc. Therefore, it is a pressing issue to establish the measurement method/technology of water discharge. However, it is very difficult to measure cross-sectional averaged velocity in complex flows such as tidal estuaries or during extreme hydrologic events.

For continuous measurement of water discharge, a few different equipments are available, e.g., acoustic velocity meters (AVMs), horizontal acoustic Doppler current profilers (H-ADCPs), etc. ([1], [2]). The main drawback of previously presented methods is that often number of velocity sample points in the cross-section of stream is not sufficient to estimate cross-sectional averaged velocity. Although several methods are introduced to estimate the velocity distribution, e.g. [3], the results are disputable in complex flow field such as stratified tidal flows. Thus, an innovative method and or equipment are required for continuous measurement of water discharge in tidal estuaries.

In the present study, a river acoustic tomography (RAT) system is developed and utilized to measure cross-sectional mean velocities in the Ota River diversion channel with large changes of water depth and salinity. The RAT system have advantages compared to competing techniques, namely accurate measurement of travel time using GPS clock, high signal-to-noise ratio due to modulation by 10th order M-sequence.

2. MEASUREMENT PRINCIPLES

The applied basic principle is similar to what is used in an acoustic velocity meter (AVM), in other words the cross-sectional averaged velocity is calculated using “time of travel method” [4]. In fact, the RAT system is able to estimate cross-sectional averaged velocity using sound paths that cover the section.

The travel time along the reciprocal ray path $r^\pm$ between a couple of transducers in the flowing medium is formulated as:

$$t_i^\pm = \int_{r_i^\pm} \frac{ds}{c(x,y) \pm u(x,y) \cdot n} \quad (i = 1, 2, \ldots, M)$$

(1)

where $+/-$ represent the positive/negative direction from one transducer to another. $c$ is the sound speed, $ds$ the increment of arc length measured along the ray, $u$ the water velocity, $n$ the unit vector along the ray and $i$ the number of ray. The path integrals are taken along rays. We assume that the two-way path geometry is reciprocal and . The two-way travel time difference may be expressed as:

$$\Delta t_i = (t_i^- - t_i^+) = \int_{r_i^\pm} \frac{2 \ u \cdot n}{c^2 - (u \cdot n)^2} \ ds \approx \int_{r_i^\pm} \frac{2 \ u \cdot n}{c^2} \ ds \approx \frac{2L_i u_m}{c_m^2}$$

(2)

where $u_m$ and $c_m$ are the range averaged water velocity and the sound speed along the ray path, respectively.
\[ u_{m_i} = \frac{1}{L_i} \int_{R_i} \mathbf{u} \cdot \mathbf{n} \, ds \]  \hspace{1cm} (3)

\[ c_{m_i} = \frac{1}{L_i} \int_{R_i} c \, ds \]  \hspace{1cm} (4)

where \( L_i \) is the length of ray path. \( c_{m_i} \) is calculated from

\[ t_{m_i} = \frac{1}{2} (t_i^- + t_i^+) = \int_{R_i} \frac{c}{c^2 - (\mathbf{u} \cdot \mathbf{n})^2} \, ds \approx \int_{R_i} \frac{1}{c} \, ds = \frac{L_i}{c_{m_i}} \]  \hspace{1cm} (5)

The cross-sectional averaged velocity \( v_m \) is estimated from

\[ v_m = \frac{u_m}{\cos \theta} = \frac{1}{\cos \theta} \sum_{i=1}^{M} u_{m_i} = \frac{1}{\cos \theta} \sum_{i=1}^{M} \frac{c_{m_i}^2}{L_i} \Delta t_i \]  \hspace{1cm} (6)

where \( u_{m_i} \) is averaged velocity along the ray path, and \( \theta \) the angle between ray path and streamline.

In order to estimate the cross-sectional averaged velocity \( v_m \), the ray paths have to get through all layers between bottom and water surface. If the sound speed has inhomogeneous distribution in water, rays draw a curve obeying the Snell’s law of refraction. In this paper, ray simulations were implemented by solving the following differential equations [5]:

\[ \frac{d\varphi}{dr} = \frac{\partial c}{\partial r} \tan \varphi - \frac{\partial c}{\partial z} \frac{1}{c} \]  \hspace{1cm} (7a)

\[ \frac{dz}{dr} = \tan \varphi \]  \hspace{1cm} (7b)

\[ \frac{dt}{dr} = \frac{\sec \varphi}{c} \]  \hspace{1cm} (7c)

where \( \varphi \) is angle of the ray regarding to horizontal axis \( r \), \( z \) is vertical coordinate, and \( t \) is the time. Here, the sound speed \( c \) was estimated by Medwin’s formula as a function of temperature \( T \) (°C), salinity \( S \), and depth \( D \) (m) [6].

3. EXPERIMENTAL SITE AND METHOD

A RAT experiment was carried out during June 3 to July 17, 2008 at the Ota River
diversion channel (Figs. 1 and 2). The Ota River bifurcates into two main branches at about 9 km upstream from the river mouth. The upstream border of tidal compartment in the Ota River estuary is about 13 km upstream far from the mouth. The tidal range of an extreme spring tide at the mouth is about 4 m. Freshwater runoff is usually limited by the Gion sluice gates that are located at the bifurcation place. Only one sluice gate is opened slightly in order to make a cross-sectional area of stream is 32 m×0.3 m to spill the flow.

The experimental site was located at 246 m downstream from the Gion sluice gates as shown in Fig. 2. The Ota River 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 broad-band transducers were installed diagonally across the channel as shown in Fig. 2. The central frequency of transducers was 30 kHz, angle between sound pass and stream direction $\theta$ was 30 degrees, transducers were mounted at the height of 0.2 m above the bottom. The altitudes of left and right transducers were −0.46 m and −0.7 m, respectively. The sound pulses of the RAT system were simultaneously transmitted from the omni-directional transducers every minute triggered by a GPS clock.

Three moored ADCPs were used to validate the velocity data of the RAT system. Three ADCPs arranged along the Gion Sluices in a way that each two ADCPs were 30 m far from each other while the central ADCP location aligned with the river centerline. The distance between each ADCP and the Gion Sluice was 59.1 m. Vertical distribution of water temperature and salinity were measured every 10 minutes by CT sensors attached to the pier of the Gion Bridge at 40 m from the left bank as shown in Fig. 2. Besides, cross-sectional distributions of temperature and salinity were measured by CTD casts from the Gion Bridge. Transverse interval of the CTD casts was 20 m and crossing time was about 10 minutes.
4. RESULTS AND DISCUSSIONS

4.1. Ray tracing

Fig. 3 shows distributions of the sound speed and results of the ray simulation just after HWS and just before LWS as typical examples. The tiny effect of current is not considered in the present ray simulation. The sound speed was calculated from projected data of the CTD casted from the Gion Bridge. The salinity increases with depth, as a result of this distribution sound speed ranged from 1515 m/s in deeper layers to 1485 m/s near the surface.

Most of the time, the sound paths cover the cross-section as shown in Figure 3 (a). Unfortunately, sometimes a near-bed established salt wedge caused the sound paths to be reflected; consequently, a part of sound paths were not able to penetrate into the lower layers, e.g. in Fig. 3 (b) typical condition under strong stratification is depicted. In this case, the cross-sectional averaged velocity is somewhat overestimated by the RAT system.
So, the cross-sectional averaged velocity should be modified using velocity distribution of two layers flow when there is a salt wedge under the transducer.

4.2. Correlation wave forms

The cross-correlation waveforms of signals transmitted from the upstream and downstream transducers are shown typically in Fig. 4. The cross-correlation forms are plotted every 5 minutes. The cross-correlations obtained from the both sides are similar form. This suggests that the two-way path geometry is reciprocal. The broad peaks are composed of multi arrival rays. It seems that the single peak is also composed of multi arrival rays because the experimental site is shallow. The mean arrival time changes because of salinity change. Sometimes there are not clear peaks. In the present study, the two-way travel time difference is calculated when the SN ratio is over 14 dB.

4.3. Comparison between RAT and ADCP

Fig. 5 shows temporal variations of the cross-sectional averaged velocity $v_m$ and the water level $H$. The thick red line denotes the cross-sectional averaged velocity deduced from 3 moored ADCPs. The broken line denotes the water level. Unfortunately, we cannot
discuss the accuracy of the RAT system using the results of ADCPs because the accuracy of velocity data from three ADCPs is low; the strong nonuniformity of flow at the observation site needs more ADCPs. However, it is found in Fig. 5 that the difference between velocities acquired by the RAT system and ADCPs is small. Thus, we can deem the cross-sectional mean velocity obtained from RAT system fulfils an acceptable compliance with the results derived from the ADCPs.

4.4. Temporal variation of water discharge

The water discharge was calculated by the RAT system from:

\[ Q = A(H) v_m \sin \theta = A(H) u_m \tan \theta \]

where \( A \) is the cross-sectional area in which sound paths travel, \( H \) water level.

Temporal variations of the water levels at Yaguchi and Gion, and flow discharge are illustrated in Fig. 6 for 44 days. As it can be seen, limiting the runoff from the Gion Gates led to dominance of tidal effects. Long duration measurement of the discharge are done successfully. Thus, employing RAT system is a promising method for continuous measurement of the discharge. The values modified using velocity distribution of two-layers flow were averagely about 10% smaller than values without correction for the observation period. The trend of discharge, which is denoted by thick line, corresponds with the change of water level at the Yaguchi gauging station.
5. CONCLUSIONS

A river acoustic tomography (RAT) system that utilizes a GPS clock and 10th order M-sequence modulation was developed and applied to a shallow tidal river with a complex flow field. The RAT system composed of a couple of transducers, which are installed diagonally across the channel, was able to measure the cross-sectional mean velocity. The sufficiently high signal-to-noise ratio was obtained owing to the 10th order M-sequence modulation. Thus, we believe that the RAT system works well even throughout flood events in which turbidity and sound noise are very high.

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An acoustic tomography experiment in the Luzon Strait with strong currents and tides

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Abstract: A deep-sea acoustic tomography experiment with a triangular configuration of side length about 40km (Stn.s M1, M2 and M3) and transducer depths about 800m was carried out at the northern part of the Luzon Strait, during April to October 2008 where the Kuroshio Current passes and also quite strong internal tides and waves occur. The four-month reciprocal travel time data were successfully acquired between Stn. M1 and M2 while there were no data at St. M3 because of the failed recovery of the M3 mooring line. Two ray paths, passing near the underwater sound channel at depth about 900m and drawing an upward convex refracted curve between the depth 100m and the sea bottom are identified as ones with different travel times in the received data. The inversion analyses of current velocity and sound speed, which uses the travel time data for the two rays, are performed for three depth layers, 0-400m (upper), 400-800m (middle) and 800-1200m (lower). The results show that the current velocity is rapidly decreased from the upper to lower, and internal tide variability with periods of day to fortnight is quite prominent.

Keywords: ocean acoustic tomography, Luzon Strait, ocean current, internal tides
1. INTRODUCTION

The Luzon Strait is located at the western edge of the North Pacific subtropical front. ENSO (El Niño and Southern Oscillation) related variability occurring in the tropical Pacific is directly transferred to the strait through the meridional shift and transport change of the North Equatorial Current (NEC)\cite{1},\cite{2}. Also the Luzon Strait throughflow, a branch of the Kuroshio, produce a significant source not only to Kuroshio variability in the East China Sea downstream of the strait, but also to the climate variability in the South China Sea. The volume and heat exchange between the western North Pacific and South China Sea by the strait throughflow also serve to be a trigger to global climate changes through the air-sea interaction there. On the other hand, there are intense internal tide activities in the Luzon Strait because of the interaction of barotropic tides with sills at the depth (800-900m) of main thermocline \cite{3}. Strong turbulent mixing, generated by the breaking of internal tides and the associated internal waves, increases eddy viscosity and diffusivity and modifies the parameterization of the throughflow and the circulation model on the downstream side.

The Luzon Strait is a key place to study the climate variability, caused in the East Asian countries in relation to the Kuroshio, but still a region where the direct measurement of current is lacked until the recent year \cite{4}.

The Luzon strait ocean acoustic tomography (OAT) experiment is carried out to increase information on current and wave variability in the Luzon Strait. Also internal tides are targets of this study.

2. SITES AND METHODS

The experimental site is at the northern part of the Luzon Strait south of Taiwan, where the northward flowing Kuroshio begins to turn to the east of Taiwan and its small branch flows toward the Taiwan Strait (Fig. 1).

The mooring lines, equipped with an 800Hz OAT transducer at depth 800m, were deployed at the stations M1, M2 and M3. The deployment and recovery of the three mooring lines were performed onboard the R. V. Dong Fang Hong II of the Ocean University of China during April 23 to 25, 2008 and October 5 to 6, 2008, respectively. Notice that not the broad-band, but the narrow-band transducers of frequency range 5.12 to 12.5 Hz are used in this experiment to reduce largely the power consumption in a long-term operation \cite{5}. The large cycle/digit value of 24 is selected to transmit the 12th order M sequence from the narrow-band transducer. One period of the M sequence (120s) is transmitted every 8 hours. The transducers are produced as part of the SOFAR (SOund Fixing and Ranging) float. The mooring positions and depths are summarized in Table 1. Special attention is paid to clock accuracy. The internal quartz clock is synchronized by the GPS clock prior to the deployment into the ocean. In the subsurface, the more accurate Rubidium clock runs for the first 24 hours together with the less accurate internal quartz clock, and the drift rate of quartz clock per second is estimated by comparing the oscillation number of the quartz and Rubidium clocks during the 24 hours. The following sound transmission is done after the clock correction by the clock drift rate. In the recovery, the drift of the internal quartz clock is also measured in comparison with the GPS clock.

The upward-looking moored ADCP is set at the top of the subsurface mooring lines M1 and M2. The RDI ADCP at M1 and M2 is operated at the working frequency of 150 kHz for M1 and 75kHz for M2. The deployment depth of ADCP is 300m for M1 and 500m for M2. CTD casts were performed around the mooring sites in the deployment and recovery cruises.
Table 1  Mooring sites and depths

<table>
<thead>
<tr>
<th>Station</th>
<th>Latitude (N)</th>
<th>Longitude (E)</th>
<th>Depth (m)</th>
</tr>
</thead>
<tbody>
<tr>
<td>M1</td>
<td>21°18.377'</td>
<td>120°35.872'</td>
<td>970</td>
</tr>
<tr>
<td>M2</td>
<td>20°59.961'</td>
<td>120°30.332'</td>
<td>1633</td>
</tr>
<tr>
<td>M3</td>
<td>21°00.385'</td>
<td>120°52.315'</td>
<td>970</td>
</tr>
</tbody>
</table>

3. SOUND TRANSMISSION SIMULATION

Sound transmission simulation is done by using the sound speed profile as shown in Fig.2. This profile is constructed by averaging the CTD data, obtained at the stations M1 and M2 in April. The axis of the underwater sound channel (the minimum point of sound speed) is seen at depth about 900-1000m.

The sound transmission between the stations M1 and M2 are simulated by means of the ray tracing method, using the above CTD data (Fig.3). The correlation diagram between the launch angle and the travel time is shown in Fig.4. A number of simulation rays are not identified as separated rays due to complicated bottom profiles, but there are a group of bottom-reflected rays which make travel times between 23.8s and 24.1s. On the other hand, the surface-reflected rays have travel times around 24.9s. Considering the inter-comparison with the actual transmission data described later, we focus the ray-① and ray-② with travel times near 23.9s and 24.05s, respectively. The ray-① passes around the axis of the underwater sound channel, and the ray-② makes an upward convex ray, passing the depth range from 100m to 1200m. The vertical section between the stations M1 and M2 is segmented into three depth layers of 0-400m, 400-800m and 800-1200m. The length of these two rays crossing each depth layer will be used in the inverse analysis of travel time data to get the three-layered profiles of velocity and sound speed.
Fig. 2 Vertical profiles of temperature (T), salinity (S) and sound speed (C), constructed by averaging the CTD data at M1 and M2.

Fig. 3 Simulation result of the sound transmission between M1 and M2

Fig. 4 Correlation plot of the travel time and the launch angle
4. DATA ANALYSES
4.1 Received data and clock correction

The received data are cross-correlated with the M sequence, used in the transmission, to increase remarkably the signal-to-noise ratio (SNR). Typical examples of received waveforms, obtained on May 2, 2008, are shown in Fig. 5. The received waveforms make multi arrival peaks in the travel time duration of 23.9s to 24.4s. Among the arrival peaks, Ray-① and Ray-② are identified for stations M1 and M2 as indicated with arrows. The stack diagrams of the received waveforms for the whole observation period are shown in Fig. 6. In this figure, time is growing from lower to upper. The temporal variation of multi arrival peaks is quite large due to internal tide activity. Furthermore there is a persistent drift of the internal clock for both the M1 and M2 systems (Fig. 6a). The M1 clock is continuously delayed while the M2 clock goes ahead of the correct time, as seen in the diagram.

The weak current is reasonably expected in the depth range where Ray-① passes. Under this assumption, the travel time difference for Ray-① should be nearly zero. In this paper, the clock drift is corrected, considering the special aspect of no travel time difference accompanied by the zero current. The earliest arrival time is determined at the up-slope point of the first arrival peak over SNR=10 in the arrival peak for Ray-①. This results in the clock correction, as provided by the following polynomial:

$$\Delta t = 0.00287 - 5.123 \times 10^{-6} t + 2.433 \times 10^{-7} t^2 - 1.236 \times 10^{-10} t^3$$
$$+ 3.072 \times 10^{-14} t^4 - 2.869 \times 10^{-18} t^5$$

where t denotes the time (hour) elapsed from the deployment of the system into the ocean. In the stack diagram after clock correction, the clock drift is diminished significantly as seen in Fig. 6b.

![Fig. 5 Typical examples of the received waveform obtained on May 2, 2008. The upper and lower panels show the waveforms at M1 and M2, respectively.](image-url)
4.2 Inverse analysis

The generalized inversion is applied to reconstruct the three-layered velocities and sound speeds deviation from the travel time difference data and mean travel time data, respectively [6]. The time plots of the three-layered velocities are shown in Figs.7a and 7b after the 2-day smoothing (ensemble average of seven data) and the 14-day smoothing (ensemble average of forty three data), respectively. The mean of the three-layered velocities and the depth-averaged velocity for Ray-② are also shown in the figures to confirm the validity of the
inverse analysis. Both the data are in good agreement and their mean difference is as large as 0.01 m/s. This is because Ray-② passes the depth-layer from 100m to 1200m which covers almost all depth layers in the inversion.

In Fig. 7a, the upper-layer (0-400m) velocity varies in an unnatural magnitude in the range of -1.5 ~ +2.0 m/s, including significant errors. This is caused by the shortage of ensemble number. The developed internal tides require a larger number of ensembles. The errors are remarkably reduced by taking an ensemble average of forty three (Fig.7b). In this figure, the upper-layer velocity increases with time, changing from 0.2 m/s at the beginning of May to 0.6 m/s at the middle of June. During this period, the velocity is remarkably decreased with depth. The mean velocity at the middle and lower layers is about 0.15 m/s and 0.10 m/s, respectively. The upper layer velocity varies in the range of -0.2 m/s to -0.1 m/s in July and August while the middle- and lower-layer velocities keep constant around 0.1 m/s.

Fig. 7 Time plots of the three-layered velocities for the whole observation period, smoothed through the 2-day and 14-day running means.
The power spectral density is calculated by using the 2-day interval data of velocity. The power spectral diagrams are shown in Fig. 8 for the upper, middle and lower layers. The spectral diagram for the mean velocity along Ray-② is also presented for comparison with the inversion results. The ray identification in the reciprocal direction is sometimes so difficult due to the missing of arrival peaks for Ray-① or Ray-②. This causes the data lacking of as large as about 60%. The lacked data are converted to the continuous data through a linear interpolation which uses the neighboring data. The spectral analyses are severely disturbed by the data lacking and linear interpolation. In spite of the bad data condition, significant spectral peaks are seen at 12 days and 30 days.

![Power spectral density diagrams of the two-day interval data, obtained for the three-depth layers](image)

**Fig. 8** Power spectral density diagrams of the two-day interval data, obtained for the three-depth layers

### 5. SUMMARY AND DISCUSSION

The ocean acoustic tomography experiment with a triangular arrangement was carried out at the northern part of the Luzon Strait south of Taiwan during April to October 2008. The number of the acoustic stations is not large enough to get a 3D mapping of velocity and sound speed. Then this experimental set-up might be called the acoustic profiler of velocity and sound speed rather than the acoustic tomography. Another noticeable aspect of this experiment is that instead of the broad-band transducer the narrow-band one is operated to reduce the power consumption and make the system cost-effective in a long-term operation. The transmitter of the SOFAR floats is here applied.

The four-month travel time data are successfully acquired between the stations M1 and M2. However, the correlation waveform of received signals is severely disturbed by strong
internal tides of periods half day to thirty days. Also the simulated ray paths are not well separated over the whole depth because of the complicated bottom profile. As a result, only two ray paths (Ray-① and Ray-②) are identified as rays which construct the separated travel times. The inverse calculation for the three depth-layers is performed, using the travel time data of Ray-① and Ray-②. The upper-layer velocity varies in the range of 0.2 to 0.6 m/s for the first two months and is remarkably decreased in the later two months. The middle- and lower-layered velocities vary in the range of 0.1 to 0.2m/s over the observation period. The 12-day and 30-day period spectral peaks are seen in the power spectral density diagrams instead of the fortnight period peak. This is caused by the inappropriateness of linear interpolation to the large region of data lacking.

The usefulness of the acoustic profiling of velocity and sound speed by a triangular system is verified in this experiment. It is proposed that the profiling observation across the Luzon Strait can be carried out in the near future by an acoustic profiler network with a sequence of connected triangles.

ACKNOWLEDGEMENTS
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REFERENCES
**EFFECT OF WARM EDDY ON LOW-FREQUENCY SOUND PROPAGATION IN THE EAST/JAPAN SEA**

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**Abstract:** It has been known that sound waves in the sea generally propagate under the influence of sea surface wave, water depth, sound speed profile, sea floor sediment, etc. In particular, an abrupt change of the sound speed profile with depth in an eddy can greatly affect the sound propagation. Many studies on the sound propagation through eddy and oceanic front have been done in the Gulf Stream, the East Australian Current, the Western Greenland Sea, etc. Eddies are frequently generated at the polar front of the East/Japan Sea near the Korean Peninsula. A warm eddy with diameter of about 200 km is often observed and sound speed profile is frequently affected within about 300 m of water depth at the centre by the eddy in the East/Japan Sea. This sound speed variation can affect long-range propagation of sound wave. The characteristics of the low-frequency sound propagation across the observed eddy are investigated by a sound propagation model. As a result, it is ascertained that the low-frequency sound propagation is affected by the warm eddy and the phenomena are dominated in the upper ocean. And the low-frequency sound propagation from the eddy centre to the eddy edge is more affected by the eddy in comparison with the sound propagation from the eddy edge to the eddy centre.

**Keywords:** Low-frequency sound propagation, Warm eddy, Sound speed profile, East/Japan Sea
INTRODUCTION

An eddy frequently observed in the ocean is a physical phenomenon. In general an eddy is characterized by its rotation direction. There are two types of eddies, that is, a cyclonic eddy (cold eddy) which rotates counter-clockwise and an anti-cyclonic eddy (warm eddy) which rotates clockwise in the northern hemisphere. An eddy in the ocean can change strong temperature structure with vertically and horizontally. An abrupt change of the sound speed structure in an eddy can greatly affect sound propagation. Many studies on the sound propagation through eddy and oceanic front have been done in the Gulf Stream, the East Australian Current, and the Western Greenland Sea by propagation experiments and numerical calculations [1]-[6]. Eddies are frequently generated at the polar front formed by the East Korea Warm Water and the North Korea Cold Current in the East/Japan Sea near the Korean Peninsula. A warm eddy with diameter of about 200 km is often observed and sound speed profile is frequently affected within about 300 m of water depth at the eddy centre by it in the East/Japan Sea. The characteristics of the low-frequency sound propagation across the eddy observed during the sound propagation experiment are investigated by a sound propagation model to understand influence of the warm eddy on the sound propagation.

SITE AND METHODS

A site map for the sound propagation is shown in Fig. 1. The propagation site locates between the eastern middle part of the Korean Peninsula and the Ullungdo in the East/Japan Sea. The total propagation range is about 75 km. The Station EC5 is nearly located at a centre of the warm eddy and the Station EC9 is nearly located at an edge of the warm eddy.

Fig. 1. Site map for sound propagation.

The HARCAM (Hodgson And RAM Composite Acoustic Model) [7] is used in this study as a sound propagation model. The HARCAM is a kind of range-dependent ray program. The measured data during the sound propagation experiment conducted in October 1996 [8] are used as input parameters of environment for the propagation model. That is, it is used that wind
speed 5 m/s, range-dependent sound speed profiles, and mixture sediment of sand-silt-clay of sound speed 1530 m/s, density 1583 kg/m$^3$, and 50 m thickness.

Validity of the propagation model results is confirmed by comparison between calculated results with measured results on propagation loss of sound frequency 100 Hz. Acoustic rays and propagation losses are calculated by the model with conditions of the eddy present and propagation direction from eddy centre to eddy edge. They are also calculated with conditions of the eddy present and propagation direction from eddy edge to eddy centre. And they are calculated by the model with conditions of the eddy absent, eddy centre to eddy edge propagation and reverse propagation, respectively. Then the acoustic rays and the propagation losses with the condition of eddy present are compared to those with the condition of eddy absent.

RESULTS

Figure 2 shows a water depth profile along the sound propagation track. The depths are between 1500 m and 2300 m. The depth is about 1500 m at the Station EC5 of the eddy centre and the depth is about 2150 m at the Station EC9 of the eddy edge. Sound speed profiles and sound speed contours are shown in Fig. 3. Those are obtained by CTD (conductivity, temperature, and depth) casts done at 5 stations with 18.52 km intervals during the sound propagation experiment conducted in autumn season. Sonic layer depth and sound channel axis are also appeared in the figure. The sonic layer depths are not variable with range because those are between 20 m and 30 m. However, the sound channel axes are variable with range because those are between 220 m and 350 m. The sound channel axis is the deepest at the Station EC5 of the warm eddy centre, but the sound channel axis is the shallowest at the Station EC9 of the warm eddy edge. The sound speed profiles are a little changed in the upper ocean under the influence of the warm eddy. That is, the sound speed profile is affected within about 300 m of water depth at the centre by the warm eddy.

![Fig. 2. Water depth profile along sound propagation track.](image-url)
Fig. 3. Sound speed profiles (upper) and sound speed contours within 500 m depth (lower).

Figure 4 shows acoustic ray diagrams in case of sound wave propagated from the eddy centre to the eddy edge. Here, ray departure angles from the source are between -20° and +20° with 2° intervals. Source depths are 18 m and 244 m, respectively. In case of eddy present, the range-dependent sound speed profiles are used to calculate the acoustic ray by the model. Acoustic ray diagrams in case of eddy absent are also shown in the figure. At that time single profile of the sound speed measured at the Station EC9 of the eddy edge is used to calculate the acoustic ray by the model. According to Fig. 4, the acoustic rays in case of the eddy present are quite different from the rays in case of the eddy absent and especially such phenomena are more remarkable at the source depth 244 m than the source depth 18 m. This is occurred because variation of sound speed gradients by the eddy is bigger around the depth 244 m than the depth 18 m. Figure 5 shows acoustic ray diagrams in case of sound wave propagated from the eddy edge to the eddy centre. Difference of the acoustic rays between the case of the eddy present and the case of eddy absent are not so big in Fig. 5 as in Fig. 4.

Figure 6 shows contours of propagation loss calculated by the acoustic model in case of sound wave propagated from the eddy centre to the eddy edge. Here, sound frequency is 100 Hz and source depths are 18 m and 244 m, respectively. In case of the eddy present,
Fig. 4. Acoustic ray diagrams in case of propagation from eddy centre to eddy edge.
Fig. 5. Acoustic ray diagrams in case of propagation from eddy edge to eddy centre.

Fig. 6. Propagation loss contours of frequency 100 Hz in case of propagation from eddy centre to eddy edge.
Fig. 7. Propagation loss contours of frequency 100 Hz in case of propagation from eddy edge to eddy centre.

Fig. 8. Contours of difference between propagation loss with eddy and propagation loss without eddy.

the range-dependent sound speed profiles are used to calculate the propagation loss by the model. Contours of propagation loss in case of the eddy absent are also shown in the figure. At that time single profile of the sound speed measured at the eddy edge is used to calculate the propagation loss by the model. According to Fig. 6, the propagation loss contours in case
of the eddy present are quite different from the propagation loss contours in case of the eddy absent and especially such phenomena are more remarkable at the source depth 244 m than the source depth 18 m. Figure 7 shows contours of propagation loss calculated by the model in case of sound wave propagated from the eddy edge to the eddy centre. The propagation loss contours in case of the eddy present are not so different from the propagation loss contours in case of the eddy present in Fig. 7 as in Fig. 6.

Figure 8 shows contours of difference between the propagation loss with the eddy present and the propagation loss with the eddy absent. Here sound frequency is 100 Hz and source depths are 18 m and 244 m, respectively. The contours of propagation loss difference are variable within ±15 dB with depth and range. Such phenomena are more remarkable at the source depth 244 m than the source depth 18 m. And those are more dominant at the propagation from the eddy centre to the eddy edge than the propagation from the eddy edge to the eddy centre.

SUMMARY

Eddies are frequently generated at the polar front of the East/Japan Sea near the Korean Peninsula. The warm eddy with diameter of about 200 km is often observed and sound speed profile is frequently affected within about 300 m of water depth at the eddy centre by it in the East/Japan Sea. The acoustic rays and the propagation losses are investigated by a range-dependent propagation ray model in order to understand the influence of the warm eddy on the low-frequency sound propagation. The low-frequency sound propagation is affected by the warm eddy and it is dominant in the upper ocean. And the sound wave propagated from the eddy centre to the eddy edge is more affected by the warm eddy in comparison with the sound wave propagated from the eddy edge to the eddy centre.

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http://www.oad.tv/.
AN APPROACH TO THE INVERSION OF INTERNAL WAVE PARAMETERS

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Abstract: Due to the large deviation of sound speed fields caused by internal waves, the sound propagation through this range-dependent medium induces an energy exchange between different acoustic normal modes. Two kinds of methods are proposed to reconstruct the internal solitary wave (ISW) parameters: phase speed and wavelength. First, in case of internal waves moving between an acoustic source and receiver, the time series of the mode amplitude behave like some quasi-period phenomena, and the periods are described as a product of the ISW moving-speed and the difference of normal mode wave number. The ISW speed is inverted by means of the spectrum analysis of the time series of mode amplitude. Second, the mode coupling only occurs when the wave number difference between the two coupling modes is equal to the wave number of the ISW spectrum, serving to estimate the ISW wavelength. The simulation is performed to evaluate the above inversion methods, setting an internal solitary wave or an internal solitary wave packet in the simulation domain. The inversion errors due to the wave phase speed are about 5% and 10% for the flat and slope topographies, respectively. The lower the sound frequency, the more accurate the inversion result. When the suitable resonance frequency is adopted, the inversion errors of wavelength are smaller than 10%. The inversion error is increased with the increasing amplitude of internal solitons, also the error is changed with the number of solitons included in the packet.

Keywords: Internal solitary wave, wave parameters, mode coupling, inversion
1. **INTRODUCTION**

Internal waves (IWs) often occur at interfaces between water layers of different density by tidal currents, surface waves, currents over abrupt topography, or wind blowing, etc.\(^1\) The IWs play an important role in water mixing and sediment transport. Internal solitary waves, a kind of nonlinear phenomena, have been observed in the relatively shallow seas such as Yellow Sea, the Luzon Strait at the boundary of the Pacific and the South China Sea, the Australian North West Shelf, and so on.

Zhou et al.\(^2\) reported the anomalous, frequency-dependent transmission loss of sound due to strong internal solitons. Many ocean acoustic transmission experiments have shown that the travelling sound can be greatly damped by shallow water internal waves. The result of SWARM’95 experiment, which was conducted off the New Jersey in 1995\(^3\) showed the intensity fluctuation of about 7dB. Rubenstein\(^4\) showed relatively small sound variability from the experimental results, obtained in the Gulf of Mexico, but that variability still had a strong correlation with internal solitons.

Besides the fruitful field experiment, many researchers have focused on the study of acoustic scattering by internal wave. By considering simple soliton model, Zhou et al\(^2\) concluded that the anomalous transmission loss was due to strong acoustic mode coupling caused by an interaction between acoustic waves and internal waves (Resonant interaction). Rouseff\(^5\) formulated the acoustic normal mode coupling equation, applicable to the shallow water internal wave propagation by considering one-way coupled mode.

A number of observation techniques such as moored sensors, towed sensors and remote sensing are proposed to measure internal waves. A number of mooring lines, equipped with point sensors, should be deployed at appropriate distances in order to get the spatial information of ocean processes. Remote sensing by aircrafts and satellites provides another method to monitor subsurface IWs through the sea surface deformation caused by IWs. It is the most important and difficult issue to relate the surface deformation and the IW properties. Only sound can propagate over a long distance in water. Then sound transmission is quite an effective method to monitor the ocean environment over a wide area by using acoustic signal.

The purpose of this paper is to identify internal wave parameters by the sound transmission analysis, based on the normal mode method. The arrangement of this paper is as follows. In section 2 we describe the acoustic normal mode coupling formula and the resonant condition. The simulation results are presented in section 3 to clarify the validity of the present inversion approach. Finally section 4 is reserved for the conclusion and discussion.

2. **MODEL OF ACOUSTIC COUPLING MODE AND RESONANT CONDITION**

2.1. **Model of coupled acoustic mode**

The derivation for the range-independent situation in the acoustic normal mode equations can be frequently found in the classical textbooks on underwater sound transmission. By considering one-way coupling mode\(^6\), Rouseff\(^5\) formulated the acoustic normal mode coupling equation in the shallow water internal wave environment, and by which Liu et al\(^7\) executed the inverse analysis for the normal mode coupling coefficient matrix. Here we shall introduce the brief derivation of the acoustic normal mode coupling equation for the range-dependent environment.

In the range-independent problem, supposed a time-harmonic \(\exp(-i\omega t)\) located at depth
$z = z_s$, the acoustic field can be derived by the summation of sequential normal modes which begins with the Helmholtz equation under the suitable bottom and surface conditions.

**Fig.1 Schematic diagram of the range-dependent environment and its time evolution.**

The dashed line shows the position of ISW at time $t$.

In the range-dependent situation where the internal solitary wave packet exists between a source and a receiver array (Fig.1), the traditional acoustic field expression is not applicable and must be extended. The region $r_0 < r < r_1$ and $r > r_N$ outside of the wave packet are still range-independent and the region $r_1 < r < r_N$ become range-dependent due to strong perturbation of sound speed profile, generated by ISWs. In order to obtain a range-dependent solution, the region $r_1 < r < r_N$ is divided into $N$ segments, and in all segments the sound-speed fields are approximated as range-independent ones and possesses their own mode function and wavenumber. The solution of acoustic fields can be constructed by considering the standard normal mode solution in each segment and the connectivity condition between the neighbouring segments.

After applying the one-way coupled model, let us calculate the weighting coefficient and propagator matrix recursively. The pressure field at range $r$ yields

$$p(r,z) = \sum_m \Psi_m(z_s, r_N) e^{ik_m^2 (r-r_N)} \sum_n P_{mn}(k_n^0 r)^{-1/2} a_n \Psi_n(z_s) e^{ik_n^0 r}$$

where $\Psi_m$ is the $m$th particular normal mode function, and $k_n^1$ and $k_n^0$ are the wave-numbers at source and receiver, respectively. The $P_{mn}$ is the mode coupling matrix which reflects the acoustic mode coupling when sound propagates across the segments.

Even though the soliton packet may disperse when they propagate in the real ocean, we still assume the shape of ISW is invariable and the ISW travels with a constant speed $u$. Taking into account the orthogonal condition of normal modes and applying the mode filtering, the $m$th modal amplitude at time $t$ is given by

$$A_m(t) = \exp(i(k_m^1 (r-r_N))) \sum_n P_{mn}(k_n^0 r)^{-1/2} a_n \Psi_n(z_s) \exp(i(k_n^0 r_1 + \omega_m t))$$

$$\omega_m = (k_m^1 - k_n^0)u$$

In the above derivation, the amplitude of $m$th mode is a linear summation of discrete signal components and their frequencies is the product of the wavenumber difference between two normal modes and the phase velocity of ISW. When an ISW crosses the region of acoustic propagation, the amplitude of each normal mode constructs a quasi-period oscillation and its period is related to the ISW phase velocity by Eq. (3). By applying the power spectral analysis to the amplitude time series for one normal mode, the phase velocity of ISW is estimated from the multi-frequency structure in the power spectral diagram.

### 2.2. Resonant condition

When acoustic signals propagate through a medium with a range-dependent sound-speed environment, the energy exchange, called the mode coupling, occurs between different normal modes. If mode coupling is absence, it reduces to the adiabatic propagation case,
making the off-diagonal elements of coupling matrix zero. Milder\cite{8} proposed a criterion for adiabatic invariance between two modes, and defined the mode interference length $X_{mn}$ in relation with the wavenumber as $X_{mn} = 2\pi/(k_m-k_n)$.

When the effective spectral peak wavenumber $k_{ISW}$ of internal wave perturbation can approximate the difference of wavenumber between two modes, these two modes are coupled. By considering $k_{ISW} = 2\pi/\lambda_{ISW}$, the resonant condition is obtained as $\lambda_{ISW} \approx 2\pi/(k_m-k_n)$, where $\lambda_{ISW}$ is called the characteristic width or wavelength of ISW. Then by calculating the difference of wavenumber for two coupled modes, the wavelength of ISW is here determined.

3. SIMULATION RESULTS

3.1 Inversion of the ISW phase velocity

Figure 2 shows the simulation scheme of the topography and sound speed field, which comes from the experiment of AEYSFI 2005\cite{9}, conducted off the coast of Qingdao in September 2005. The distance between the acoustic source and receiver is 30km. An ideal ISW packet, composed of 3 waves of amplitude 8m is translated from the source to receiver with a constant velocity of 0.5m/s. Initially the ISW packet is 4.5km away from the source. The simulation is performed by using the FOR3d and Kraken code for the flat and slope topography case.

Figure 3 shows the amplitude time series for the first 5 modes when a 300Hz source is located at depth 30m for the flat bottom case. When the ISW packet is at the range of 4.5-6.5km away from the source, almost all the mode amplitudes are small and the coupling phenomena are weak. If the distance is larger than 6.5km, 1st-4th mode amplitudes are increased and the coupling between the modes becomes strong. Figure 4 shows the result of the spectral analysis for the first 3 modes, and the associated inversion result of ISW is presented in Table 1. Except the inverted result of 3rd mode, others results are quite accurately obtained with error levels smaller than 2.5%. The large error of the 3rd mode suggests that not only two-mode couplings in the first 4 modes, but also multi-couplings among several modes occur. For the 700Hz source, the maximum error of the inverted ISW speed reaches about 12% by considering the 1st and 3rd mode coupling. There are no obvious coupling effects for the 500Hz source.

For the slope topography case, the time series of mode amplitude is much modified in comparison with those for the flat topography case. When the source is located at depth 30m, the amplitude of the first 3 modes become larger, and the coupling phenomena is obvious for any position of ISW (the result are not shown). For the 300Hz/30m source, errors in the inverted phase speed are less than 10.4% by considering the 1st mode. For the source placed at 50m, the inversion errors become as small as 0.63%. For the 500Hz source, the inversion
errors are 14.7% for both the source depths 30m and 50m. For the 700Hz/30m source, many higher modes are coupled with the 1st mode and the maximum inversion error is 5.6%.

Fig. 4 Spectrum analysis of the time series of the 1st (left) and 2nd (right) mode amplitude

Table 1 Inversion results of the ISW phase velocity for the flat topography case. (inverted velocity/error in unit of m/s)

<table>
<thead>
<tr>
<th>ISW packet in 6.5-9.5 km away from source</th>
<th>Mode 1</th>
<th>Mode 2</th>
<th>Mode 3</th>
<th>Mode 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mode 1</td>
<td>—</td>
<td>0.5116/0.0116</td>
<td>0.05236/0.0236</td>
<td>0.5004/0.0004</td>
</tr>
<tr>
<td>Mode 2</td>
<td>0.5116/0.0116</td>
<td>—</td>
<td>0.5311/0.0311</td>
<td>0.4971/-0.0039</td>
</tr>
<tr>
<td>Mode 3</td>
<td>0.5236/0.0236</td>
<td>0.3186/-0.1814</td>
<td>—</td>
<td>0.4674/-0.0326</td>
</tr>
</tbody>
</table>

3.2 Inversion of the ISW wavelength

The flat topography of constant depth 60m is used in the simulation. The sound source is located at depth 30m and 20 km away from the receiver. The mode interference length of the first 12 modes makes a little difference between the 700Hz and 800Hz source.

The mode coefficient ratio \( B_{mn}(r) \), which represents the energy transfer between the mode pair, is useful to judge the mode coupling and may be expressed by

\[
B_{mn}(r) = \frac{\int p(r,z)\psi_n(z)dz / \rho(z)}{\int p(r,z)\psi_m(z)dz / \rho(z)}
\]  

(4)

Fig. 5 Ratio of \( B_{mn} \) in case of with ISW for that without ISW; 700Hz (left) and 800Hz (right).

Figure 5 shows the ratio of \( B_{mn} \) (equals to \( B'_{mn} \)) in case with ISW for that without ISW. Firstly only one soliton with amplitude 5m and wavelength 100m is considered. When the \( B'_{mn} \) increase, the associated mode coupling becomes strong. The more large the \( B'_{mn} \), the smaller the difference between the wavelength of ISW and mode interference length. For the 700Hz source, the \( B'_{mn} \) is larger between the 9th and the first 4 modes, the inverted wavelength of ISW are 102, 107, 118 and 137m, respectively, and the errors for the first two modes are 2% and 7%, respectively. The larger inversion error for the latter two results may be due to the multi-coupling among several modes. For the 800Hz source, the \( B'_{mn} \) is larger between the 11th mode and the first 4 modes, the maximum inversion error is as large as 17%.

If the number of soliton (NS) included in the packet is increased, the inversion error is also
changed. For the 700Hz case, if NS=3, the direct coupling between the 2nd and 9th modes is strongest, and if NS=10 for the 1st and 9th mode pair, the inversion error are 7% and 2%, respectively. If the amplitude of soliton increases to 8m, the direct couplings between the 2nd and 9th mode pair and between the 3rd and 9th mode pair are strongest for the 700Hz and 800Hz cases, respectively, and the associated inversion error are about 7% for the both cases.

4. CONCLUSION AND DISCUSSION

The simulation result suggests that the two proposed methods mentioned above are applicable in the inversion of phase velocity and wavelength of ISW. The error of phase velocity inversion is increased for the slope topography case because the mode coupling is caused not only by the ISW, but also by the variability of topography. The both factors are combined each other, and then the direct-coupling between a mode pair and the multi-coupling among several modes may occur simultaneously. By choosing a suitable frequency and depth of sound source, the inversion error can be reduced. It is very important to choose a suitable resonant frequency in the inversion of ISW wavelength. Under frequencies, deviated from the resonant frequency, the multi-coupling among several modes may become stronger. As a result, the inversion errors become large.

Further theoretical and numerical simulation works are required to understand the resonant interaction between the sound propagation and internal solitary waves, and especially the sensitivity of acoustic mode coupling to internal wave parameters. A field experiment is scheduled to validate the feasibility and applicability of the above inversion methods in the Yellow Sea during the coming summer of 2009.

5. ACKNOWLEDGEMENT

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Using tomography to map turbulence in shallow water regions

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Abstract: Current tomography has been developed and examined in the last two decades to obtain spatial and temporal current profile in the shallow and very shallow water regions. Similar to the deep water tomography, this technique utilizes the reciprocal travel times to accurately measure the peak of an identified acoustic path between the source and the receiver to obtain the current value. However, the strength of the signal reception is usually not considered due to the complexities introduced by the surface and bottom scattering combined with the volume scattering that the signal suffers. However, the signal strength can be used in order to obtain information about the scattering of the acoustic waves due to moving media. In this paper, a theory is presented to formulate the acoustic propagation through a moving media. This theory is used to obtain the acoustic wave scattering. The feasibility of using the theory to get estimates of turbulence is shown with experimental data collected in Kanmon Starit, Japan during 2003 and 2005[Work supported by ONR-321OA program].
FINE-SCALE ACOUSTIC TOMOGRAPHY OF SHALLOW WATER ENVIRONMENTS BY BAYESIAN FILTERING

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Abstract: In the context of the recent Maritime Rapid Environmental Assessment sea trial (MREA/BP07), this paper presents a range-resolving tomography method based on the ensemble Kalman filtering (EnKF) of full-field acoustic measurements on a vertical array. The measurements are assimilated in a Gauss-Markov model of the sound-speed field time variations with known statistics. The reformulation of the inverse problem in an ocean data assimilation framework enables the sequential tracking of time- and space-varying environmental parameters. The trucking scheme is here applied to a realistic simulation of a vertical slice in a shallow water environment. Sea-surface sound-speed measurements are augmented to the measurement vector to constrain the range-dependent structure. Known bottom and subbottom properties are taken into account in the propagation model. When compared to the extended Kalman filter, the EnKF is shown to properly cope with the nonlinearity introduced by the full-field approach.

Keywords: coastal acoustic tomography, nonlinear Kalman filter, empirical orthogonal functions, inversion

1. INTRODUCTION

The continuous monitoring of space- and time-dependent structures in coastal waters is an attractive challenge for the physical oceanography. This complex inversion problem is here reformulated in a data assimilation framework, allowing to recursively estimate the environment states by correcting the state predictions with assimilated acoustic measurements [1]. The ensemble Kalman filter, or EnKF [2], have already proved his efficiency for acoustic data assimilation in physical-based ocean model in the case of range-resolving currents inversion from multiple sources and receivers [3]. This paper presents a range-resolving tomography method based on EnKF applied in a realistic vertical section of a shallow water environment (113-m depth). In a previous work [4] we showed the feasibility of an extended Kalman filtering (EKF) scheme applied to full-field acoustic mea-
measurements with a dense vertical receiver array (VRA) spanning the full water column (48 hydrophones, with 2-m inter-element spacing). The present work makes use of a VRA with fewer elements (16 hydrophones) partially covering the water column, as more often used in practice. Furthermore, the approximations made in the standard EKF required the use of acoustic transmissions closely spaced in time to reduce the amplitude of the state variations between the filter iterations. The use of sampling strategies enables the increase of the time interval between iterations and reduces estimation errors by correctly handling the high nonlinearity between the measurements (the multi-frequency acoustic pressure field) and the environmental parameters, i.e., the sound-speed field (SSF). In support of the Maritime Rapid Environmental Assessment MREA/BP07 sea trial southeast of Elba, Italy (Fig. 1), prediction results from Navy Coastal Ocean Model (NCOM) provide realistic scenarios to test the SSF-tracking algorithm. The model configuration consists of multiple nests with different resolutions going from regional to local domains, i.e., from 4 km to 0.6 km horizontal resolution. The complete prediction period extends from April 19th to May 1st, 2007, with a new SSF predicted every hour. This prediction database serves also to compute an empirical orthogonal functions (EOFs) set which enables the reduction of the parameterization dimension of a vertical slice of the SSF used in the filtering scheme. More details on the NCOM setup can be found in [5].

The remainder of the paper is organized as follows. The state-space model is introduced in Sec. 2 and details the range-resolving parameterization. Section 3 then reminds the basic principles of the Kalman filtering and summarizes how it is applied for this specific inverse problem. Simulation results of EKF-based and EnKF-based tracking are compared in Sec. 4. We conclude the paper in Sec. 5.

2. STATE-SPACE MODEL

The complex acoustic pressure field received from a distant acoustic source at frequency \( \omega \) can be expressed as a general nonlinear function of the SSF of the environment \( c(r, z) \) and the boundary conditions (surface, bottom and subbottom properties)

\[
P = P[r, z, \omega, c(r, z), BC]
\]  

(1)

where BC states for the boundary conditions.

2.1. Sound-speed field parameterization

A sound-speed profile (SSP) can be parameterized in a low-dimensional scheme using empirical orthogonal functions, or EOFs [6]. The EOF parameterization is an orthogonal decomposition obtained from the sound-speed data covariance matrix estimated from historical measurements database. Each SSP \( c(z) \) is approximated by a finite number of EOF coefficients \( \{\alpha_i\} \), so that

\[
c(z) \approx \bar{c}(z) + \sum_{i=1}^{I} \alpha_i \delta c_i(z).
\]  

(2)

where \( \bar{c}(z) \) is the mean profile of the database and \( \delta c_i(z) \) are the eigenfunctions of the covariance matrix.

In this paper oceanic predictions on the transect A-B from the NCOM model constitute the database for the computation of the EOFs. The mean SSP and the first three EOFs computed from the NCOM predictions are shown in Fig. 2.

As proposed in [7], the vertical slice is decomposed in a finite number of non-overlapping rectangular regions. Dividing the range between the source and the receiver array into \( J \)
regions and using $I$-EOFs, the SSF is expressed as
\[ c(r, z) \approx \tilde{c}(z) + \sum_{j=1}^{J} \sum_{i=1}^{I} \alpha_{i,j} \delta c_i(z) u_j(r), \]
where $u_j(r)$ is a gate function equal to 1 if $r$ belongs to the $j$-th region and to 0 in the other regions. The coefficients $\alpha_{i,j}$ constitute the SSF parameters. The resulting vector dimension is therefore the product between the number of EOFs considered and the number of regions of the vertical slice. The Fig. 3 shows the range discretization of the SSF used in this work.

### 2.2. Gauss-Markov state-space model

We assume that in a given region the associated EOF coefficients change slowly. With this in mind we choose to represent the coefficient evolution as a random walk. This assumption on the EOF coefficients can easily be placed in state-space form where the second-order statistics include the coefficient variations. The measurement function relies
Figure 2: (a) Mean sound-speed profile (solid line) of the oceanic predictions database (gray) on the entire sea trial period (from the 19th of April to the 1st of May, 2007). (b) First three EOFs computed from this database, weighted by their respective eigenvalue (blue line: EOF 1, green line: EOF 2, red line: EOF 3).

Figure 3: Range-dependent environmental model used for the synthetic cases. The source coordinates are (0 km, 60 m) and the 16 hydrophones coordinates are (15 km, 30–90m). The sound-speed field is discretized in 7 regions (R1,...,R7) and color-coded with the corresponding SSP overlaid. The bottom is modeled as described in [8].

on the SSF relationships to the complex acoustic pressure field measured on a VRA as embedded in Eq. 1. Sound-speed values at the sea surface along the transect are augmented to the measurement vector in order to constrain the range-dependent structure. The state-space model can be summarized as
Figure 4: Simulation and tracking diagram of the range-resolving tomography scheme. At each step, the SSF estimates are corrected with acoustic and surface measurements in an EnKF algorithm.

\[
\alpha(\tau_m) = \alpha(\tau_{m-1}) + w(\tau_{m-1}) \\
y(\tau_m) = h[\alpha(\tau_m)] + v(\tau_m)
\]

where \(\tau_m\) is the m-th discrete time index, the state vector \(\alpha(\tau_m)\) contains the EOF coefficients \(\alpha_{i,j}\) defined in Eq. 2, the measurement vector \(y(\tau_m)\) contains the complex acoustic pressure on the VRA at the frequencies \(\Omega = [\omega_1, \ldots, \omega_J]\) augmented with the \(J\) surface sound-speed values \(s(\tau_m) = [s_1(\tau_m), \ldots, s_J(\tau_m)]^T\) along the transect.

\[
y(\tau_m) = \begin{bmatrix}
P(\Omega; \tau_m) \\
\vdots \\
s(\tau_m)
\end{bmatrix}
\]

and \(w(\tau_m)\) and \(v(\tau_m)\) are zero-mean Gaussian random vectors of covariance \(R_{ww}\) and \(R_{vv}\), respectively.

3. KALMAN FILTER AS AN INVERSION PROCESSOR

The problem of tracking the SSF of a vertical slice of a shallow water environment can be expressed as follows: GIVEN a set of noisy complex acoustic field measurements on a vertical array FIND the best (minimum error variance) estimate of the SSF of the environment.

The state-space formulation enables a straightforward implementation of a Kalman filter. The nonlinearity of the measurement function requires nonlinear filtering methods. The Kalman filter operates as a predictor-corrector algorithm (Fig. 4). More details about the Kalman filtering can be found in [9].

The common Kalman filter applied to nonlinear systems is the EKF. In an EKF, the covariance matrices are propagated through the nonlinear model with the Jacobian of the transition and measurement functions. Such a first-order approximation can result in large estimation errors and sometimes causes the filter to diverge. The more recent EnKF represents the underlying distributions with large stochastic ensemble of models,
as in the Monte-Carlo methods. Each member is propagated through the transition and measurement functions and the statistics of the estimates are propagated without the use of any linearization step. More details about the EnKF can be found in [2].

4. RESULTS

Prediction results from NCOM provide realistic scenarios to test the tracking of range-dependent features of the SSP of the water column. The complete set of predictions covers the period from the 19th of April to the 1st of May, 2007. The bathymetry of the transect A-B shows slight variations between 111 m and 115 m, and the thickness of the soft sediment layer varies between 5 m and 9 m [8]. In this first study all the simulations were made with a constant depth of 113 m. The VRA is configured with 16 elements equispaced 4 m that spanned the 30-90-m water depths.

Three frequencies are used for the simulated acoustic transmission: 250 Hz, 400 Hz and 630 Hz. The known geoaoustic properties are inputs to the acoustic propagation model and are kept constant temporally. These parameters were previously obtained by geoaoustic inversion during the Yellow Shark 94 experiment [10]. These results were confirmed during the MREA/BP07 sea trial by inversion of higher-frequency acoustic data measured on a sparse array [11].

The algorithm is first initialized by a range-independent acoustic inversion with a global search algorithm. The initial conditions are therefore range-independent. As shown in Fig. 5 the filter is able to track both the spatial and temporal variations of the SSF. The EnKF estimation errors are clearly smaller than those from the EKF, with a resulting mean RMSE on the SSF equal to 0.30 m/s for the EKF and 0.18 m/s for the EnKF.

5. CONCLUSION

Simulation results showed that it is possible to track the time-evolving sound-speed field of a shallow water environment from full-field acoustic transmission measurements at multiple frequencies and additional range-related observations such as the sea-surface sound speed. The EnKF scheme is able to track range-dependent EOF coefficients by using a range-independent environment as an initial guess, as it can be estimated from a global search. The EnKF outperforms the extended Kalman filter and constitutes a more robust scheme. Furthermore the scheme enables to account for range-dependent bottom properties and can be extended to track their spatial variations when using moving sources or receivers [11]. Further improvements can be obtained by using non-Gaussian statistics, as in particle filters, or by incorporating a physical-based transition model.

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Figure 5: EOF coefficients tracking on a 2-day sequence, with a new measurement each hour (true values in blue line, EKF in green-dashed line and EnKF in red-dotted line. Each column corresponds to a region of the transect, the source being on the left side (R1) and the VRA on the right side (R7) of the figure. Each row corresponds to an EOF coefficient.

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2008.
MEASUREMENTS OF ACOUSTIC PROPAGATION IN RECIPROCAL TRANSMISSION EXPERIMENT FOR SHALLOW WATER

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\textbf{Abstract:} Development of coastal acoustic tomography has been carried out as a long term measurement technique to get shallow water information. Since sound is propagated with reflection at sea bed and sea surface, it is important to estimate characteristics of acoustic propagation in shallow water. In this paper, we investigated acoustic characteristics and estimated water temperature and current speed using data of a reciprocal acoustic transmission experiment in Uchiura Bay, Japan. Direct and surface reflected rays were used to estimate travel time and amplitude variation. We estimated water temperature with high accuracy and observed that an amplitude fluctuation happened with a rapid change of travel time. We got basic acoustic characteristics of reciprocal transmission signals that are useful for temperature and current measurements in shallow water.

\textbf{Keywords:} ocean acoustic tomography, coastal acoustic tomography, shallow water
1. INTRODUCTION

Ocean acoustic tomography (OAT) and coastal acoustic tomography (CAT) have been considered as a powerful approach for observation of mesoscale ocean fluctuations. The objective of the OAT and CAT are to estimate ocean physical parameters (temperature distribution, current variability, sediment structure) in wide areas using acoustic data analysis. Measurement of ocean currents by reciprocal sound transmission was successively carried out on a scale of 10-1000 km. However, the small-order and short-time currents are not precisely measured. The reason is that the travel time is measured by the daily average of reciprocal amplitudes because ocean fluctuations such as internal waves can split the basic ray path into micromultipaths, which then interfere at the receivers, sometimes destructively. In a series of publications\cite{3}-\cite{7}, the authors and others have estimated travel times and the stability of acoustic reciprocal transmission in the 1999 and 2000 OAT experiments.

In bays and coastal waters, there is also a great demand for information on ocean structure in bays and coastal waters. Since sound is propagated with reflection at sea bed and sea surface in coastal water, it is important to estimate characteristics of acoustic propagation in shallow water. In this paper, we present measurement results of water temperature and current velocity using both the amplitude and the phase information to measure the reciprocal travel time.

2. EXPERIMENTS

We use the data of a reciprocal transmission experiment that was performed at Uchiura Bay in Numazu city, Japan from 11 to 18 on November. The transducers at seashore and offshore side were located at about 5m and 30m depth, respectively, shown in Fig. 1. A distance between the transducers was about 350m. In this experiment, M-sequence signal was used as transmitted signals. M-sequence code is a pseudo random code generated by shift-registers and exclusive-or logic. Degree-7 code with 127 digits consists of 7 shift registers. The sequence was repeated four times during each transmission. The transmitted signal is a periodic repetition of the 7th-order M-sequence signal with the following characteristics:

a) carrier frequency $f_0 = 31.25$ kHz,

b) digit length = 2 cycles of 31.25 k Hz = 0.064 ms,

c) sequence length $L = 127$ digits = 8.128 ms, and

d) transmission length = 4 sequence periods = 32.512 ms.

The 31.25 kHz M-sequence acoustic signals were transmitted every 20 s. Clocks at both sides were synchronized using GPS system. At H[h]: M[min]:00[s], 20[s] and 40[s], the transducer at offshore side transmitted the signal and at H[h]:M[min]:10[s], 30[s], 50[s], the transducer at seashore side transmitted the signal. The received signal was sampled at 250 kHz (eight times center frequency).
Figure 2(a) shows the received signal at November 17, 02:00:00. Figure 2(b) shows the cross-correlated signal with a M-sequence replica. A horizontal axis corresponds to travel time, and a vertical axis corresponds to the normalized amplitude. The predicted improvement of the signal-to-noise ratio for the Gaussian noise was 21 dB. The correlated results are equivalent to the results when 4 sequences of a two-cycle pulse of 31.25 kHz are transmitted. Signals of 1 and 1’ correspond with propagated signals of first sequence of signals. 2, 2’ are second sequence signals, 3, 3’ are third sequence signals and 4 is forth sequence signal, respectively. Signals of 1 consist of direct and surface reflected rays and signals of 1’ consist of surface and bottom reflected rays. The surface and bottom reflected ray of 1’ arrived after the direct ray and surface reflected ray of 2 of second sequence arrived. Figure 3 shows a bird’s eye view of correlated signals. We use the data after the correlation processing for the analysis in this paper. Especially second sequence of each transmission was used to eliminate the effect of sequence truncation. We use the second sequence signal but the travel times were corrected using sequence length, 8.128 ms which become the real travel time between transducers.

Figure 4 shows a two-dimensional plot of the amplitude of reciprocal signals of second sequence from November 11 to November 18. Amplitudes were normalized using the maximum amplitude. The signal to arrive first shown with a black arrow is the direct wave which propagates directly without reflections at surface and bottom. The second arrival signal shown with a white arrow is the surface reflected ray. In this paper, we use direct and surface reflected rays shown with black and white arrows to estimate the travel time. The travel times of direct and surface reflected rays increased periodically during one day. The biggest change of travel time happened from 15:00 to 0:00 midnight. It is because there were periodical decreases of the water temperature on a propagation path of the measurement sea area at that time that the travel times of rays increased.

Figure 5 shows estimated travel times of direct (a) and surface reflected (b) rays as a function of time. The surface reflected ray has a smaller stability than direct ray, but travel time estimation was stable.
Fig. 2: (a) received signal and (b) correlated signal of M-sequence signal.

Fig. 3: 3D plot of Received signals.
Figure 4: 2D plot of received signals of direct and surface reflected rays.

Figure 6 shows estimated travel time difference using amplitude only (a) and amplitude-phase information (b). The travel time differences, estimated using combined amplitude and phase information in complex plane, have reasonable magnitudes, and then the low-frequency trend is presumably caused by fluctuations in this ocean region. The current velocity was a small about order of cm/s, but we were able to measure current velocity from time difference between reciprocal propagation.

Figure 7 shows the comparison between acoustical estimated temperature of direct ray and thermometer temperature (15 m deep). Figure 8 shows the comparison between acoustical estimated temperature of surface reflected ray and thermometer temperature (0 m deep). The accuracy of the water temperature was estimated to be around 0.02 degrees Celsius by time resolution.
Fig. 5: Estimated travel times of direct (a) and surface reflected (b) rays.

Fig. 6: Estimated travel time difference using amplitude (a) and phase (b).
Fig. 7: Comparison between estimated temperatures. (a) acoustic method and (b) thermometer (15m).

Fig. 8: Comparison between estimated temperatures. (a) acoustic method and (b) thermometer (0m).
3. CONCLUSION

In this paper, we estimated water temperature and current speed using data of a reciprocal acoustic transmission experiment in Uchiura Bay, Japan. Direct and surface reflected rays were used to estimate travel time and amplitude variation. We estimated water temperature and current speed with high accuracy. The combination of the amplitude and phase information is effective for coastal water measurements.

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SIGNAL ANALYSIS APPROACH FOR PASSIVE TOMOGRAPHY: APPLICATIONS FOR DISPERSE CHANNELS AND MOVING CONFIGURATION

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Abstract: Underwater channel is an example of a natural environment potentially characterized by signals generated by various sources: underwater mammals, human activity noise, etc. In order to take advantage of these sources, the concept of passive acoustic tomography has been introduced. According to this concept, the environment parameters could be extracted from the analysis of the received signals. While the signal’s parameters are intimately related to physical parameters of the environment, their accurate extraction is crucial. That is, this task is complex while we work in completely passive context and when we deal with a large diversity of underwater signals. Generally, signals issued from underwater environment have complex time-frequency structures: non-linear time-frequency and multi-components. Two typical non-linearities are generated by the dispersive systems and the relative motion between transmitter and receiver. Despite the origin of these phenomena, the signal approach proposed in this paper will provide a unique framework for parameters extraction. This approach is based on the time-frequency-phase coherence of any natural non-linear time-frequency component. Taking advantage of this coherence, the non-linear structures can be efficiently extracted and used for physical parameters estimation.

Keywords: Time-frequency analysis, passive tomography, dispersive channels, motion effect
1. INTRODUCTION

One well known technique for underwater environment characterization consists in transmitting acoustic waves and analyzing, at the receiver, the distortions induced by the environment. This technique, called oceanic acoustic tomography and introduced in 80s, allows monitoring ocean properties at variable spatial and time scales [1]. Once the feasibility of active acoustic tomography proved [2], starting in earlier 2000, new constraints are imposed by operational considerations:

- **Fast deployment** of tomography system, knowing that the active tomography concepts use often complex sensors networks with high costs;
- **Acoustic discretion** knowing that the transmitted signals used for channel investigation could interfere with other signals existing in the environments.

These constraints were the main arguments behind the definition of passive acoustic tomography concept where the acoustical transmitted signals are replaced by signals transmitted by **natural opportunity sources** existing in the environment [2].

The difficulties in terms of signal processing are related to the lack of a priori information about signal's type as well as the complexity of underwater environment (in terms of noise, propagation effects, etc). In this context, the signal analysis methods are aimed to extract the parameters of received signals and to transform them in physical parameters related to the channel properties.

In this paper, two applications contexts are considered. The first one consists of **dispersive environments** which concern the underwater systems operating at low frequencies (mainly, below 200 Hz). A dispersive channel modifies a transmitted signal in a complex way, bringing it deeply non-stationary. More precisely, a dispersive channel produces different delays of spectral components according to their frequencies: high frequencies are generally less delayed than the low ones [3]. The transformation of signal propagating in dispersive channel can be characterized by a non-linear change of the phase function of the signal: \( \xi(t/t_r) \) where \( t_r \) is the reference time. For example, in shallow water environment, the phase changes according to \( \xi(t/t_r) = \sqrt{(t/t_r)^2 - (\alpha/t)^2} \) \( (t > \alpha) \) [3]. This change is specific to each propagation path. In conclusion, a signal issued from a dispersive channel is composed by many versions of transmitted signal, each one having distinct time-frequency shape according to its propagation mode [1]. In addition, these versions are very close in time-frequency plane which bring difficult their separation.

The second application context concerns the characterization of the **motion effect** existing in the received signal. This characterization can either improve the performances of the existing systems either enable the use of some concepts. Namely, the motion effect impinging on a signal arising in a communication system could decrease the receiver performances [4]. If the motion is correctly estimated, its effect can be compensated increasing also the performances of the communication system. On the other hand, the motion constitutes a source of additional information since the successive positions of the source-receiver configuration allow the characterization of the underwater environment from different angles of “view” [4]. This property could be exploited in applications like sonar imagery or acoustic tomography. In all cases, the success of the operation is conditioned by the motion effect estimation.

In this paper we propose a common methodology to deal with the signal’s structures specific to dispersive and moving configuration. Since no a priori on signals are authorized, the methodology exploits the coherence of fundamental parameters of any type of signals:
amplitude, frequency and phase. Specifically, the time-frequency structures of received signal will be separated by analyzing their continuity in terms of instantaneous amplitude, phase, frequency. Since these structures are closed (because of the multipath fading effect) the continuity criteria will be aimed to provide high-resolution capabilities. Furthermore, once the time-frequency structures estimated, they will be filtered by using the generalized time-frequency filters structures [5]. Finally, individually analysis of each structures and comparison between successive arrivals can provide information about phase changes due to the dispersivity or to the motion.

This methodology is analyzed in the context of underwater dispersive channel context and moving configurations. Realistic configurations will be used in order to illustrate the outperforming of the proposed approach. Although the outperforming is illustrated in the underwater configuration, the proposed approach is general since it exploits fundamental items related to any type of signals (amplitude, phase and frequency).

The paper is structures as follows. In the section 2 we define the concept of time-frequency-phase continuity and propose the signal processing methodology. The characterization of dispersive phenomena is illustrated in section 3. The motion effect analysis is illustrated in the section 4. We conclude in section 5.

2. SIGNAL PROCESSING METHODOLOGY

The methodology proposed in this paper has the benefit to be general in the sense that it could be used for a large number to signal’s types. For this purpose, the general model of analyzed signal is:

\[ x(t) = \sum_{i=1}^{N} A_i e^{i\phi_i(t)} + n(t) \]  

where: \( x(t) \) is the received signal composed by \( N \) components, \( A_i \) is the amplitude of \( i^{th} \) component, \( \phi_i(t) \) is the time-varying phase of \( i^{th} \) component and \( n \) is the noise. The general framework proposed in this section is aimed to provide an estimation of the phase law of each component.

Our strategy consists in performing an exhaustive search over a local but general model of the instantaneous phase. Hence, we firstly limit the observation range over a time window. This is illustrated in the figure 1 where the characterization of an arbitrary time-frequency component (solid line) is considered.

![Fig. 1: Illustration of the exhaustive search procedure](image)

As indicated by this figure, on each window, a set of time-frequency component of order 3 is constructed for different nodes over a grid parameter. For each component, the Log-
likelihood is estimated over the window and only a number of \( i \) components giving highest likelihood is selected. One such component over the window \( k \) are denoted \( M_i^k \).

Next step consists in regrouping the detected components in each window. As each component \( M_i^k \) represents a local model of order three, the goal of the regrouping step is to find the chain \( \{ M_i^k, \ldots, M_i^{k+1} \} \) that best represents the original component \( i \) with respect of maximum likelihood criterion. Hence, the strategy consists to associate two components if they verify some time-frequency-phase coherence criteria. Four different criteria are defined:

- **C\(0\) time-frequency continuity.** This criterion is given by \( C^0(M_i^k, M_i^{k+1}) = |f_i(M_i^k) - f_j(M_{i+1}^k)| \) where \( f_i(M_i^k) \) is the initial frequency of component \( M_i^k \) and \( f_j(M_i^{k+1}) \) is the final frequency of component \( M_i^k \). This criterion points on the frequency discontinuities between the component \( M_i^k \) and \( M_i^{k+1} \) (figure 2.a). More precisely, if \( C^0(M_i^k, M_i^{k+1}) \) has high value, the probability that \( M_i^k \) and \( M_i^{k+1} \) belong to the same time-frequency component is low;

- **C\(1\) time-frequency continuity.** The criteria is given by \( C^1(M_i^k, M_i^{k+1}) = |\partial f_i(M_i^k) - \partial f_j(M_i^{k+1})| \) where \( \partial f_i(M_i^k) \) and \( \partial f_j(M_i^{k+1}) \) respectively denote the initial and final instantaneous frequency rates of the components \( M_i^k \) and \( M_i^{k+1} \). As indicated by figure 2.b this criterion rules the connection of \( M_i^k \) and \( M_i^{k+1} \) in the following way if the rates of IFLs are almost the same the time-frequency content variation is smooth being subject to a single component (figure 2.b). In this case \( M_i^k \) and \( M_i^{k+1} \) are merged. If the rates are too different the both components do not belong to the same structure;

![Fig. 2: C0 and C1 time-frequency continuity](image)

- **Amplitude continuity.** The criteria is given by \( d A(M_i^k, M_i^{k+1}) = |A(M_i^k) - A(M_i^{k+1})| \) where \( A(M_i^k) \) denote amplitude of \( M_i^k \) estimated. This criterion materializes the general observation that the energy of a time-frequency component varies slowly;

- **Instantaneous phase continuity.** The criteria is given by \( d\phi(M_i^k, M_i^{k+1}) = |\cos(\phi_i(M_i^k) - \phi_j(M_i^{k+1})) + i\sin(\phi_i(M_i^k) - \phi_j(M_i^{k+1}))| \) where \( \phi_i(M_i^k) \) and \( \phi_j(M_i^{k+1}) \) respectively denote initial and final instantaneous phase \( M_i^k \). This criterion merges two candidate components if the phase is continuous.

With these criteria, we define the penalty function of two components belonging to two consecutive windows by

\[
p(M_i^k, M_i^{k+1}) = \alpha C^0(M_i^k, M_i^{k+1}) + \beta d A(M_i^k, M_i^{k+1}) + \gamma C^1(M_i^k, M_i^{k+1}) + \delta d\phi(M_i^k, M_i^{k+1})
\]

where the coefficients \( \alpha, \beta, \gamma, \delta \) allow to define the weight of each criteria. Based on these criteria, the regrouping strategy consists to search over all the possible chains \( \{ M_i^k, \ldots, M_i^k \} \) the one that minimizes the penalty function.
\[ p_{\text{opti}} = \arg \min \sum_k p(M_i^k, M_i^{k+1}) \] (3)

The minimization of this penalty function leads to each individual time-frequency structure. This procedure is illustrated in the figure 3.

After the optimisation procedure (3) we merge the local 3\textsuperscript{rd} order components in order to get the time-frequency trajectory of the most energetic component. Furthermore, this trajectory is used to design the time-frequency filter (figure 3, right part) and to extract the samples corresponding to this component. While generally the time-frequency trajectory is non-linear, a time-frequency filter with non-linear time-frequency shape has to be designed. The solution, proposed in [5], consists of time-warping the component in order to get a stationary signal. Thus, a band-pass filter is used to isolate the stationarized component from the mixture. An inverse warping will bring the extracted component in time domain.

This methodology, based on local 3\textsuperscript{rd} order component matching, structures fusion and time-frequency filtering, is iterated until all signal’s component are extracted. Its using in the context of dispersive channels and moving configuration is addressed in the next sections.

3. ANALYSING SIGNALS FROM DISPERSIVE ENVIRONMENTS

In this section we illustrate how the methodology previously defined performs in the case of signals issued from dispersive environments. Such a signal is illustrated in the figure 4 where we consider an isovelocity channel with \( h=16 \) m, a transmitter and receiver located at 4 m depth. The range transmitter-receiver is 2000 m. The spectrogram of the first four modes is indicated in the figure 4.

As shown by this figure, identifying the four components is not a simple task. The linear representation (figure 4.a.) shows only two modes the other two being invisible because of their much smaller energies with respect of strongest ones. One can use a logarithmic representation (figure 4.b) the price to pay being the poorer resolution. For these reasons the
extraction of each individual mode could be a complex problem for a classical time-frequency technique.

Using the proposed methodology, at the first iteration (figure 5), the fusion of the local most matched cubic FMs provides a time-frequency shape corresponding to the third order modal arrival (figure 5.a). This shape allows us defining a time-frequency filter (figure 5.a) which extracts the corresponding signal (figure 5.b). The spectrogram of residual signal is illustrated in the figure 5.c. We remark that the time-frequency-phase criteria provide an accurate estimation of time-frequency structures of the signal. The time-frequency filters designed from this estimation (figure 5.a, d, g) allow accurately extracting the modal arrivals (figure 5.b, e, h et i). After second iteration, we remark also that the filtering-based extraction highlights less energetic components. It is a consequence of filtering out the first two most energetic components. This property of the proposed methodology proves its interest for signals composed by several arrivals with large differences of magnitude.

Fig. 5: Extraction of the third order mode from the signal defined in the figure 4

The time-frequency-phase criteria provide the time-frequency shape of the most energetic component of the signal. Furthermore, this shape produces the time-frequency filter which physically extracts the component. This is actually the novelty of the proposed methodology with respect of Matching Pursuit-based approaches which performs the component extraction via a subtraction. While this operation requires the estimation of component's magnitude (which is not always an easy task) the method proposed in our paper performs component's extraction via a time-frequency filtering procedure. In this way, the extraction is independent of component's amplitude which is of great benefit in the study of dispersive phenomena.
4. ANALYSING SIGNAL ISSUED FROM MOVING CONFIGURATION

In this section we focus on the motion effect analysis using the methodology proposed in section 3. For this purpose, we consider the scenario defined in figure 6: the transmitter, located at a depth of 5 m, moves as indicated in figure 6, with 5 m/s. Two sensors, located at 40 m and at a relative range of 100 m, are used as receiving structure. The transmitter emits a pure 12 kHz sinusoid. The spectrogram of the signal received by the sensor 2 is plotted in the bottom of figure 6. This figure shows intuitively the non-linear time-frequency Doppler modulation due to the source moving. Nevertheless, the exact time-frequency modulation is difficultly estimated by exploiting spectrogram while the multi-path propagation produced destructive interferences which create gaps in the time-frequency distribution.
Applying the time-frequency-phase coherence we can extract exactly the time-frequency modulation as indicated in the figure. The time-frequency trajectories extracted from the signals received by the two sensors correspond well to the moving scenario.

This result is explained by the robustness of the time-frequency-phase concept in the presence of coherent perturbation as the multi-path interferences.

5. CONCLUSION

In this paper we presented a general methodology for multi-component signal analysis in the passive tomography context. The signal’s model is general while it is composed by multiple components having arbitrary non-linear time-frequency behaviours. In addition, no hypothesis about the noise is considered. In these conditions, the proposed methodology attempts to extract all components which could be very helpful in the context of passive tomography. Each individual component contains information about the propagation path and/or about the trajectory of the source/receiver. The main idea is to exploit the time-frequency-phase coherence of each component. More precisely, while the time-frequency
content of each component is typical to the physical phenomenon of interest, the instantaneous phase and/or its derivatives of the component are continuous. This general property of (almost) all physical phenomena is the central item of the proposed methodology. 

The first step of the methodology is to project the short time signals (obtained by windowing the full signal) on a dictionary of 3rd order components (or cubic frequency modulation). The best locally matched components of each window are retained. In the second step, the 3rd order components of close window are merged according to the minimisation of continuity criteria. The combination of the components minimizing the penalty function constitutes the time-frequency trajectory of one of the signal’s components. This trajectory serves to filter out this component. The procedure is iterated until all components are estimated.

Thanks to the universality of this property, this methodology could be successfully applied in different area of applications arising from passive tomography domain. Two cases have been considered in this paper: characterization of underwater dispersive phenomena and monitoring of motion effect of an underwater sources.

The further works will be concentrated to more complex scenarios in order to improve the operational aspects related to this methodology. One of the work directions will be focused on the automatic selection of the continuity parameters.

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Passive geoacoustic inversion using ship's noise and a single moored hydrophone

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Abstract: Passive geoacoustic inversion using ship's noise and a single moored hydrophone. Applications from very shallow waters to shallow waters. C. Gervaise+, B. Kinda+, S. Vallez+, Y. Stephan*, Y. Simard^ +: EA3876, ENSIETA, 2 Rue Francois Verny, 29200 Brest, Université Européenne de Bretagne, *: SHOM, 13 Rue Chatellier, 29200 Brest, France ^: Institut des sciences de la Mer, Université du Quêbec à Rimouski, 310 Allée des Ursulines, Rimouski Québec, G5L3A1 & Institut Maurice Lamontagne, Ministère des pêches et océans, 850 Route de la mer, Mont-Joli, Québec, G5H-3z4 Corresponding Author: cedric.gervaise@ensieta.fr

Ships radiate acoustic noise from 10 Hertz to more than 1 KiloHerz. Range of ship's noise extends from 1 kilometer to more than 10 kilometers in shallow waters. These two properties make ship noise to be a valuable applicant for passive geoacoustic inversion. Ship noise is divided in a continuous spectral wide band component and a discrete spectral lines component. Taken into account ship’s movement and normal mode propagation, measurements from a single moored hydrophone are modeled and relations between their time-frequency features and channel properties allow designing inversion schemes. For the wideband component, the 'Bath Tube' interference pattern is used to estimate features such as differences (versus frequency) between horizontal wave numbers of propagative modes or modes cut off frequencies. These measured features are fit with predicted theoretical ones to estimate geoacoustic parameters of the channel. Successful examples are given in ultra shallow water (depth: 15 meters, fishing boats off Cataluña coast, Spain, 2006) and in shallow water (depth: 135 meters, 50 meters long boat in Bay of Biscay, France, 2005). For the spectral lines, combined effects of modal propagation and ship movement affect modal components with different Doppler modulations. Signal processing tools are designed to exploit Doppler effect to estimate the horizontal wave numbers of propagative modes as a function of frequencies. This measured feature is fit with predicted theoretical ones to estimate geoacoustic parameters of the channel. Promising results are given in shallow water (depth: 300 meters, several cargo in Laurentian Estuary, Canada, 2006)
Operational feasibility of an acoustic passive monitoring of the Ushant front

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Abstract: The western part of Brittany, the so called Iroise Sea, around Ushant Island, has been recently declared as a marine protected area (MPA). Some initiatives are emerging to develop non intrusive monitoring tools to assess the oceanographic environment, which is highly dynamic in this area and qualify the ecosystems. Among possible candidates, passive ocean tomography, (POT) which consists in inverting opportunity acoustic signals to characterize the environment, has to be considered with interest in this area for two main reasons: - a high density merchant ship traffic can be observed in the area (about 150 ship a day). This traffic can provide an almost permanent source of opportunity to be inverted for environmental characterization. In addition, the traffic is permanently monitored by a maritime Rescue Coordination Centres (MRCC) which may provided useful information on the traffic (type of ship, speed, positions...) necessary for real-time data inversion - POT is a remote sensing method, so that the necessary moorings can be deployed outside the traffic area, minimizing constraints due to navigation safety and risks of losses and damages to the instrumentation. This paper examines the possibility of building an operational tomographic network to monitor the environment in the Ushant area. Both the scientific and technological points of view are discussed. On the one hand, we present results of POT data assimilation in the Ushant environment. A particular attention is paid the Ushant front monitoring. For this purpose, typical front variations are predicted by a regional HYbrid Coordinate Ocean Model (HYCOM) run at SHOM. The simulated environment is used to generate acoustic pressure field in the 200-800 kHz frequency band. These data are assimilated in a Kalman-based scheme enabling the tracking of the principal properties of the thermal front, using a basic feature model. Simulation results for a number of realistic environmental scenarios demonstrate the feasibility of the method. Comparison between the feature model and true oceanic modeling outputs are discussed. On the other hand, we examine the technological feasibility of designing a tomographic network in this area. A systemic approach will be presented and each node of the network will be discussed in terms of data access, fluxes and achievable performances of the system. The outlines of an at-sea demonstration will be given.
Accuracy of current tomography in shallow water using reflected ray paths

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Abstract: Reciprocal high frequency broadband (1-25 kHz) acoustic transmissions and oceanographic experiments were simultaneously conducted in the shallow water area of Delaware Bay (average depth 15 m) in order to study current tomography in shallow waters. Unlike deep water tomography, where refracted ray paths can provide an accurate account of travel time, the shallow water structure of the arriving signals is difficult to resolve due to multi-path interference and multiple reflections from the sea surface and the bottom. In this study, the accuracy of travel time measurements of direct and surface bounce rays was examined for multiple center frequencies and bandwidths in variable environmental conditions. Because acoustic ray signatures are strongly affected by the signal’s center frequency and bandwidth, currents can be accurately estimated through the optimization of frequencies and bandwidths for different ray paths and sea state conditions. Comparisons with ADCP measurements show that in general, the current can be accurately estimated using multiple bounce rays for relatively calm sea states (wind speed less than ~ 5 m/s) and one bounce rays for both calm and rough sea states, with accuracy decreasing as sea surface roughness increases. Results show that more accurate estimations can be achieved by using higher center frequencies for earlier arrival groups, and lower center frequencies for later arrival groups [Work supported by ONR-321OA].
Abstract: This paper presents the application of a probabilistic approach for variational inversion in acoustic tomography. The aim is to determine the time-evolving, range-averaged, vertical profile of speed of sound $c(z, \tau)$ in a shallow water environment from the acoustic pressure fields generated by a monochromatic sound source and measured on a sparse vertical hydrophone array. A variational approach that minimizes a cost function which measures the distance between observations and their modelled counterparts is used. As the tomographic inversion is an ill-posed problem a regularization term in the form of a quadratic restoring term to a background is added. To avoid inverting for the variance-covariance matrix associated with the above weighted quadratic background, it is proposed to model the sound speed vector using probabilistic principal component analysis (PPCA). The probabilistic PCA introduces an optimum reduced number of non correlated latent variables $\eta$ which determine a new control vector and introduce a new regularization term, expressed as $\eta^T \eta$. PPCA represents a rigorous formalism for the use of a priori information and allows for an efficient implementation of the variational inverse method. In the present work the probabilistic PCA is applied to an acoustic tomography scenario in the South Elba environment.

Keywords: probabilistic principal component analysis, empirical orthogonal function, tomographic inversion, variational method, adjoint method, regularization.
1. INTRODUCTION

The application of a probabilistic approach for variational inversion in shallow-water acoustic tomography is presented. The classical approach consists of introducing a cost function $J$ that represents the distance between actual measurements and their prediction computed from the so-called direct model. In this paper, the direct model is a numerical acoustic propagation model based on the wide-angle PE (WAPE) due to Claerbout [1]. Some applications of the variational approach in tomographic inversion is given in [2], where the gradient of the cost function is computed by the adjoint approach using the semi-automatic adjoint code generator YAO via modular graph approach [2,3,4].

In tomographic inversion, two problems have to be considered: first, the observed and computed measurements cannot be identical due to correlated additive noise on the vertical receiver array (VRA) signals and uncertainties in the physical modelling of the shallow water environment and acoustic propagation, and, second, the inverse problem is ill-posed having several different solutions. Different approaches have been proposed to overcome this second problem. One can use a regularization method involving the introduction of a penalty term in the cost function $J$ [2].

Another approach proposes to restrict the search of the control vector, which is here the vertical sound speed profile $c = c(z)$, to a subspace of reduced dimension defined by Principal Components Analysis (PCA) [5,6,7]. The drawback is that the dimension of the PCA subspace is provided implicitly by the PCA methodology without any criterion of optimality, and does not explicitly introduces a penalty term into the cost function.

The present work proposes to address the issues of dimensional reduction and regularization by using a Probabilistic Principal Component Analysis (PPCA) model to decompose the sound speed profile. This approach allows the intrinsic dimension of the data (here, the sound speed profile) to be determined, and provides the necessary a priori knowledge required for the regularization of the variational inversion solution. The PPCA model associated with Bayesian formalism allows us to define a generalized cost function with a penalty term. This approach is applied to acoustic data synthesized from NCOM oceanic model predictions, obtained during the MREA BP’07 experiment, southeast of Elba, Italy.

The paper is organized as follows. Section 2 reviews the variational approach and the chosen cost function together with its background terms. Section 3 introduces the PPCA model and the associated Bayesian formalism. Section 4 demonstrates the adequacy of the PPCA approach using MREA BP’07 ocean prediction data and synthetized acoustic data.

2. VARIATIONAL TOMOGRAPHIC INVERSION

The principle of variational inversion consists of minimizing a given distance between the measurements and the outputs of the so-called direct model whose input parameters (the quantities to be retrieved) are adjusted to obtain the best fit between the observed and simulated measurements (here, acoustic pressures on a vertical array). Here, the used direct acoustic model, denoted by $G_f$, is based on the wide-angle parabolic equation (WAPE) model [1]. For further details on $G_f$, see [2]. For a given sound speed profile, at a given time $\tau$, $c(z,\tau)$, a predicted vector field $\psi = G_f[c]$ can be computed at each of the $N$
elements of a vertical receiver array (VRA) for frequency \( f \), and compared to the measurements (processed acoustic signals \( s_j(t), j = 1, \ldots, N \)). The mismatch between the computed and observed values is quantified by using the following cost function, proposed in \[8\],

\[
J_o(c) = \text{tr} \hat{R} - \psi^* \hat{R} \psi \psi^* \psi,
\]

(1)

where \( * \) is the Hermitian transpose operator, \( \text{tr} \) is the trace operator, \( \hat{R} \) is the estimated spatial correlation matrix at frequency \( f \) and \( \psi^* \hat{R} \psi \) is the linear Bartlett processor. The matrix \( \hat{R} \) is estimated from the acoustic signals \( s_j(t) \) \[8\]. The gradient of \( J_o \) is computed by the adjoint approach which is implemented, here, by using the semi-automatic adjoint code generator YAO \[4\]. For further details on the implementation, see Refs. \[2,3\]. Because of additive noise on the VRA signals, uncertainties about the environment, and inaccuracies in the acoustic propagation modeling, \textit{a priori} information about the desired control vector must be introduced. A well-known procedure used in data assimilation is to modify the cost function such as that:

\[
J(c) = \frac{1}{T} J_o(c) + J_b(c),
\]

(2)

where \( J_o(c) \) is given by (1) and

\[
J_b(c) = (c - c_b)^T B^{-1} (c - c_b).
\]

(3)

The vector \( c_b \) is called the “background”, and \( B \) is the covariance matrix of the distance to the background. Equation (3) leads to a local search of the desired control vector \( c^* \) in the vicinity of \( c_b \). In the cost function (2), \( T \) is a continuous hyper parameter which determines the weighted compromise between \( J_o(c) \) and \( J_b(c) \). Due to the high dimensionality of the \( c(z, \tau) \) vectors and the strong correlation between their components, it becomes difficult to estimate the matrix \( B^{-1} \), which is often ill conditioned. A possible approach is to introduce an \textit{a priori} information constraint explicitly by restricting the subspace of possible control vectors and removing the background term. This explicit approach has been introduced for environmental inversion \[5,6,7\] by using Principal Component Analysis (PCA), also known as Empirical Orthogonal Functions (EOF). Another transformation, which is implicit, consists of using the Probabilistic PCA model (hereinafter called PPCA) and the Bayesian formalism.

### 3. PPCA APPROACH TO VARIATIONAL INVERSION

In the following, \( c(z, \tau) \) is assumed to be evaluated at \( M \) points of the discrete space with respect to depth \( z \). The Probabilistic PCA (PPCA) model \[9,10\] allows the control vector \( c \) to be interpreted probabilistically. It introduces an explicit latent variable \( \eta \in \mathbb{R}^q \) (\( q << M \))
whose prior distribution $N(0, I_q)$ is isotropic and normal, where $I_q$ is the identity matrix of order $q$. Let us assume that:

$$c = W\eta + c_b + \varepsilon,$$

(4)

where $\varepsilon \sim N(0, \kappa^2 I_M)$ is a stochastic isotropic and normally distributed process with standard deviation $\kappa$, $I_M$ the identity matrix of order $M$, $W$ a $M \times q$ matrix of range $q$ and $c_b$ a vector over $IR^M$. The columns of $W$ define a linear subspace $E_q$ of dimension $q$ over $IR^M$, and $W\eta + c_b$ represents the associated affine subvariety, which contains the vector $c_b$. The different profiles $c$ can be considered as the sum of a vector belonging to the affine subvariety and a noise $\varepsilon$. Under these conditions, it can be shown that the profile $c$ is normally distributed, that its mean is the vector $c_b$, and that its variance-covariance matrix is given by:

$$B = WW^T + \kappa^2 I_M.$$

(5)

The matrix $B$ can be determined by estimating the model parameters $(W, c_b, \kappa^2)$ based on an observation data set $A$. In the following, $A$ is a subset of sound speed profiles and locally represents the control vector. Through Maximum Likelihood (ML) estimation [9,10], the three parameters of the PPCA model can be determined. A possible optimal solution is such that:

- $c_b$ is the mean of the data set $A$;
- $W = U(L - \kappa^2 I_q)^{1/2}$ where $U = (u_1, u_2, ..., u_q)$ is comprised of the first $q$ eigenvectors of the empirical variance-covariance matrix of $A$. $L$ is a diagonal matrix $(q \times q)$, whose elements are the corresponding eigenvalues $\lambda_i$;
- and finally,

$$\kappa^2 = \frac{1}{M - q} \sum_{i=q+1}^{M} \lambda_i.$$

(6)

The sum $\sum_{i=q+1}^{M} \lambda_i$ represents the residual variance of the data, not taken into account by the first $q$ principal axes. Therefore, $\kappa^2$ is the average of the residual variance of the remaining $(M - q)$ principal axes. If one assumes that the data subset $A$ is statistically representative of the control vector $c$ and can be generated by the PPCA model (4), the residual variance must be evenly distributed among the $M - q$ remaining principal axes. Thus, the number of axes $q$ must be chosen in such a way that this property is verified. On the other hand, if $\sum_{i=q+1}^{M} \lambda_i$ is small enough and the number $M - q$ of remaining axes is large enough, the value of $\kappa^2$ found when estimating the residual variance (6) is small. The choice of $q$ is not as critical as for the explicit approach of PCA. Only a few values of $q$ will be sufficient and this choice will lead to a rigorous inverse methodology. In this case, the parameter $\varepsilon$ in (4) turns out to
be a normally distributed noise, and all of the useful information in $A$ is contained in the principal affine linear subspace of dimension $q$. As a consequence, $c = W\eta + c_b$, where $\eta$ is the latent variable associated with $c$, and is normally distributed since $\eta \sim N(0, I_q)$. The affine subvariety $E_q + c_b$ of dimension $q$ and the density function $N(0, I_q)$ totally describe the control vector $c$.

The solutions of the variational inversion must be found in this subvariety. The latent variable becomes the control vector and the first term of the total cost function (2) can be rewritten as a function of $\eta$ as:

$$J_o(c) = J_o(W\eta + c_b) = \varphi_o(\eta).$$

Using Bayesian formalism, the following total cost function is found:

$$J(\eta) = \frac{1}{T} \varphi_o(\eta) + \eta^T \eta,$$ since the a priori information on the new control vector $\eta$ is known.

The inverse problem is solved by minimizing (8) with respect to $\eta$. The main advantages of the PPCA reformulation (8) are that the difficulty of estimating $B^{-1}$ is circumvented, and the control vector becomes $\eta$, which has a smaller dimension ($q \ll M$), and whose components are not correlated. This provides improved preconditioning for the minimization process. Finally, this approach selects $q$ according to its ability to fully reproduce the process without any loss of useful information. The minimization process thus leads to a realistic solution, which follows the a priori distribution of the data. In the following, one illustrates this methodology with a realistic test of acoustic tomography in a shallow water environment.

4. NUMERICAL VALIDATION USING MREA BP’07 DATA

During the MREA BP’07 experiments, southeast of Elba Island in the Mediterranean Sea, a large set of in-situ acoustic and environmental data were collected for developing geoaoustic inversion [11] and acoustic tomography [12] methods. In the present work, 4-day prediction results of the temperature and salinity fields, obtained with NCOM model [12], were used to create a set of sound speed profiles (SSP). The SSPs were predicted at 1-h interval given rise to 96 profiles (Fig. 1). See [11,12] for details on the experimental data and ocean modelling.

The MREA BP’07 profiles represent the evolution of the range-averaged SSP along a 15-km transect (A-B in Fig. 1 of Ref. [12]) for a duration of four days, and constitute the data set $A$. The PPCA provides a model of this behaviour. Figure 2 shows the Cumulative Percentage of Total Variability (CPTV) with respect to total energy, for each of the first 15 PPCA axes. For $q = 8$ axes, the CPTV is approximately 99%. Following the discussion about the choice of $q$ in the previous section, and the results shown in Fig. 2, values of $q$ greater than 8 provide an appropriate trade-off.
Figure 1. MREA BP'07 sound speed profiles (blue) and their ensemble average (cyan). The sediment layer and bottom geoacoustic properties are given in Ref. [8].

Figure 2. Energy of the first 15 axes. The bar graph represents the percentage of total variability of each PCA axis. The cumulative percentages of total variability (CPTV) are indicated by circle. The first eight axes include almost 99% of the total energy.

Under twin experiments, a monochromatic acoustic source of frequency of 500 Hz is positioned at \( z_s = 69.2 \) m depth and the seafloor depth is \( z_l = 113.1 \) m. The synthesized acoustic signals are sampled on a vertical array (VRA) of 32 hydrophones, spaced at 2 m intervals, between the depths of 37.2 m and 99.2 m. To simulate a realistic scenario, we added a normally distributed noise of amplitude 0.01 to the acoustic signals. This gives a signal-to-noise ratio comparable to the levels observed during the Yellow Shark experiment [8]. These simulated measurements were later used to compute the matrix \( \hat{R} \) used to determine the observation error (1).

In this article, we consider a snapshot in the tracking scenario developed in Ref. [14]. The profile \( c^* \) farthest from \( c_h \) in \( A \), based on the Euclidian distance, is selected as the “actual profile” to be retrieved. In other words, the synthesized acoustic signals were generated using \( c^* \) as input to the WAPE model \( G_f \). The initial values at the start of the inversion were given as \( \eta = 0 \). It corresponds to mean profile \( c_h \).
For the dimension of the reduced subspace we choose \( q = 8 \) as the largest possible value allowed by the PPCA model. We thus minimized the cost function (8) with respect to \( \eta \), which is a 8-dimensional normally distributed variable \( \eta \sim N(0, I_q) \). The “L-curve” method [13] is used to estimate the value of \( T \) in (2), which is here \( T_{LC} = 2.5 \times 10^{-4} \).

Figure 3 shows the true centred profile \( (c_{ztrue} = c^* - c_b) \), together with that estimated using \( T_{LC} \) as regularization parameter with 8 axes \( (c_{zest1} = c_{est1} - c_b) \). Also shown is the estimated centred profile \( (c_{zest2} = c_{est2} - c_b) \), with the background term removed, in order to illustrate the usefulness of this term.

![Figure 3](image)

**Figure 3.** Acoustic tomography results using 8 PCA axes. Starting from \( \eta = 0 \) (corresponding to the mean profile) the true centred profile is shown by the blue solid line \( (c_{ztrue}) \). The profile that was estimated using \( T_{LC} = 2.5 \times 10^{-4} \) as the regularization parameter is shown by the red dashed line \( (c_{zest1}) \), and the one estimated with the background term removed is shown by the black dotted line \( (c_{zest2}) \).

Clearly, the control vector \( c_{est} \) estimated according to inverse PPCA methodology is close to the “true” profile \( c^* \) to be retrieved. We can also see from the estimated profile \( c_{est2} \) that the performances are improved when the background term is used.

For the present twin experiment we have used a monochromatic acoustic source but the methodology can be easily extended to variational inversion across multiple frequencies, like in Refs. [2,3,14].

5. CONCLUSION

In this paper we have presented a variational approach to solve the inverse problem of acoustic tomography in a shallow water environment, based on the probabilistic principal component analysis (PPCA). We have shown that this method can provide an appropriate representation of the regularization term in the cost function, and can significantly reduce the number of control parameters. In the context of twin experiments with additive measurement noise, the methodology proposed here leads to satisfactory and robust results.
6. ACKNOWLEDGEMENTS

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REFERENCES

A COMPARISON OF VARIATIONAL AND KALMAN FILTERING PROCEDURES FOR THE ASSIMILATION OF ACOUSTIC TOMOGRAPHY DATA

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Abstract: In this paper, we compare an adjoint-based variational data assimilation approach with ensemble-based Kalman filter procedures, and in particular the Unscented Kalman filter (UKF), for the full-field acoustic tomography of shallow water environments. In this inverse problem we seek to track the evolution of a range-average sound-speed profile in a vertical slice of a shallow water environment from multi-frequency, complex acoustic pressure data recorded on a vertical array. The UKF is a nonlinear extension of the Kalman filter based on a deterministic sampling of the assumed Gaussian distribution of the state estimation errors. In contrast to optimizing recursively a cost function that quantifies the fit between data and model, as in the variational procedure, Kalman filtering provides the minimum error variance estimate of the states analytically derived from the Gauss-Markov assumptions of the process and measurement models. Our results show that the Kalman filtering performs equally well and presents the considerable advantage of not needing any analytical derivation or numerical implementation of the adjoint model, and associated computation. Moreover, update of the uncertainty of the estimates is embedded in the Kalman filtering scheme.

Keywords: adjoint model, ensemble Kalman filter, shallow water tomography, variational approach.
1. INTRODUCTION

Over the past three decades, data assimilation (DA) techniques have been an important topic of research for the meteorological and oceanographical communities. There exist several DA methods, with different advantages or disadvantages depending on the field of application. Most of the DA techniques can be gathered in two general categories: variational methods and stochastic filtering methods. A comprehensive review can be found in [1, 2].

The application of data assimilation techniques to ocean acoustics problems has gained an increasing interest over the last decade. This paper compares the performances of two inversion methods, inspired by DA techniques, on a full-field shallow water tomography application. The first method, based on the adjoint of a full-field acoustic propagation model, can be related to a variational technique. The second method is based on ensemble-based Kalman filtering, which belongs to the category of stochastic filtering methods.

Variational techniques for data assimilation were initiated in the meteorological community in the 70's and remain the main stay of most weather prediction systems in the world. Recently, in [3, 4] we have applied this approach, based on analytical and semi-automatic computer generated adjoint models, to inverse problems in shallow water acoustic tomography. It was shown that the adjoint, variational approach combined with multi-frequency, broadband optimization, yields robust and accurate inversion results for the geoaoustic properties of the subsea floor as well as for the sound-speed profile of the water column. Stochastic methods, based on optimal interpolation and Kalman-type filters, are currently undergoing much research and innovation. A complete study can be found in [5]. The ensemble Kalman filter, introduced by Evensen [6], was introduced to overcome the high computational costs and to better treat nonlinear problems.

This paper is organized as follows: the two first sections expose briefly the adjoint-based and Kalman-based implementations of our inversion methods for full-field tomography purposes. The synthetic environment and the simulation setup are described in Section 4. Section 5 presents and discusses the results. The paper is concluded in Section 6.

2. ADJOINT-BASED OPTIMIZATION

If the environment is described by the model vector \( m, j = 1, 2, \cdots, M \), the cost function \( J \) is a function of predicted (replica) sound field vectors \( \psi(m, \omega_l) \) and the measurements \( \psi_{\text{obs}, l} \) across an \( N \)-element vertical array at the temporal frequencies \( \omega_l, l = 1, 2, \cdots, L \). A generalized maximum likelihood estimate of the model vector \( m \) is obtained by minimizing

\[
J(m) = \frac{1}{L} \sum_{l=1}^{L} \left[ \tr \hat{R}(\omega_l) - \frac{\psi^\dagger(m, \omega_l) \hat{R}(\omega_l) \psi(m, \omega_l)}{\psi^\dagger \psi} \right],
\]

where \( \hat{R}(\omega_l) \) are the measured spatial correlation matrices at the frequencies \( \omega_l, l = 1, 2, \cdots, L \).

The adjoint method provides a mathematical means to calculate gradients \( \nabla_m J \) of the cost function which can be efficiently minimized with respect to the model parameters \( m \) using gradient descent methods [4].

As a local optimization method, the adjoint approach is attractive for data assimilation problems where rough a priori estimates of the control parameters are available from the previous time step (Fig. 2).
3. UNSCENTED KALMAN-BASED OPTIMIZATION

A Kalman filter is typically derived from a Gauss-Markov state-space model. For the acoustic tomography application, the environment constitutes the state vector \( \mathbf{m} \), \( j = 1, 2, \ldots, M \), and the observation vector contains the complex acoustic pressure field measurements \( \mathbf{\Psi} \) across an \( N \)-element vertical array at the frequencies \( \omega_l, l = 1, 2, \ldots, L \) [9]. The assumed (discrete) Gauss-Markov state-space model is given by

\[
\begin{align*}
\mathbf{m}(t_k) &= \mathbf{m}(t_{k-1}) + \mathbf{w}(t_k) & \text{[transition]} \\
\mathbf{\Psi}(t_k) &= \mathcal{H}(m(t_k), \omega_l) + \mathbf{v}(t_k) & \text{[measurement]}
\end{align*}
\]

(2) (3)

where \( \mathbf{w}(t_k) \) and \( \mathbf{v}(t_k) \) are zero-mean Gaussian random vectors of covariance \( R_{ww} \) and \( R_{vv} \), respectively. As in the cost function (1), the measurement function (3) includes the maximum likelihood estimate for the source phase and magnitude. It is noteworthy that this function embeds the acoustic propagation model and is therefore nonlinear.

Kalman filters enable a prediction/correction formulation. At each time step, the predicted SSP is corrected with acoustic measurements weighted by the Kalman gain (Fig. 3). The common Kalman filter applied to nonlinear systems is the extended Kalman filter (EKF). More recent nonlinear extensions of the Kalman filter use an ensemble of models to represent the underlying distributions, by Monte-Carlo sampling, as in the ensemble Kalman filter, or EnKF [6], or by combining deterministic sampling and statistical transformations, as in the unscented Kalman filter, or UKF [8]. Such extensions are required when model nonlinearities are important.

4. SYNTHETIC DATA AND ENVIRONMENT

We use oceanographic data obtained in the framework of the MREA/BP07 experiments that were conducted from 16 April to 4 May 2007 in the Ligurian sea, South East of Elba Island [10]. The environmental scenario is a 2-day prediction of the sound speed field along a 3-km transect, generated with the Navy Coastal Ocean Model (NCOM) [11].

The inversion is carried out jointly across 7 different source frequencies \( \{200, 250, 315, 400, 500, 630, 800 \text{ Hz} \} \) and the complex field is sampled on a vertical receiver array (VRA) at a range of 3 km. The acoustic source is positioned at \( z_s = 45 \text{ m} \) depth and the total water depth is 113.1 m. A partial water column spanning VRA is used with
32 hydrophones, equispaced at depths between 8 m and 70 m. The bottom geoaoustic parameters are given in [12]. This configuration corresponds to the first 3 km of transect A-B of the recent MREA/BP07 sea trial [10].

The corresponding 48 realizations of the range-averaged sound-speed profiles are shown in Fig. 4. It illustrates clearly the temporal variability of the range-averaged profiles over the 2-day period.

5. SIMULATION RESULTS

We compare both methods on a tracking run for estimating the three first EOF coefficients, where the synthetic acoustic observations are generated with the full series of EOF coefficients representing the original sound-speed profile (Fig. 5a).

At the first inversion the EOF parameters are initially set to zero, i.e., the initial profile is set equal to the average profile of the EOF database.

Variational and UKF methods are in very good agreement. For both methods, the first two EOF coefficients are well retrieved with respect to the coefficients of the series representation of the true profile. The third coefficient follows the general trend of the true profile representation but is not retrieved as closely as the first two coefficients.

In addition to the evolution of the estimated EOF coefficients, Fig. 5b shows at each time step the corresponding depth-integrated error

\[ \Delta c = \sqrt{\sum_i |c_{\text{true}}(z_i) - c_{\text{estim}}(z_i)|^2 / \sum_i |c_{\text{true}}(z_i)|^2} \]

between the original profile and the reconstructed profile using the three estimated EOF coefficients obtained by both methods.

For the adjoint-based method, each individual inversions consist of approximately 16–18 iterations. Although the Kalman filtering involves only one inversion step, it should be noticed that the UKF scheme requires however several forwards run to capture correctly the nonlinearity of the system (see [8] for more details). This number of iterations is directly related to the state and measurement dimensions. In contrast to the Kalman filters, the adjoint-based method does not require several time steps to converge because
Figure 3: Temporal variability of the range-average sound-speed profiles over a 2-day period (28–29 April 2007) during the MREA/BP07 sea trial. The time sampling interval is 1 hour. The warming of the upper layer during the afternoon hours of both days leads to a clearly visible increase of sound speed. There is also a progressive increase of the mixed layer thickness.

of its recursive scheme of optimization. Convergence speed in Kalman filters is typically handled by the tuning of the system covariance matrices.

6. CONCLUSION

In this paper, two different inversion methods were compared on the same problem of full-field acoustic tomography of a shallow water environment, involving the tracking of the range-averaged sound-speed profile of a vertical transect, by inverting the multi-frequency complex acoustic field on a vertical array. The continuous monitoring of the water-column sound-speed profile in terms of the time-varying coefficients of the strongest EOF coefficients over a 2-day period is well performed by both the variational and Kalman-based methods. The low complexity and flexibility of the Kalman-based method, here an unscented Kalman filter, is however very attractive. Furthermore, the embedded processing of the uncertainty in the Kalman schemes is an important attribute for robustness when applying to real data. Hybrid schemes could be developed to enhance the accuracy of the estimates by local adjoint-based optimization on the Kalman filtering outputs.

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Figure 4: Acoustic monitoring of the time-varying sound-speed profile of the 3-km transect over a 2-day period. Acoustic observations were synthesized using the original SSP profiles. (a) The evolution of the three adjoint-based (continue, red) and UKF-based (dashed, blue) estimated EOF coefficients is shown together with the first three EOF coefficients (solid, gray) of the full series representation (normalized). (b) Evolution of the depth-integrated error Δc between the original profile and the reconstructed profile using the three estimated EOF coefficients by variational method (solid, red) and UKF method (dashed, blue). The error between the original profile and its 3-EOF representation indicates the lower error bound (solid, gray).

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An operational approach to Sound Speed data assimilation in high resolution ocean models

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Abstract: Sound propagation in the ocean and sonar performance are determined by multiple environmental parameters that include water column temperature and salinity, usually estimated by local observations combined into single snapshot analysis and by numerical models that provide space-time extrapolated snapshots through a forecast range. Since local observations are rarely enough to fully represent the area of interest special attention has been given in the recent past into inverting acoustic propagation anomalies to obtain sound speed or temperature profiles. The extrapolation in both space and time from these initial observations can be carried by the ocean forecast models but can have large uncertainties due to errors in the initial conditions, boundary and forcing fields and unrepresented physics (numerical approximations and sub-grid variability). The combination of models and local observations has been implemented in operational ocean models using data assimilation of local state-parameters observations and has shown to lead to improved consistency of the sound speed forecasts. In the recent past some work has also been done in bringing synthetic ocean profiles derived from the acoustic anomalies into the model assimilation process, though requiring a special care due to their difference in the representation and instrumental when compared to local in-situ profile measurements some problems. This work will discuss a possible operational methodology to combine both in-situ temperature and salinity profiles and high resolution (space and time) sound-speed profiles derived from acoustic measurements using a Kalman filter tracking scheme, into the assimilation process. The method aims to improve local sound speed short range forecast within the US-Navy NCOM-NCODA framework, combining standard assimilation and model ensembles training techniques. Results will be discussed regarding local improvements in 12 to 24 hour sound speed forecasts, when compared with model estimate not using the acoustic data and observations persistency. The data used in this analysis was obtained during the BP07 sea trial that took place off Italy in the Ligurian Sea, south of Elba Island.
Structured Session 16

Buried Object Detection using Underwater Nonlinear Acoustics

Organizer: Eugeniusz Kozaczka
EXPERIMENTAL INVESTIGATION OF THE GDANSK BAY BOTTOM STRUCTURE BY MEANS OF NONLINEAR ACOUSTIC METHOD

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Abstract: The specific properties of generated parametric waves, called the waves of difference frequency, that are generated as the result of the propagation of the waves of high intensity are used for many years. These unique properties of the parametric sources are first of all low frequency of the wave and narrow directivity pattern side lobes. Taking into account higher attenuation of the wave in the sediments in the comparison with water the sonar that is based on nonlinear generation of the waves is a very good equipment for the investigation of the seabed. The results can be used for multiple purposes, including the determination of the structure of sediments of the seabed and for underwater archeological exploration. A very attractive application is to use these methods to search for items that are in the mud or buried in the bottom of the sea. The method of searching for objects covered with a layer of sediments is not effective while the classic sonars are used. The parametric sonars are filling the gap. While there is a real threat of placing underwater destruction measures, the methods of searching for objects that are in the mud, gain the additional importance. The paper presents the results of sounding of the selected areas of the Gdañsk Bay by means of the parametric sonar connected with complementary hydroacoustic devices, such as multi-beam and high resolution side sonar. There will be shown the acoustic images with a particular focus on the results of the sounding of the seabed.
MULTI-VIEW TARGET CLASSIFICATION IN SYNTHETIC APERTURE SONAR IMAGERY

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Abstract: This work proposes an elegantly simple solution to the general task of classifying the shape of an object that has been viewed multiple times. Specifically, this problem is addressed in the context of underwater mine classification where the objective is to discriminate targets (i.e., mines) from benign clutter (e.g., rocks) when each object is observed in an arbitrary number of synthetic aperture sonar (SAS) images. The proposed multi-view classification algorithm is based on finding the single highest maximum correlation between (i) a set of views of a training shape of interest and (ii) a set of views of a given testing object. Classification is performed by using this measure of similarity, which we term the affinity, directly. This approach obviates the need for explicit feature extraction and classifier construction. Moreover, the framework induces no constraints on the number of views that each object can possess. Promising experimental results using real SAS imagery demonstrate the feasibility of the proposed approach for multi-view classification of underwater mines. In particular, it is shown that classification performance improves dramatically as the number of views of the objects increases.

Keywords: Target classification, mine detection, multi-view classification, data fusion, synthetic aperture sonar (SAS)

1. INTRODUCTION
This work proposes a solution to the general task of classifying the shape of an object that has been viewed multiple times. Specifically, this problem is addressed in the context of underwater mine classification where the objective is to discriminate targets (i.e., mines) from benign clutter (e.g., rocks) when each object is observed in an arbitrary number of synthetic aperture sonar (SAS) images.

If a mine is observed from one orientation, it may be difficult to distinguish it from a rock; however, viewing it from a second orientation may reveal previously obscured characteristics that differentiate it from a rock. In general, the information accrued from multiple views of an object should translate into improved classification performance. This basic concept motivates the collection of multi-view data for the classification of underwater mines.

The nature of data collected for the underwater mine classification problem differs from most multi-view classification problems in that an arbitrary number of views of a given object will be an unordered set of observations. In this work, we propose a new classification method that combines information from multiple views, when each object can be viewed an arbitrary number of times. Specifically, the multi-view classification approach proposed here is based on finding the single highest maximum correlation between (i) a set of views (i.e., images) of a training shape of interest and (ii) a set of views of a given testing object. Classification is performed by using this measure of similarity, which we term the affinity, directly.

This approach is particularly well-suited for the underwater mine classification problem for several reasons. For one, it fully exploits the recent advances in SAS systems by focusing on the detailed shape information of the objects that the high-resolution imagery provides. But more importantly, the approach also overcomes the unique challenges presented by the general underwater mine classification problem that cause standard (single-view) classification approaches to fail.

A fundamental assumption of machine-learning algorithms is that the underlying mechanisms (and hence statistics) that generate the training and testing data are the same [1]. In the underwater mine classification problem, however, this implicit assumption is often violated because different types of clutter objects can be encountered at different sites. As a result, a classifier learned from clutter training data collected at one site often will not generalize well on testing data collected at a different site.

Furthermore, in the underwater mine problem, the universe of target shapes that one is interested in detecting and classifying is typically small. However, within this small class of target shapes, marked differences exist among the different shapes. As a result, the features that are salient for certain shapes are not relevant for classifying other shapes. This fact induces a need for larger feature spaces, which in turn necessitates more training data to build a reliable classifier.

The fact that the image of a target is highly aspect-dependent further increases the need for more training data (namely, from multiple aspects). For example, the image of a broadside cylinder will look very different from the image of an end-fire cylinder. In general, the relative paucity of training data — of different targets, at different ranges, at different aspects, and in different site conditions — contributes to the difficulty of building a robust classifier.

The approach proposed in this work combats these challenges inherent to the underwater mine classification task by obviating explicit feature extraction and classifier construction. Moreover, in the proposed multi-view classification method, there is no constraint on the number of views that can be handled. Importantly, the elegantly simple approach is naturally suited to allow each object to be viewed an arbitrary number of times.

The remainder of this paper is organized in the following manner. The proposed multi-view classification approach is described briefly in Section 2. Experimental results using real
SAS imagery are presented in Section 3. Concluding comments are made in Section 4. Page constraints limit the detail into which the method can be described here; the interested reader can find a longer, more detailed presentation of this work in [2].

2. MULTI-VIEW CLASSIFICATION

The multi-view classification approach we propose is based on computing the maximum correlation between (i) a set of views of a known object of interest and (ii) a set of views of a test object of unknown identity.

We define the affinity to be the maximum over all of the correlation maxima obtained between every possible combination of a view of the training shape and a view of the testing object. That is, the affinity is a quantitative measure of the highest degree of similarity between the training shape and the testing object, considering all of the available views of each.

By construction, the affinity between any two objects will be monotonically increasing with increasing numbers of views (of one or both of the objects). Admittedly, there may be cases, for some limited number of views, where the affinity between two objects of different shapes will actually be greater than the affinity between two objects of the same shape. However, as the total number of views increases, the increase in affinity for the latter will in general be faster, and will therefore surpass the affinity of the former. That is, with enough views, the affinity between two objects of the same shape would be greater than the affinity between two objects of different shapes. This hypothesis relies on the belief that, eventually, the two objects of the same shape will be observed at similar enough orientations such that they appear nearly identical. It also assumes that two objects of different shapes will never appear exactly identical at any orientation.

It is this affinity — “the maximum of the maxima of the correlations” — with which classification decisions will be made. Specifically, if the affinity between a testing object and the given training object of interest is above a set threshold, the testing object is classified as the training object's shape. Otherwise, it is declared to be some other shape.

The proposed multi-view classification algorithm assumes that the target types (i.e., shapes) of interest are known a priori. However, all supervised classification approaches always make this same assumption. If one does not know which targets are to be classified, devising a set of sensible features is not possible. Moreover, standard classification approaches implicitly assume that the targets in the training set will match the targets in the testing set. Therefore, our assumption about knowing the target types of interest is completely justified.

It should also be noted that the proposed approach does not require a training data set of both targets and clutter. Instead, training views of only the targets of interest are required. This aspect is important because significantly different types of clutter objects can be found in different areas (e.g., at a training site and at a testing site). Thus, our approach circumvents the problem of training on clutter objects that are not representative of the type of clutter objects that one may encounter at a new testing site. Moreover, this approach also obviates the feature extraction and classifier learning processes, thereby avoiding the difficulties associated with them.

Finally, the proposed multi-view classification algorithm is predicated on the fact that multiple views of an object will be available. Nevertheless, if a given object is viewed only a single time, the proposed classification approach is still valid.
3. EXPERIMENTAL RESULTS

3.1. Data Set

In April-May 2008, the NATO Undersea Research Centre (NURC) conducted the Colossus II sea trial in the Baltic Sea off the coast of Latvia. During this trial, high-resolution sonar data was collected by the MUSCLE autonomous underwater vehicle (AUV), which is equipped with a 300 kHz sonar.

On 29 April, before the data collection was performed, six known targets — three cylinders and three truncated cones — were placed on the seabed. The vehicle then made multiple passes over the target area in different orientations. The data that resulted from this collection was then processed into SAS imagery.

A (contact) detection algorithm [3] that employs a matched filter to find highlight-shadow patterns characteristic of mine-like objects in the images was then applied. This detection algorithm generated a total of 2,395 detections. For each detection, a SAS image chip was then extracted.

Subsequently, all detections (i.e., views) of a given object were grouped together by using vehicle-recorded latitude and longitude information. Based on this data-association process, it was established that a total of 317 unique objects comprised the 2,395 detections (for a mean of 7.55 views per object). The maximum number of views of any object was 27.

3.2. Experimental Procedure

The objective of these experiments is to assess the classification performance of the proposed affinity-based approach in terms of discriminating the six targets — three cylinders and three truncated cones — from clutter as a function of the number of views of each object. This goal is accomplished via the following experimental procedure.

Assume one of the six objects of interest is labeled training data. The remaining 316 objects are treated as unlabeled testing data that we wish to classify.

Randomly select $n$ views of each object (or all of an object’s views if fewer than $n$ views are available). For each testing object, compute the affinity that the set of selected views of the object has with the set of selected views of the given training object.

To determine the performance for correctly classifying a testing object that is the same target shape as the training object, one can simply count the number of other testing objects that have a higher affinity than the affinity of the matched testing object. This number is exactly the number of false alarms, from which the probability of false alarm can be easily deduced.

In the above procedure, the affinity values that were obtained were between two objects when each was viewed no more than $n$ times. However, each affinity value corresponds to only one possible pair of randomly selected views. Therefore, to increase the statistical strength of the experimental results, we repeat this entire process 1000 times — in lieu of considering every possible combination of views, which is computationally infeasible — where new sets of views for each object are randomly selected in each iteration.

Thus, for a given maximum number of views of each object, $n$, we have 1000 probability of false alarm values when trying to classify a testing object of the same target shape as the
training object. The mean and variance of the probability of false alarm can then be readily calculated from the 1000 values.

All of the above is then performed for each possible value of $n$, from 1 to 27, to obtain performance as a function of the number of object views.

### 3.3. Classification Results

Fig.1 shows, for each maximum number of views of any object to be considered, the mean classification performance (averaged over the 1000 iterations, in which each iteration uses a random selection of each object's views) in terms of the probability of false alarm when each of the six objects of interest — three truncated cones and three cylinders — is treated as the training target. The error bars in the figures represent one variance (i.e., square of the standard deviation) above and below this mean value.

As can be observed in the figure, the classification performance improves dramatically as the number of views of each object increases. Moreover, the benefit of obtaining additional views is most significant when only a limited number of views is possessed.

It should be noted that the targets of interest typically possessed more views than most of the clutter objects in this data set. This fact is because the only views (i.e., images) that are considered in the classification stage are those that pass the detection stage's prescreener.

A view of a target from any aspect will usually appear mine-like, and hence pass the detection stage. In contrast, many clutter objects appear mine-like from only certain aspects, and hence pass the detection stage only occasionally. As a result, target objects will consistently be detected while clutter objects are detected only sometimes. Hence, target objects will have more views than clutter objects in the classification stage. In turn, target objects will have more opportunities to achieve a high affinity with a given training target of interest. This ostensible bias is actually desirable for the proposed approach because classification performance improves as the number of views of a target increases.

### 4. CONCLUSION

This work presented an elegantly simple solution for the general task of classifying the shape of an object that has been viewed multiple times. Specifically, this problem was addressed in the context of underwater mine classification where the objective is to discriminate targets (i.e., mines) from benign clutter (e.g., rocks) in SAS images.

The proposed approach was based on computing the maximum correlation between each of the multiple views of a known shape of interest with each of the multiple views of a test object of unknown shape. Specifically, the maximum of that set of correlation maxima, which we termed the *affinity*, was used directly to make classification decisions. A set of experiments with real SAS imagery demonstrated the feasibility of using the proposed method for multi-view classification of shapes.

A more detailed presentation of this work (in which the actual multi-view SAS imagery of the objects is shown) can be found in [2].
Fig. 1: Performance of proposed approach for correctly classifying the objects noted in the legends, as a function of the number of views considered for each object in the data set, when the training object is that shown in the subfigure titles.

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SYNTHETIC APERTURE SONAR:
A TOOL IN UNDERWATER ARCHAEOLOGY

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Abstract: The principle of synthetic aperture sonar (SAS) is coherent combination of successive pings such that the along-track resolution is improved. The along-track resolution in SAS images become range and frequency independent. SAS technology is efficient – it provides high resolution and large area coverage simultaneously. This makes SAS an ideal tool in search for small objects over large areas. The first HUGIN 1000-MR autonomous underwater vehicle (AUV) was delivered to the Royal Norwegian Navy mid-2008. The main sensor is the HISAS 1030 interferometric SAS, which allows imaging at resolutions better than 5x5 cm, an area coverage rate exceeding 2 km\textsuperscript{2} per hour, and full-swath, co-registered, high resolution bathymetry. The vehicle is designed to be used in various military applications and other applications such as underwater archaeology, detailed mapping of the seafloor, and search. In this paper, we present the system and the benefit of SAS, and show SAS images of various objects and wrecks from HUGIN 1000-MR missions.

Keywords: synthetic aperture sonar, sonar imaging, search, underwater archaeology
1. INTRODUCTION

In 2008, the first HUGIN 1000-MR autonomous underwater vehicle (AUV), shown in Fig. 1, was delivered to the Royal Norwegian Navy (RNoN) [1]. The vehicle is equipped with the HISAS 1030 high resolution interferometric synthetic aperture sonar (SAS) [2], and an EM-3000 multibeam echosounder (MBE). The sensor package combined with state-of-the-art navigation on a well-proven vehicle, makes the HUGIN 1000-MR an excellent tool in search for objects over large areas of the seafloor [3].

A typical HUGIN AUV-mission is conducted in the following way. A mission plan containing waypoints for the vehicle is programmed. The tracks typically follow a lawn-mower pattern. The vehicle executes the mission plan, either supervised or autonomously, running along the tracks in optimal height over the seafloor (see the left panel of Fig. 2). The vehicle then returns and data are downloaded to the Post Mission Analysis (PMA) system. The operator, or a computer program, then investigates the data collected, in order to detect and classify objects of interest (see the right panel of Fig. 2).

In this paper, we briefly describe the benefits of synthetic aperture sonar, and show example SAS images collected by the HISAS 1030 on HUGIN 1000-MR.

Fig. 1: The HUGIN 1000-MR onboard the RNoN MCMV Hinnøy in August 2008.

Fig. 2: Left: The HUGIN Operator station with programmed vehicle tracks. Right: Visualization of sonar data in search for small objects.
2. BENEFITS OF SYNTHETIC APERTURE SONAR

Fig. 3: Synthetic aperture sonar (left) and sidescan sonar (right) coverage and resolution.

Synthetic aperture sonar (SAS) is different from sidescan sonar (SSS) in that the along-track resolution is independent of range [4], as illustrated in Fig. 3. This is a consequence of the increased length of the synthetic array (or aperture) as function of range. When successful SAS processing is performed, the SAS image can have much higher resolution than high resolution SSS [5]. For typical high resolution SSS, the imaging resolution is much poorer along-track than cross-track at far range. In SAS, square pixels of high resolution, independent of range is feasible.

Along-track resolution in SAS is also independent of frequency. This allows for very high resolution sonar imagery with much lower frequency. This has two major implications: 1) The sonar range is limited by the frequency dependent absorption in seawater [6]. In SAS, one can simply choose a lower frequency, and thereby lower transmission loss, while still maintaining high resolution. Note that the maximum range in SAS is also limited by the receiver array length and vehicle speed [7]. 2) SAS technology also makes it possible to produce high resolution imagery at low frequencies for which the acoustic waves penetrates into objects and sediments [8].

The HISAS 1030 interferometric SAS is a wideband widebeam sensor with 160 receiver elements in total (for both sides), with many different ways to process the collected data. We have developed a complete processing suite, named FOCUS toolbox, for processing of SAS data. In Fig. 4, we show some of the possible products we can produce. See [9] for further details.

Fig. 4: FOCUS SAS processing toolbox output product overview.
3. EXAMPLE IMAGERY FROM HUGIN 1000-MR AUV

Fig. 5: SAS image with 300 m range. The image shows the wreck of the German WWII submarine U735 outside Horten, Norway, at approximately 200 m water depth.

In this section, we show examples of SAS images made of data collected by HUGIN 1000-MR. Fig. 5 shows a SAS image with large area coverage (300 m range). The vehicle track is along the left edge of the image. The red thin vertical lines (along-track), indicate 25 m intervals in range. At approximately 175 m range, the wreck of the German WWII submarine U735 appears. Note the detail level in the zoomed area.

Fig. 6 shows SAS images of two different unknown wrecks. The image size is 50 x 50 m and the range to the center of the images is 80 m. The images illustrate the detail level in documentation of the seafloor.

Fig. 6: SAS images of two different wrecks outside Bergen, Norway. The water depth is around 340 m. Courtesy of the Royal Norwegian Navy.
Fig. 7: SAS image from the northern part of Norway, collected during winter 2009. The image shows the wreck of a German WWII seaplane Heinkel He 115. Courtesy of the Royal Norwegian Navy. The upper right image shows a photograph from wikipedia.org.

Fig. 7 shows a SAS image of a German WWII seaplane Heinkel He 115. The plane was probably dumped after WWII. The original length of the plane was 17 m and the wingspan was 22 m. Note the small part outside the right wing of the seaplane – this is probably one of the floats. Fig. 8 shows two example SAS images of small objects – in this case anchors. The very high resolution and fidelity makes it easier to properly classify the objects.

4. SUMMARY

Compared to other sensors available today, SAS increases survey efficiency. The technology provides high resolution and large area coverage rate at the same time. At typical sonar frequencies, resolution down to a few centimetres in range intervals up to hundreds of metres is feasible.

In 2008, the first HUGIN 1000-MR AUV was delivered to the Royal Norwegian Navy. The main payload sensor is the HISAS 1030 interferometric SAS. The range of the system is 200 m at vehicle speed of 2 m/s, which gives an area coverage rate better than 2 square kilometres per hour. The theoretical resolution of the system is better than 5 x 5 cm. Although being specifically designed for military use, the HUGIN 1000-MR AUV is a capable tool well suited for other applications such as underwater archaeology and search for small objects.

In August 2009, the RNoN will use their HUGIN 1000-MR to conduct a search through 45 square miles in the Barents sea for the remains of Roald Amundsen’s plane Latham 47, that disappeared in 1928. The plan is to conduct this search in less than two weeks. See [10] for more details on the search.
Fig. 8: Example SAS images of small scenes. Courtesy of the Royal Norwegian Navy.

5. ACKNOWLEDGEMENTS

The authors wish to thank the Royal Norwegian Navy Mine Warfare Service for the kind permission to use data recorded during Navy operations with the HUGIN 1000-MR.

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WATER TANK EXPERIMENT OF NONLINEAR ABSORPTION BY MARINE SEDIMENTS

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Abstract: Examination of the seabed by means of nonlinear methods of generating waves is associated with the necessity of determining the fundamental characteristics of the field of used sources. The basic of them are: the frequency of radiated wave, directivity pattern of the source, the angular and spatial resolution of the source and wave attenuation. For the classical sources of acoustic waves information is known and usually provided by the producer of measuring equipment. The knowledge of the parametric sources is not so common especially if it comes to the propagation of the waves into the sediments could be said that only fragmented. For this reason, it is necessary before the measurements to make the investigation of the laboratory and verifying nature. Since these are measures that require high precision and stable conditions, they are usually made in the designed for this purpose measuring water tanks. The paper presents the results of measurements in the measuring water tank. The complete results consist of the characteristics of the radiation of the sources with particular emphasis on spatial resolution and the wave attenuation for selected sediments found in the Baltic area.
MARINE GIS FOR MONITORING OF ENVIRONMENTAL THREATS USING UNDERWATER ACOUSTIC, REAL-TIME AND SATELLITE SENSORS

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Abstract: The main goal of the project was to provide marine GIS for monitoring environmental threats. Sea environment in open and coastal zone needs multi-sensor observation and monitoring of various kinds of threats. Early detection, identification and preparation of appropriate response strategies may prevent the natural environment from devastation. Several approaches and techniques of measurements are available in marine environment monitoring. These consist of direct sampling, airborne and satellite imagery, hydrological measurements using CTD probes, remote sensing with the use of electromagnetic waves, acoustic methods based on the data acquired by multi-beam and side-scan sonars and single-beam echosounders. Recent development in the information technology provides the means and possibilities for much faster and more efficient access to survey data, allowing remote, nearly real-time management, processing and visualisation. The presented system integrates data from all of the aforementioned sources as well as others, like live radar feed and oil spill spread simulation results. The data from the investigated marine region is presented in the form of multiple, time-varying layers, rendered in up to three dimensions. The system allows authenticated end users to remotely view these layers in a geographic context, while also providing interactive features like oil spill spread animation and tools for layer query.

Keywords: environment monitoring, GIS, underwater acoustic sensors, pollution modelling
1. INTRODUCTION

An efficient method for monitoring, prediction and visualization of various marine environment processes has been a subject of great importance for many years. This is the reason for development of various research techniques using different approaches and equipment. These techniques include [1]:

- acoustic methods based on the data acquired by multi-beam and side-scan sonars, (example on Fig. 1), as well as single-beam echosounders,
- direct sampling,
- hydrological measurements using CTD probes,
- satellite and airborne imagery.

![Sample multi-beam and side-scan sonar data.](image)

The process of acquisition, processing, integration and visualization of various kinds of data constitutes an important problem in the context of numerous applications related to aquatic ecosystems management. The paper describes a marine GIS system capable of integrating the data from various sources. These include real-time sensors like radar or satellite data receiver, where imagery, after necessary pre-processing, is accessible in the system without relevant delays and can be presented on the map. Apart from real-time sensors, the system consists of the data from dynamic sources such as results from numerical simulations, e.g. prediction of oil spill behaviour, as well as other data types such as bathymetry, background maps, underwater acoustic sensors’ data and others.

2. ARCHITECTURE OF THE SYSTEM

Development of marine GIS, at the Department of Geoinformatics, began in 2005, and the early effects were presented on the IEEE Conference in Brest. The prototype version was based on ArcIMS Server, which was an appropriate solution for providing static data layers. The architecture of the system in its prototype form is presented on the Fig. 2.

While the requirements enlarged and the advanced visualisation methods, such as imaging of animated layers, were needed, the architecture of the GIS was adapted to new demands. The current version of the system architecture is shown on Fig. 3.
The kernel of the current version of the system is based on open-source solutions, namely GeoServer, used as a map server, and OpenLayers Javascript library, used to develop the client side of web application. The web application in the current version of the system provides various 2D visualisation methods, including:

- imaging of numerous, overlaid data layers,
- presenting data from real-time sensors,
- visualisation of animated layers,
- markers mechanism for imaging of advanced data.

However, in order to use some advanced 3D visualisation methods and because of browser's technical constraints, the separate, stand-alone application was developed. Using the data acquired from the map server, it provides complex mechanism of 3D imaging of custom spatial objects ranging from MBS bathymetry, side-scan and single-beam echosounder data results to 3D modelling of oil spills or pelagic fish visualisation.
module and distributed users of stand-alone applications. GIS Integration Framework was described more precisely in [3].

3. POLLUTION MODELLING

One of the central demands for system being meant for monitoring marine environment, was to implement the ability of pollution behaviour modelling, in particular oil spill spread prediction. In bilateral cooperation of Department of Geoinformatics with Hellenic Centre for Marine Research, the advanced pollution monitoring and forecasting module was developed [3]. It consists of three main modules:

- POSEIDON Oil Spill Model, able to simulate various oil spill processes, like evaporation, emulsification, beaching and sedimentation,
- forcing module, providing data for wind, waves, currents and diffusitives,
- oil-spill weathering and drift module, simulating the dispersion of oil droplets and their chemical transformations.

When supplied with complete data about the initial oil spill, the module is capable of computing complex prediction of oil spill behaviour. The output contains:

- location and depth of each particle in the sea,
- evaporated volume of the initial oil,
- emulsificated volume,
- volume of the oil that reaches the beach,
- volume of the oil that reaches the sea floor.

Visualization of the oil spill behaviour, based on the model, can be presented in the web based GIS (Fig. 4), as well as in the stand alone application.

Fig. 4: Simulation of oil spill behaviour visualised in the system.
4. EXAMPLES OF SYSTEM USAGE

The system has been tested very intensively recently. Its main task is to ensure and simplify the process of sea environment monitoring. With the usage of basic visualisation capabilities of web-based application and advanced imaging methods provided by stand-alone application, the users can observe multiple data layers, including data layers collected from real-time sensors.

![Fig.5. Baltic sea monitoring using real-time and acoustic sensors data.](image1)

The users of the system are able to monitor possible environmental threats. That includes, for example, real-time marine traffic monitoring with the usage of AIS and radar, or wrecks observation, based mainly on underwater acoustic sensors data (Fig. 5). The system is also capable of presenting floating objects visualisation, utilizing both web-based and stand-alone application (Fig. 6).

Important part of the system is the nearly real-time satellite imagery module which consists of data from Meteosat Second Generation (MSG) satellite as well as from MODIS sensors. MSG’s data is useful for creating short-scale weather forecasts and environmental pollution maps. MODIS sensors, since their higher spatial and spectral resolution, were also verified to be sufficiently accurate to monitor oil spill behaviour. Shortly the system will be upgraded with third type of near real-time satellite imagery from NOAA and MetOp-A satellites. The data will be acquired with the 1.5m HRPT-MetOp satellite ground station which is now being installed and is planed to be fully operational from July 2009.

![Fig.6. Pelagic fish schools imaging.](image2)
Researchers from the Gdansk University of Technology Department of Geoinformatics along with researchers from the Polish Academy of Sciences have made several acoustic surveys on the r/v Oceania. During those measurements, test data from different underwater acoustic sensors were collected and added remotely to the system. The possibility of immediate cooperation with data analysts on the shore was checked.

The stand-alone part of the system constitutes a powerful tool for complex and rich imaging. It can be used to present data from multi-beam echosounder, single-beam echosounder and side-scan sonar simultaneously, in the single view, like shown on Fig. 7.

The markers mechanism has been used intensively in order to add additional data to the system, for example visualisation of the ebullition of methane in the Baltic Sea. That mechanism can be used for further upgrade of system functionality.

5. SUMMARY

The paper describes a complex web-based geographic information system adjusted for monitoring of marine environment along with data management module for data integration and easy cooperation of different groups of researchers. Especially for advanced visualisation purposes, the web-based GIS is supported with stand-alone application. The whole system is particularly useful for integrating various data, including data from underwater acoustic sensors, data from real-time sensors, satellite images, nautical charts and other types of background data. It also provides numerous and complex visualisation methods, constituting useful tool for marine environment monitoring.

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Structured Session 17

**Bubble Acoustics**

Organizers: Tim Leighton & Richard Lee Culver
The Acoustic Excitation of Newly-Formed Bubbles

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Abstract: Gas bubbles in water behave acoustically like lightly-damped oscillators with a natural frequency determined by their radius. When pinched off from another body of gas, their formation is usually accompanied by a pulse of sound at the bubble's natural frequency. Thus air bubbles formed beneath waterfalls, raindrops and breaking waves all radiate sound. One of the mechanisms driving the bubbles into acoustic oscillation is the rapid retraction of the neck of air formed immediately prior to formation. A model for the neck retraction, and its role in driving sound production by bubbles released from a nozzle and fragmenting in fluid shear, will be presented with experiments and theory. The potential to use the model to remotely monitor bubbles formed by breaking waves and released by marine methane seeps will be discussed. [Work supported by the Ocean Acoustics Division of the U.S. Office of Naval Research and the U.S. National Science Foundation].
MODELLING ANALYSIS OF ECHO SIGNATURE AND TARGET STRENGTH OF A REALISTICALLY MODELLED SHIP WAKE FOR A GENERIC FORWARD LOOKING ACTIVE SONAR

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Abstract: The acoustic modelling in TNO’s ALMOST (=Acoustic Loss Model for Operational Studies and Tasks) uses a bubble migration model as realistic input for wake modelling. The modelled bubble cloud represents the actual ship wake. Ship hull, propeller and bow wave are the main generators of bubbles in the wake. The bubble volume originates from the wake velocity field, combined with properties and physical processes of air bubbles in water. The bubble volume is assumed to be insonified by a generic active forward looking sonar. The resulting back-scattered sound is modelled with the recently developed ALMOST-REATES module (=Range Estimator for Active sonar and Target Echo Strength). It takes into account bubble scattering spectra and positions, and other important factors for the echo signal, like propagation loss from sonar transmitter and back to sonar receiver, beam forming, and signal processing. The new method is a further development of the ALMOST-REACT module (=Range Estimator for ACTive sonar): The active sonar equation is programmed here based on echo arrival times, computing sonar performance for point targets. The new method uses specified bubble positions, provided by the ship wake bubble migration model, to compute first the Impulse Response function. With the given spectrum of the active sonar pulse, the echo time series is modelled by simulating active sonar processing. Detection performance and an effective Target Strength for the scenario are deduced from this echo time series. The actual sonar processor is simulated, usually equipped with matched filter or straightforward energy detector. To run the model on a standard PC, the number of bubble positions is limited, while also the run time must be acceptable. The consequences of this limitation are reduced by applying some statistics in the modelling. Some results of echo structures and sonar performance are shown in a realistic active sonar scenario.
Keywords: ship wake modelling, acoustic sonar performance, acoustic signature, Target Strength

1. INTRODUCTION

Recently wake reflection modelling was started as a further development of the already existing ALMOST model (=Acoustic Loss Model for Operational Studies and Tasks) [1]. For a pulse sent by active sonar, the reflection from the bubble clouds inside the ship wake is modelled, based on bubble positions and size distribution. The new computation module is called REATES (=Range Estimator for Active sonar and Target Echo Strength). It computes the expected echo time series in a coherent way which allows further sonar processing like matched filtering. The input wake geometry for the acoustic modelling, is generated by a wake model implemented at TNO [2], [3], [4], supplying an array of bubble positions and sizes as a function of range behind the ship. This wake model basically computes the velocity field in the wake, and combines this with the dynamics for the individual bubble where upward force, pressure and drag play a role. Originally the wake model was only based on air entrapping sources near each of the propellers and the ship hull. Recently air sources due to the bow waves left and right of the ship have been implemented, thus yielding a more realistic wake geometry compared with measurements [5].

In the REATES time series modelling, multipath propagation for a realistic sonar scenario, from ALMOST, is taken into account. Further a realistic active sonar is modelled using the sonar directivity patterns from the REACT module (=Range Estimator for ACTive sonar) for active sonar performance predictions. Here the active sonar equation is modelled for simple point targets of given Target Strength. The method uses travel times for the echoes via the various propagation paths, computed as eigen-rays [6]. Echo level and reverberation are modelled versus time, presented as active sonar range for practical reasons. The new simulator is a straightforward extension of this existing modelling, but here for targets described by a number of scatterers. Examples are reflecting targets like ships assuming a suitably dense representation by scattering pixels. A cloud of air bubbles also forms a reflecting target, for instance occurring in ship wakes. The bubbles show a resonance effect [6], [7], to be modelled including phase, in the new method.

In the next part the basics of the echo modelling will be described [8], as well as the air bubble resonance phenomenon. Then the wake geometry is taken into account, as well as the directivity of the active sonar. The wake aspect angle, which is the angle between wake axis and sonar beam, turns out to be an important operational parameter.

Further behind the ship, forming the older part of the wake, the mean bubble size is smaller than closer to the ship. Echo modelling results for a variety of sonar frequencies, computed for different ranges after the ship will illustrate this effect.

2. THEORY AND MODELLING
Inside the wake there is an air bubble cloud, which consists of air bubbles at a number of positions, but with different bubble radius at each position. In order to model this cloud as a reflecting target using active sonar, the cloud will be approximated with a number of sub divisions for this cloud. Each sub division in the cloud only consists of bubbles with a radius between two rather close limits. So in the computation for the sub division the radius distribution can be well approximated as constant. A sufficient number of such sub divisions are taken together to model the actual cloud.

Air bubbles show a resonance effect in their transfer function, as follows:

\[ H_{\text{bubble}}(\omega) = \frac{P_{\text{scattered}}}{P_{\text{incident}}} = \frac{R}{r} \frac{1}{\omega_{\text{res}}^2 / \omega^2 - 1 + i\delta} \]  
\[ \omega_{\text{res}} = \frac{1}{R} \sqrt{\frac{3\alpha P_0 g}{\rho} - \frac{2\sigma}{\rho R}} \]  
\[ \delta = \frac{\omega R}{c} + \frac{4\eta}{\omega \rho R^2} + H_{\text{thermal}} \]  

With:
- \( \delta = \) damping, 2\(^{\text{nd}}\)+3\(^{\text{rd}}\) term \( \approx 0.1 \) [ratio]
- \( R = \) bubble radius [m]
- \( \omega = \) radial frequency [Hz]
- \( \omega_{\text{res}} = \omega \) at resonance [Hz]
- \( r = \) range to bubble [m]
- \( c = \) sound speed [m.s\(^{-1}\)]
- \( \eta = \) shear viscosity of the water [kg.m\(^{-1}\).s\(^{-1}\)]
- \( \rho = \) density of the water [kg.m\(^{-3}\)]
- \( H_{\text{thermal}} = \) heat conductivity loss term [ratio]
- \( \alpha = \) polytropic constant (1 < \( \alpha < C_p/C_v = 1.4 \)) [ratio]
- \( \sigma = \) surface tension of bubble [N.m\(^{-1}\)]
- \( P_0 = \) hydrostatic pressure in bubble (= \( \rho g b D_b + P_0 \)) [N.m\(^{-2}\)]
- \( g = 9.81 \) [m.s\(^{-2}\)]
- \( b D_b = \) bubble depth below surface [m]
- \( P_0 = \) atmospheric pressure, say about 10\(^5\) [N.m\(^{-2}\)]

Apart from the radiation (1\(^{\text{st}}\)) term in (3), the other terms are difficult to evaluate.

First the Target Impulse Response function (TIR) is evaluated for supposed simple “white” delta scatterers at the bubble positions, but already including all different combinations of propagation paths to and from this cloud target. Moreover these propagation paths will each possess different grazing angles, causing specific shifts in arrival time. Also the directivity of the sonar beams for transmission and reception is included in TIR, which is subsequently transformed to the frequency domain in an efficient way applying an FFT. Combining this result with the bubble transfer function we have:

\[ s(t) = \int_{-\infty}^{\infty} \sum_{n=1}^{N_f} H_{\text{bubble}}(\omega) \cdot C_n \cdot e^{-i\omega(t-r_n/c)} \ d\omega \]  

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With:

\[
\omega \cdot r_n / c = \vec{R}_n \cdot \vec{k}_{1,\text{src}} + \vec{R}_n \cdot \vec{k}_{1,\text{rec}}
\]  \hspace{1cm} (5)

\(\vec{R}_n\) = position vector of pixel \(n\)

\(\vec{k}_{1,\text{src}}, \vec{k}_{1,\text{rec}}\) = wave vector for paths towards sonar transmitter respectively receiver.

\(C_n\) = Source Level minus propagation loss

Because of the resonance behaviour of air bubbles, only those bubbles are taken into account in the model which are more or less near resonance for the given sonar pulse. In this way an effective bubble cloud is made by selection from the original one, for the sonar pulse in question. So all other bubbles with radii outside the required bubble range interval are ignored further on. In the obtained bubble radius interval, a number of 31 sub divisions is chosen, each with its specific mean bubble radius. By further supposing the bubble positions in these sub divisions identical but slightly randomly shifted as a whole in order to avoid artefacts, the final frequency response is determined by summation over the responses from all subdivisions. The random shifts of the sub division clouds only result in an extra complex phase factor in this summation.

Multiplying this result with the transmitted pulse spectrum, the modelled echo signal, in the frequency domain is fit for any kind of processing, like for instance matched filtering. The detector is modelled using the Hilbert transform to obtain the envelop signal. Also SAS processing can be applied to the time series using different sonar positions.

The sonar detection scenario is presented in Fig.1, where a sonar system is pinging from aside towards a ship wake. Because of the horizontal beam width of the sonar, the bubble object insonified by this beam will be quite large, dependent on “wake target” range and also “wake target” aspect angle. The number of bubbles as well as the TIR function would also become very large in such cases. Therefore only a part of the bubble cloud is selected for the TIR function. After modelling of this echo, time shifted copies are added including retardations dependent on the arrivals from the remaining insonified wake. Some statistical phase shifts and compensations for different propagation losses are added before coherent summation for the response from the entire insonified part of the wake.

![Fig.1 Sonar detection scenario with wake](image)
In the following the above scenario is modelled in REATES, where the ship wake is modelled by the wake model [2] computing the velocity field behind the ship, combined with the bubble dynamics, [3], [4]. This wake modelling has been extended with wake originating from the bow waves left and right of the ship (not shown in Fig.1), resulting in a horizontally much broader wake structure, as observed in measurements [5].

3. PARAMETRIC STUDY USING THE WAKE ECHO MODEL

Inside the wake, the volume contains air bubbles with a bubble radius distribution which is dependent on the range after the ship, further called “wake range”. Cross sections of the modelled wake at wake ranges 100 and 400 m are shown in Fig.2. In the left and the right areas in these figures, the wake from the bow waves left and right are shown, with the central part originating from the two propellers and the hull. At 400 m smaller bubble radii are shown than at 100 m.

\[ a) \quad b) \]

Fig.2: Cross section of wake a) at 100 m wake range (behind ship) b) at 400 m.

Running REATES for both above mentioned wake ranges, at some different frequencies, for the scenario of Fig.1, we get the results in Fig.3 to Fig.5. The horizontal axis represents active sonar range which is virtually the time scale just like the sonar display. The vertical axis shows the received level in dB re 1 \( \mu \)Pa (taking 0 dB gain for the matched filter). The black curves are the envelop echo signal versus time, after matched filtering. The cyan curves are the background level, modelled as realistically processed envelop signals, applying the echo modelling method described above, but here for the rough bottom as a target. This bottom roughness is taken from literature [9]. A maximum filter output is shown in the plots, for signal as well as for background, in order to better indicate detection probability of the “wake target”.

Particularly the echo for 10 kHz is much lower at 400 m than for 100 m. An explanation here is that the resonating bubbles are relatively large here, while larger bubbles will vanish faster than smaller ones. So there will be considerably less bubbles of large size at 400 m wake range. This effect is also checked in the output of the wake model, being the input for REATES.
Fig. 3: modelled echo structure of wake at 10 kHz a) at wake range 100 m b) at 400 m; in red a 0 dB point target.

Fig. 4: modelled echo structure of wake at 30 kHz a) at wake range 100 m b) at 400 m; in red a 0 dB point target.

In Fig. 4a, also the response for a point target with Target Strength 0 dB (referred to 1 m) is plotted. A maximum filter, which is often applied for detection purposes, shows 93 dB for the point target of 0 dB Target Strength, and 85 dB for the wake echo. The dB’s in the above received levels are relative to 1 μPa, assuming that the matched filter gain is set 0 dB. So at this specific target range of 1000 m and aspect angle 120 degree, the echo indicates an effective Target Strength of -8 dB (=85-93) at this wake range of 400 m and 30000 Hz frequency.
Fig. 5: modelled echo structure of wake at 60 kHz a) at wake range 100 m b) at 400; in red a 0 dB point target.

In Table 1 the various TS values for the above examples are shown.

<table>
<thead>
<tr>
<th>Wake range</th>
<th>frequency</th>
<th>10k Hz</th>
<th>30 kHz</th>
<th>60 kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td>100 m</td>
<td>-1</td>
<td>-6</td>
<td>-14</td>
<td></td>
</tr>
<tr>
<td>400 m</td>
<td>-22</td>
<td>-8</td>
<td>-13</td>
<td></td>
</tr>
</tbody>
</table>

Table 1 Target Strength of wake echo (dB re 1 m) for some special cases.

4. CONCLUSIONS

A new wake echo modelling method has been developed. Its input is a realistic wake consisting of air bubbles of different sizes, generated by a separate model. The model generates realistic wake structures, in agreement with measurements. Besides air entrapped near the ships hull and propellers, also air entrapped near the bow waves left and right of the ship has now been implemented. The model computes a realistic target echo structure applying a fully coherent modelling method, yielding echo time series and envelope. The method, being a further development of the ALMOST/REACT modelling of propagation and active sonar performance, takes into account full sonar processing, beam directivity patterns, environmental propagation effects, as well as specific scenario geometries for the wake detection using a forward looking medium or high frequency active sonar. The model can be used for parametric studies. The wake echo level appears to vary considerably with parameters like the distance to the ship, the aspect angle and the centre frequency of the transmitted pulse. The echo level can be quantified using an effective Target Strength. The cases for which the effective TS values have been derived show medium to low TS values.

REFERENCES

INFERRING BUBBLE POPULATIONS IN INTERTIDAL SEDIMENTS FROM ATTENUATION AND SCATTERING MEASUREMENTS

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Abstract: The presence of free gas can dramatically alter the acoustic properties of marine sediments. The effect of different shapes and sizes of gas pockets is of particular interest. Results from one acoustic transmission and two-frequency acoustic scattering experiments at intertidal gassy mud sites on the south coast of England provide evidence for the presence of both spherical and non-spherical gas voids. The characteristics of the bubble population can be estimated from new models of nonlinear bubble dynamics.

Keywords: gassy sediments, intertidal mud, nonlinear scattering, gas bubble dynamics

1. INTRODUCTION

The presence of seafloor gas-bearing (gassy) sediments can impact on the activities of the offshore industry and can influence biologic cycles of marine life. In gassy sediments, both the shape and size of the gas voids are important because their influence on the sediment’s bulk strength. Frequently spherical bubbles are used to model gas voids although there is evidence that gas voids in muds can form cracks or oblate spheroids \cite{1}. In this paper, acoustic transmission and backscattering measurements are employed with theory to devise methods for quantifying scattering from the shallow sub-seabed. Field measurements were carried out in the top metre of selected intertidal sediments on the south coast of England where methane production is of biogenic origin \cite{2}.

Methane production takes place below the sulphate reducing zone from anaerobic decomposition of organic matter (generally greater than one meter depth) \cite{3}. Aerobic
decomposition is the major reaction producing gas in the first top meter [4]. When entrained air is present this can lead to the presence of O₂, N₂ and CO₂.

Fig. 1: (a) Schematics of the experimental geometry (all dimensions are in cm). (b) Sound speed and attenuation as measured with the set up shown in (a) where the dotted line is a linear best fit to the data points connected with a solid line and the error bars indicate intrinsic errors at these points.

Fig. 2: (a) schematic of the experimental geometry for the scattering measurements (all dimensions are in cm). (b) Bubble size versus natural frequency where the solid line is predicted by equation 22 of ref. [14] and the dots are calculated from equation (1).

2. TRANSMISSION MEASUREMENTS

In situ transmission experiments were carried out at Calshot (50° 48’, 002° 2’ W) a site which had previously been characterized as poorly sorted coarse silt with porosity 62±5 % [5]. For the transmission experiments (Figure 1a) the acoustic source (which transmitted 1 ms tone-burst pulses from 26 to 100 kHz at 2 kHz steps) was buried below the sediment surface, its acoustic axis (along which 2 hydrophones were aligned) angled at 45° to the sediment surface. The transmission data were processed according to the technique described in ref. [5]. The results are shown in Figure 1b where the error bars indicate intrinsic errors [5]. According to these results the compressional sound speed (c_p) showed no dispersion (average value c_p=1340 ± 60 m/s) and the attenuation coefficient α [dB/m] follows a linear dependence with frequency f. Based on these observations, the attenuation data were fitted
to the typical expression \( a = K f^q \), where \( K \) is the constant of proportionality and \( q \) is the exponent of frequency \([6]\). The best-fit values for frequencies between 26 kHz to 100 kHz are 0.56 dB/m/kHz for \( K \) and 1 for \( q \). Comparing these results with previous data \([6]\) and \([7]\) for non-gassy muddy sediments suggests that gas was present. In theory, this is justified from the fact that \( c_p \) is lower than the values suggested by literature and the value of \( K \) is much greater (typically values of \( K \) for muddy sediments lay below 0.3 dB/m/kHz). However no clear resonant peaks where observed; such peaks would indicate the presence of gas in spheroidal form that is resonant in the frequency range tested.

3. SCATTERING EXPERIMENTS

Combination frequency scattering experiments were carried out within 1 metre of the transmission experiments location. The experimental set up was buried in the sediment (Figure 2a) and only volume scattering was taken into consideration. The high frequency transmitter (producing the “imaging frequency”) and receiver have a common focus point where their acoustic axes intersect each other at 90°, their axes being 45° either side of the axis of the pump transmitter (which is also the axis of symmetry of the set up). Beam pattern calculations were calculated in water and then a frequency-dependent correction was applied for the different sound speed in sediment. The “imaging frequency” \( f_1 \) was kept constant at 220 kHz and the “pump frequency” \( f_2 \) varied from 30 kHz to 100 kHz in increments of 2 kHz. The acoustic sources were calibrated in water such that at the intersection point of their acoustic axis, the pressure of the pump frequency was 30 kPa and the imaging frequency 35 kPa respectively (nominal zero-to peak amplitude) within the 3 dB limit of the common volume of these devices.

3.1. Theoretical considerations

The sediment is assumed to be the only source of nonlinearity. This assumption is based on previous work \([8]\) which suggests that the nonlinearity associated with bubble-free sediment is greater than twice that of the nonlinearity of bubble-free water (which is much smaller than the nonlinearity of water containing spherical pulsating bubbles) \([8]\)”. If two frequencies \( f_1 \) and \( f_2 \) are projected at a population of bubbles containing a wide distribution of bubble sizes, a spectra of various frequencies can be detected (\( f_1, f_2, 2f_1, 2f_2, |f_1 \pm f_2|, |f_1 \pm 2f_2| \) etc.). Commonly, interpretation of these scattered spectra relies on an assumed one-to-one mapping between a spectral component and a bubble size. This may be a valid assumption by suitable choice of \( f_1 \) and \( f_2 \). For example if \( f_1 \approx f_2 \) and there are no bubbles resonant at \( f_1 \) or \( f_2 \) then the only sources of spectral energy at difference frequency (\( f_1 - f_2 = f_{1-2} \)) are the bubbles resonant at that frequency, i.e. \( f_{1-2} \). One problem with this approach is finding suitable frequency ranges for sediments. Leighton et al \([9]\) developed a scheme whereby the contributions of all bubbles in the population to each spectral component are considered in the inversion that estimates the bubble population from the scattering. This is more rigorous than application of the above assumptions, but also is particularly important in gassy sediment, where the high attenuation makes it difficult to exploit a frequency which can be guaranteed to be much higher than the resonances of any bubbles present. In order to
interpret such spectra, a new bubble model was required. The bubble radius time response $R$ and natural frequency $f_0$ were estimated from the nonlinear bubble models build on earlier models for sediment $^{[10,11]}$ and biological tissue $^{[12]}$. The model incorporates shear effects from first principals, and can cope with amplitude-dependent effects, and two-frequency insonification, which cannot be captured by a linear model, for example of Anderson and Hampton $^{[13, 14]}$. In the small-amplitude linear limit the model of $^{[11]}$ predicts a pulsation resonance frequency predicts the linear resonance frequency:

$$f_0 = \left(2\pi R_0\right)^{-1} \sqrt{3\kappa p_{b0} + 4G_s} / \rho_s \tag{1}$$

of a bubble with equilibrium radius $R_0$ assuming adiabatic pulsations, where $\kappa (=1.3)$ is the polytropic index of the bubble gas (assuming CO$_2$), $p_{b0} (=104$ kPa) is the bubble ambient pressure for the set up shown in Figure 2a and the parameters $\rho_s$ and $G_s$ are the density and the shear modulus of the gas-free sediment having values $1640$ kgr m$^{-3}$ and $2.6$ MPa respectively. The thermal effects are of minor importance in sediment of the type discussed here. This is demonstrated in Figure 2b where equation (1) is compared with equation 22 of ref. $^{[14]}$. For these simulations the gas was assumed to be atmospheric air at $10 \degree C$ and the compressional wave speed (in the bubble host medium i.e. gas-free sediment) equal to $1430$ m/s. As expected there is good agreement of the two equations as $G_s$ is the dominant term.

The spatial distribution of any bubbles present is assumed to be random and hence the measured scattering is interpreted as incoherent and the concept of scattering cross section is invoked. The nonlinear differential extinction cross section of the individual bubbles ($\sigma_s$) was computed numerically from the nonlinear bubble model using the input parameters mentioned in the previous paragraph and definition of extinction cross section $^{[15]}$: $\sigma_s = R_0^2 |P_b|^2 / |P_s|^2$, where $|P_b|$ and $|P_s|$ are the amplitude pressure spectral components of the incident and scattered field respectively at the frequency of interest ($f_{i-2}$), where in (2) the scattered field is evaluated at the bubble wall ($r = R_0$):

$$P_b = \rho_s \left(2R \dot{R}^2 + R^2 \ddot{R}\right) / R = R_0$$, where dots represent time derivatives. \tag{2}

The received pressure spectral component at $f_{i-2}$ from a number $N$ of identical bubbles depends on the radial distance, $R_m$, of the centre of the receiver face to the centre of the sensing volume at difference frequency:

$$|P_r|^2 = \int V N \sigma_{rv} |P_s|^2 R_m^2 dV \left(\Omega_v\right)$$, where $\sigma_{rv}$ is the receiving cross section. \tag{3}

(i.e. $\sigma_{rv}$ is the $\sigma_s$ corresponding to the sensing solid angle), an expression which can easily be extended to a continuum of bubble sizes.
Fig. 3. Pressure spectral component $|P|$ in dB (with a common dB reference) as measured by the receiver. The imaging frequency is kept constant at 220 kHz. The figures show measurements with pump frequency at (a) 32 kHz, (b) 38 kHz, (c) 66 kHz and (d) 82 kHz.

4. RESULTS & DISCUSSION

Figure 3 shows example spectra obtained when two frequencies are projected into the sediment. Generally such data are ambiguous: for example, scattering at the difference frequency $f_{1-2}$ is studied, and this can arise from bubbles being resonant at the primary frequencies ($f_1$ and $f_2$) and those resonant at $f_{1-2}$. However for the site in question the transmission results showed no resonances at the frequency range of 26-100 kHz. Therefore it is proposed that occlusions corresponding to these bubble sizes i.e. from 130 to 500 micron do not exist in spherical form. The consequence of this working hypothesis would be that the scattering at $f_{1-2}$ is generated from bubbles resonant at $f_{1-2}$ i.e. bubbles smaller than 130 micron. As shown in Figures 3a-c resonances at the difference frequency ($f_{1-2}=188$ kHz, 182 kHz and 154 kHz respectively) are clearly observed which correspond to resonant sizes from 60 to 70 microns. However this is not the case for the Figure 3d ($f_{1-2}=132$ kHz) which corresponds to approximately 100 microns (see Figure 2b). In conclusion these preliminary results reveal the existence of spherical voids in muddy sediments with radius smaller than 70 microns (larger gas pockets probably forming aspherical gas pockets e.g. cracks [16]). In later work these preliminary experimental data will be inverted to estimate bubble size distributions.

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REFERENCES


THE EFFECT OF WIND-GENERATED BUBBLES ON SEA-SURFACE BACKSCATTER

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Abstract: Predictions of sea-surface back-scattering strength are needed for sonar performance modelling. Such predictions are hampered by two problems. First, measurements of surface back-scattering are not available at small grazing angles. These are of special interest to low-frequency active sonar since they mainly contribute to long range propagation. Second, existing theoretical models based on a bubble-free interface underestimate the surface back-scattering strength at larger grazing angles. We investigate whether wind-generated bubbles can explain this deficit. For this purpose, we develop a theoretical description that includes the effect of refraction and scattering of sound by wind-generated bubbles. The comparison of the theoretical predictions to Critical Sea Test measurements show that a good fit is obtained between theoretical predictions and measurements for wind speeds up to 10 m/s. For larger wind speeds, the surface back-scattering strength critically depends on the population density of large (radius > 1 mm) bubbles. This provides an opportunity to estimate the number of large bubbles. We observe a change in the spectral slope in the bubble population model for large bubbles that is in agreement with high-speed camera observations in breaking waves and with the Hinze scale.

Keywords: sonar, scattering, bubbles, sea-surface, LFAS
1. INTRODUCTION

Scattering of sound at the sea surface can have a significant impact on the performance of low-frequency active sonar (LFAS). For this application, the scattering at small grazing angles (i.e. below 5 degrees) is of special interest since these are most relevant for long range propagation, e.g. in a surface duct. It remains nevertheless a challenge to assess the consequences on the LFAS performance since rough-surface back-scattering measurements are not available at these low grazing angles [1],[2].

Empirical or semi-empirical models, such as proposed by Ogden and Erskine [3] are generally used for the extrapolation to small grazing angles. The Ogden-Erskine empirical formula is tuned on a comprehensive set of low-frequency measurements (in the range between 70 and 940 Hz) of surface back-scattering strength available from the Critical Sea Test (CST) experiments [3],[4]. These cover grazing angles in the range 5 to 30 degrees and wind speed values up to 18 m/s. The measured reverberation levels exceed those expected from rough-surface scattering alone, especially for large wind speeds [1],[4],[5]. The Ogden-Erskine empirical formula combines the rough-surface scattering term with the Chapman-Harris empirical model [6] which predicts the surface back-scattering strength at high frequencies and wind speeds.

For small grazing angles, the existing empirical predictions are not constrained by measurements. In this region, the Ogden-Erskine model uses a theoretical extrapolation based on rough-surface scattering (perturbation theory). Since this mechanism is not able to explain the measurements at large wind speeds, we question whether the extrapolation provides reliable results in (say) sea state 4. This is especially relevant in winter conditions; high wind speeds are then commonly observed and cooling at the sea-surface results in an upward-refracting sound speed profile.

In order to improve the reliability of sonar performance prediction, a physical model that captures all essential processes governing the sea-surface back-scattering is therefore needed. Several authors proposed a surface scattering model that contains, in addition to the sea-surface contribution, a volume scattering term. At high frequencies, resonant scattering from individual bubbles is known to contribute significantly to the back-scattering strength [4],[7],[8]. At lower frequencies (below 5 kHz), scattering from bubble clouds is proposed as a mechanism [9]-[11].

In this paper, we hypothesize that scattering from individual bubbles may contribute significantly to the total surface back-scattering strength, at or around 1 kHz. We investigate to what extent this mechanism, combined with rough-surface scattering, may suffice to explain the total of CST measurements.
2. THE EFFECT OF BUBBLES

We study both the reflection and scattering of sound by the rough sea surface and the absorption and scattering of sound by entrained gas bubbles. The gas bubbles have two effects on the back-scattering:

- A modification of the sea-surface scattering contribution: As a result of the bubbles in the near-surface layer, the bulk modulus and therefore the sound speed decrease, leading to an increasing ray grazing angle $\theta$ at the sea-surface (see Figure 1). According to perturbation theory, the surface contribution is proportional to $\theta^4$ [3].

- Introduction of a volume scattering contribution due to scattering of sound at the bubbles. We consider both the direct scattering contribution and the interaction with the sea-surface. As shown in Figure 2, this results in four scattering contributions for each bubble [12].

![Fig. 1. The effect of a near-surface bubble layer on the grazing angle](image)

![Fig. 2. The four different volume scattering contributions for a single bubble.](image)
In our physical model, the near-surface bubble layer is parameterized using the empirical “Hall-Novarini” (HN) bubble population model. This model describes the distribution of air bubbles as a function of wind speed, depth, and radius [1],[13],[14] quantified using the bubble population spectral density (PSD). It gives the number of bubbles per unit volume of ocean that have radii within a unit increment in radius. The PSD varies with depth $z$, bubble radius $a$, and wind speed (at a height of 10 m) $v_{10}$. The HN model assumes that no bubbles exist with a radius $a$ less than $a_{\text{min}} = 10\mu\text{m}$, or greater than $a_{\text{max}} = 1000\mu\text{m}$. Figure 3 shows the PSD as a function of radius for wind speed $v_{10} = 14\text{ m/s}$ at three different depths. It shows that the bubble PSD rapidly decreases with depth. Furthermore, it illustrates the discontinuous nature of the HN model through a bubble radius of 1 mm.

Figure 4 shows the theoretical back-scattering predictions based on the HN model. It shows that for wind speeds larger than 10 m/s, the theoretical model underestimates the backscatter, whereas there is a good agreement between the model predictions and the observations for lower wind speeds.

The target strength of an individual bubble (Figure 5) suggests that only a small number of large bubbles contribute significantly to the backscattering strength. The scattering strength of individual bubbles increases quadratically with radius and at 940 Hz bubbles may be excited close to their resonance frequency. This motivated us to investigate whether large bubbles are able to explain the underestimation of the sea-surface back-scattering by the theoretical model. For this purpose, we extended the HN model that truncates the bubble PSD at a radius of 1 mm to include larger bubbles, i.e. bubbles with a radius larger than 1 mm (see Figure 3). The spectral slope of the bubble PSD for these large bubbles is determined using a least-squares fit to the observations.

The extrapolated HN model at $v_{10} = 14\text{ m/s}$ is shown in Figure 3, and the corresponding fit to the observations in Figure 6. Based on the improvement in the match between the theoretical predictions and the observations, we conclude that a small number of large bubbles significantly contribute to the sea-surface back-scattering strength.

![Fig. 3. Hall-Novarini bubble population spectral densities for a wind speed ($v_{10}$) of 14 m/s (solid). The dashed curves show the extrapolation of the Hall-Novarini model for large bubbles. The curves correspond to 0.7, 1.8 and 4.0 m depth, respectively.](image-url)
Fig. 4. Theoretical (curves) total backscattering predictions at 940 Hz based on the HN model compared to surface back-scattering measurements (symbols) obtained during the Critical Sea Test Experiments [4]. The legend indicates the wind speed values ($v_{10}$).

Comparing the theoretical predictions to the Ogden-Erskine empirical curves reveals that there are significant differences at low grazing angles for all wind speeds. As a consequence, including the effect of gas-entrained bubbles in the near-surface layer is expected to be of importance to sonar performance prediction.

Fig. 5. Bubble target strength as a function of radius at 940 Hz. The red line indicates the separation between the bubbles that are included in the HN model and the extended HN model.
A final observation is that the discontinuity in the spectral slope of the bubble PSD (Figure 3) is in agreement with observations obtained with high-speed cameras in breaking waves. This phenomenon is referred to as the Hinze scale [15][16].

Fig. 6. Theoretical (solid curves) total backscattering predictions at 940 Hz based on the extended HN (including bubbles larger than 1 mm) model compared to surface backscattering measurements (symbols) obtained during the Critical Sea Test Experiments [4]. The legend indicates the wind speed values ($v_{10}$). The dashed lines are the empirical Ogden-Erskine predictions for the same conditions as the solid lines.

3. SUMMARY AND CONCLUSIONS

A theoretical model is formulated that is able to explain sea-surface back-scattering measurements at 940 Hz obtained during the Critical Sea Test Experiments.

- This model reveals that a small number of large bubbles, i.e. bubbles with a radius larger than 1 mm, significantly contribute to the sea-surface back-scattering strength at 940 Hz.
- For all wind speeds, we observe large differences with the empirical Ogden-Erskine curves at low grazing angles ($> 10$ dB), while both the theoretical predictions and the empirical Ogden-Erskine curves have a good fit to the observations that are available for larger grazing angles. These differences at small grazing angles are important for sonar performance prediction.
- The results obtained with the theoretical model are sensitive to the prescribed bubble PSD. As a result, the theoretical model can provide indirect observations for the bubble PSD by matching the predictions to observations. The shape of the bubble PSD obtained in our theoretical model is in agreement with high-speed camera observations, i.e. the Hinze scale.
4. ACKNOWLEDGEMENTS

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REFERENCES


ACOUSTIC AND OPTICAL MEASUREMENT OF BUBBLE POPULATIONS IN THE ATLANTIC OCEAN AND THE MODELING OF GAS TRANSFER THROUGH THESE BUBBLE CLOUDS

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Abstract: Bubbles, formed by breaking waves, play an important role in the transfer of gases between the Earth’s oceans and atmosphere and have been shown to increase the flux of gases during periods of heightened sea state. Having been formed, these bubble clouds evolve through the effects of buoyancy, gas exsolution and dissolution, and the fragmentation and coalescence of bubbles. A number of experimenters have successfully measured sub-surface bubble clouds using a variety of acoustic and optical techniques, although data over a wider range of bubble radii are required for fuller comparison with models of how these clouds evolve and contribute to air/sea transfers of mass, momentum and energy. This paper presents data measured in the Atlantic Ocean, using an 11 metre spar buoy, between 16th June and 18th July 2007. An acoustic system measured the additional attenuation due to bubbles to infer the bubble size distribution whilst an optical system exploited the change in refraction caused by a bubble at the tip of an optical fibre probe. The measured bubble populations are then used as an input to a gas transfer model and the resulting fluxes and their significance will be calculated.

Keywords: Bubbles, atmosphere/ocean gas flux, underwater acoustics, ocean waves
1. INTRODUCTION

With the current high profile of the global climate systems, greater understanding of the factors affecting these systems is required. In response to this, the Natural Environment Research Council set up a programme called SOLAS, Surface-Ocean Lower-Atmosphere Study. The research presented here was carried out as part of this programme.

It has been shown that in areas of increased breaking wave activity, the flux of atmospheric gases into the ocean is increased [1]. This increase has now been shown to be caused by bubbles dissolving atmospheric gas into the ocean [2] and produce a slight supersaturation of these gases in the upper ocean [3].

Bubble populations under breaking waves can number millions per cubic metre, and contain bubbles ranging in radius from microns to centimetres. Of the available techniques for measuring such populations, acoustic methods are the most applicable [4, 5]. However any such acoustic technique contains ambiguities, and so it is best to check the results against an independent measurement (preferably a non-acoustic one) [6, 7].

This paper presents results from experiments carried out in June/July 2007, using acoustic and optical systems mounted on a free-floating, autonomous spar buoy. These data will then be applied to the issue of air/sea gas transfer.

2. EQUIPMENT

The bubble sensing equipment designed by the authors was attached to an 11 metre spar buoy (Figure 1) that was built by the authors’ collaborators at the National Oceanography Centre, Southampton, UK (NOC). To this buoy the NOC collaborators also attached wave wires and downward-looking cameras to image the sea surface. The buoy was designed to be free floating and fully autonomous. When deployed in the ocean, approximately 80% of the buoy is underwater, with just 2 metres protruding above the surface. The buoy had an Argos beacon situated in the dome, which sent position data to the ship allowing the buoy to be recovered at the end of a deployment.

2.1. Acoustic system

The acoustic system for bubble counting measures the increased attenuation caused by the presence of bubbles. The acoustic setup consisted of transmit transducers, power amplifiers, an array of hydrophones and an onboard computer to control the equipment. The computer (a MagnumX 1000 low-wattage single-board computer) had a National Instruments 6110 data acquisition card installed to allow high frequency sampling of the incoming data. The power amplifiers and matching circuits used to drive the transmit transducers were custom designed and built by Paul Doust, at the time working for Blacknor Technology. The amplifiers were designed to run off batteries and yet produce a high sound pressure level in the water (approximately 190 dB re 1 µPa). For each measurement, three transducers repeatedly emitted a train of 14 pulses ranging in centre frequency from 3 kHz to 197 kHz. This allowed bubbles with radii ranging from 17 to 1107 microns to be measured. The pulses were then received by an array of three D/140 hydrophones, positioned between approximately 0.8 and 2.6 metres below the sea surface. The received waveforms were then digitised by the data acquisition card and stored on a hard drive for analysis once the buoy had been recovered.
2.2. Optical system

The optical system used three fibre-optic tips mounted along the buoy. As bubbles pass over these tips, a change in light intensity is measured and bubble populations can be inferred [8, 9]. The optical system therefore monitors the scattering of light transmitted down an optical fibre: the backscattered scattering changes depending on whether there is water or gas at the end of the tip. As such, the passage of a bubble over the fibre tip generates a transient, one per bubble, and the magnitude, duration, and rise-times of each transient can be used to estimate the bubble size.

3. EXPERIMENTS

As part of the SOLAS programme, the buoy was taken on two sea trials on RRS Discovery in the Atlantic Ocean. The first cruise, D313 in November/December 2006, was primarily a proof-of-concept sea-trial for the buoy [10], which enabled the authors to optimise the equipment and prepare for the second sea-trial. The second cruise, D320, took place in June/July 2007. The area of operation was in the North Atlantic, 400 miles west of Portugal.

The buoy was deployed 4 times during D320, with each deployment being 3 or 4 days in duration. During the first deployment, useful data was acquired by the optical system but the bubbles clouds did not penetrate deep enough for the hydrophones to record any meaningful attenuation. The acoustic system did however acquire useful data on the second deployment,
whilst a leak in the optical amplifier housing caused a failure of the optical system. The third and fourth deployments did not return any useful data owing to damage to the equipment. Therefore optical data from the first deployment and acoustic data from the second deployment are presented in the following section.

Meteorological conditions for the deployments can be seen in table 1.

<table>
<thead>
<tr>
<th>Deployment</th>
<th>Measurement</th>
<th>Mean wave height [m]</th>
<th>Mean wind speed [m/s]</th>
<th>Mean water temperature [°C]</th>
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<tr>
<td>1</td>
<td>Optical</td>
<td>1.9</td>
<td>7</td>
<td>17</td>
</tr>
<tr>
<td>2</td>
<td>Acoustic</td>
<td>2.7</td>
<td>13</td>
<td>17</td>
</tr>
</tbody>
</table>

Table 1. Meteorological conditions for the two deployments in which data were measured.

4. RESULTS

Bubble size distributions were measured using both acoustic and optical techniques.

![Figure 2. Plot of the acoustically measured bubble size distribution (black circles with dashed line). This population estimate represents the spatially averaged bubble density between two hydrophones, at mean depths of 0.8 m and 2.54 m. Historical measurements are also shown. The diamonds are those of Johnson and Cooke [11], the triangles are those of Breitz and Medwin [12] and the crosses are those of Phelps and Leighton [13].](image)

The acoustic techniques give the population for bubbles ranging from 17 to 1107 microns in radius. The optical techniques could potentially increase this range to larger bubble radii with a working range of approximately 167 to 3500 microns in radius, though the methods have not yet been reliably verified and therefore the results are not shown here. There is very good agreement between the acoustic dataset and the historic measurements. Coupled with the
optical measurements, the range of bubble radii measured would exceed any historic measurement. Although these measurements demonstrate the principal that this range is achievable, equipment failure meant that they were not taken under identical sea conditions.

In order to calculate the gas flux associated with these populations, a model of the sub-surface bubble cloud evolution can be used and this is described in more detail in the next section.

5. APPLICATION

Woolf and Thorpe [3] show how the traditional equation for air-sea gas transfer can be split into bubble mediated transfer and direct transfer, given by

\[ F = K_o (C_w - C_a) + K_b (C_w - C_a (1 + \delta)) \]  \[5.2\]

where \( F \) is the net transfer of gas across the sea surface, \( C_a \) is the concentration of the gas in air, \( C_w \) is the concentration of the gas in water, \( K_o \) is the direct contribution to the transfer velocity, \( K_b \) is the contribution of bubbles to the transfer velocity and \( \delta \) is the equilibrium fractional supersaturation of the gas.

The contribution of bubbles to the transfer velocity can be calculated using a model of the evolution of sub-surface bubble clouds and the associated gas transfer through these bubbles presented by Woolf & Thorpe [3]. The results produced by Woolf & Thorpe [3] were based on estimates of oceanic conditions. More accurate results can be obtained if the model is run with parameters taken from the meteorological conditions of D320 and adapted to produce a best fit with the measured bubble size distributions. This technique will be applied in future papers for the data of this paper, and will produce estimates of bubble mediated gas transfer based upon real data and would be an important assessment of the viability of the estimates made by Woolf & Thorpe [3]. This will also enable the variation in the bubble size distribution with depth to be determined and a comparison made of the optical and acoustic measurements.

6. CONCLUSIONS

Bubble measurements were made from a free-floating, fully autonomous spar buoy using both acoustic and optical techniques. These measurements were taken in open ocean, with a depth of approximately 4 km. A range of bubble radii potentially broader than ever before has been measured and a method has been outlined for applying these data to the calculation of bubble mediated gas transfer between the atmosphere and the ocean.

7. ACKNOWLEDGEMENTS

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REFERENCES

INTERPRETATION OF ACOUSTIC SCATTERING FROM SUBMERGED AQUATIC VEGETATION

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Abstract: The acoustic reflectivity of submerged aquatic vegetation has been known and exploited for algae detection. The acoustic impedance, which results in reflectivity, is thought to result primarily from the gas within the plants. Photosynthesis by seagrass substantially increases the quantity of oxygen in dissolved and bubble forms in the water column. Field experiments show that oxygen bubbles deposited on the leaf blades at the onset of photosynthesis affect the propagation of sound. To interpret the shape of the echo signals coming from bottom covered with vegetation, we model algae reflectivity by bubble population. Sound scattering by air bubbles near the sea surface has become a very important issue in a number of recent publications. We generalize this approach for near bottom bubble clouds. It was shown that the shape of echo signals is significantly affected by the presence of caustics caused by concave areas of the sea bottom.

Keywords: Bubble, algae, backscattering, caustics

1. INTRODUCTION

Acoustic methods have been used to map seagrass beds since the 1970s [1]. Air bubbles generated by the plants during photosynthesis dominate the acoustic behaviour. In situ investigations have confirmed the acoustic dependence on plant density and photosynthetic activity for backscatter and propagation in the water column [1, 2], but corresponding theoretical models have yet to appear thus, providing a motive for the current study.
Down-looking echosounders have long been the tool of choice for underwater habitat mapping, and type of seabed. To interpret the shape of the echo signals coming from bottom covered with vegetation, we model algae reflectivity by bubble population.

The problem of scattering from air bubbles in the vicinity of the air-sea interface has become a very important issue for near three decades. Clay and Medwin [3] first showed quantitatively that only a very low density of near-surface bubbles would be necessary to account for the levels observed in some surface backscattering measurements made at high frequency and at moderate and low grazing angles.

The backscattered pressure, from a single bubble located a distance $d$ below mean level of roughened sea-surface interface, and range $l$ from the transducer (with $d/l << 1$) can be expressed as the contribution of the four paths [4, 5]. The first one involves the direct scatter of radiation by the bubble. For the second and the third paths, Fig. 1 (b) and (c) the sound is scattered by the bubble and undergoes one reflection at the bottom. The last path, Fig. 1 (d), involves two surface reflections. Our approach to modelling backscattering from near bottom bubble clouds follows [5–7] where a model for acoustic scattering from a single bubble located close to an air-water interface was presented along with verifying experimental measurements. We generalize this approach for near bottom bubble clouds.

2. MODEL FOR BACKSCATTER FROM NEAR SEA-FLOOR BUBBLES

For a source of the strength $P_{ir}$, located at $r_s$, the far filed radiated wave has the form

$$P_{ir}(r, r_s, t) = \frac{D(\theta, \phi)P_{ir}}{4\pi |r - r_s|} e^{i(k|r-r_s| - \omega t)}.$$  \hspace{1cm} (1)
The variables \( k \) and \( \omega \) are the acoustic wave number and angular frequency, respectively. Note that the beam pattern \( D(\theta, \phi) \) has been expressed in terms of the amplitude, not the intensity, as is more common. The direct scatter of radiation by the bubble located at \( \mathbf{r} \) in response to the incident wave (1) can be generally represented at far field as

\[
P_a(\mathbf{r}, \mathbf{r}_s, t) = \frac{D(\theta_{in}, \phi_{in}) P_m}{4\pi |\mathbf{r} - \mathbf{r}_s|} e^{i[k(r-r_s) - i\omega t]} f_b \left( \frac{\mathbf{r} - \mathbf{r}_s}{|\mathbf{r} - \mathbf{r}_s|}, \frac{\mathbf{r}_s - \mathbf{r}}{|\mathbf{r}_s - \mathbf{r}|} \right) \frac{1}{|\mathbf{r}_s - \mathbf{r}|} e^{i[k(r-r_s)}. \tag{2}
\]

The direction of incidence on a bubble is described by the unit vector \( \mathbf{s} = \hat{\mathbf{r}} \) and the direction of propagation is \( \mathbf{s} = \hat{\mathbf{r}} \). \( f_b \) is the scattering amplitude. Large bubble radius and hundreds kilohertz frequency used in discussed experiments necessitate to use an angle dependent complex scattering amplitude. The first exponent is the pressure field exciting the bubble. The forward part of the path carries the spherical spreading factor \( 4\pi |\mathbf{r} - \mathbf{r}_s| \), while the corresponding factor for the backscattering is \( f_b / |\mathbf{r} - \mathbf{r}_s| \).

For the second, the third, and the fourth paths (see Fig. 1) one should account the reflection at the bottom. Traditionally [5–7], the backscatter pressure from a single bubble located near of a roughened air-water interface is expressed as

\[
P_s(\mathbf{r}, \mathbf{r}_s, t) = \frac{D^2(\theta_{in}, \phi_{in}) P_m}{4\pi |\mathbf{r} - \mathbf{r}_s|^2} e^{2ik\beta} \left[ f_b^{180} e^{2ik\alpha} - 2 f_b^\theta e^{2ik(\alpha+\beta) + i\phi} + f_b^{180} e^{2ik\alpha + 2i\phi} \right], \tag{3}
\]

where \( \alpha \) and \( \beta \) are the distances shown in Fig. 1, \( f_b^{180} \) and \( f_b^\theta \) represent the complex scattering amplitude of a bubble at scattering angles of 180 degrees and at 180 degrees \( \pm 2\theta \) respectively (Fig. 1). The influence of the roughened air-water interface is accounted by factors \( e^{\phi} \) and \( e^{2i\phi} \) representing the added phase shift, imparted by one and two reflections. One assumes the Kirchhoff approximation, for which the added phase shift equals \( 2k\sigma \sin \theta \), where \( \sigma \) is the surface displacement and \( \theta \) is the grazing angle. This approximation does not account the diffraction effects which are essential in calculating scattering from rough surface at distances compared with the radii of curvature of roughness. Since sea plants are located namely at these distances accounting for caustic structures and focusing regions is necessary.

In the Kirchhoff approximation, the Helmholtz integral for the pressure \( P_s(\mathbf{r}) \) scattered from the surface can be written as

\[
P_s(\mathbf{r}, \mathbf{r}_s, t) = \frac{1}{4\pi} \int_V \frac{\partial}{\partial n} \left[ P_m(\mathbf{r}, \mathbf{r}_s, t) e^{i[\mathbf{r}_s - \mathbf{r}]} \right] dS,
\]

where \( \partial / \partial n \) is the derivative with respect to the surface normal, \( dS \) is the surface element. The local Fresnel reflection coefficient \( V(\mathbf{r}, \mathbf{n}) \) is calculated for a flat surface oriented tangential to the rough surface at the integration point being evaluated. The normal derivative and surface element in Eq. (4) can be written as \( dS = dxdy/n_z, \partial / \partial n = (\mathbf{n} \cdot \nabla) \), \( \mathbf{n} = n_z \left( e_x - \xi_x e_x - \xi_y e_y \right), n_z = \left[ 1 + (\xi_x)^2 + (\xi_y)^2 \right]^{-1/2} \), where \( \mathbf{r} = xe_x + ye_y + \xi(x, y)e_z \); \( e_x, e_y, e_z \) are the unit orts and \( \xi_x, \xi_y \) are the partial derivatives of the surface height with respect to \( x \).
and y, respectively. Assuming that \( kR_x \gg 1, kR_y \gg 1 \) \((R_x = |r - r_x|, R_y = |r - r_y|)\) we can write the scattered pressure as

\[
P_s(r, r_x, t) = \left(\frac{ik}{4\pi}\right) e^{-i\omega t} \int \int \int VD \left[ \left( \frac{\partial}{\partial z} - \xi_x \frac{\partial}{\partial x} - \xi_y \frac{\partial}{\partial y} \right) \left( R_x + R_y \right) \right] \frac{e^{i(kR_x + R_y)}}{R_x R_y} dx dy dz,
\]

(5)

The integral expression for the scattering field can be evaluated using high frequency approximation. Applying stationary phase analysis to Eq. (5) we first seek the points on the surface where \( \frac{\partial}{\partial x}(R_x + R_y) = 0 \), \( \frac{\partial}{\partial y}(R_x + R_y) = 0 \). Next, the integral of Eq. (5) is written as a sum over all \( M \) stationary phase points, and each term in the sum comes from an expansion of the integral around a stationary phase point \( m \), i.e.

\[
P_s(r, r_x, t) = e^{-i\omega t} \sum_{m=1}^{M} D(m) e^{i(kR_x + R_y)} \left\{ \left[ \frac{1 - ((r_x - r_x)(r - r_y))}{R_x R_y} \right] \frac{z_x - \xi_x}{R_x} + \frac{z_y - \xi_y}{R_y} \right\]_{(m)},
\]

(6)

\[
\alpha = \frac{\partial^2 (R_x + R_y)}{\partial x^2}, \quad \beta = \frac{\partial^2 (R_x + R_y)}{\partial y^2}, \quad \gamma = \frac{\partial^2 (R_x + R_y)}{\partial x \partial y}, \quad \sigma = \begin{cases} 1, & \alpha \beta > \gamma^2, \alpha > 0 \\ -1, & \alpha \beta > \gamma^2, \alpha < 0 \\ -i, & \alpha \beta < \gamma^2 \end{cases}
\]

where the index \( (m) \) means evaluation at the point of the stationary phase \( m \). This equation is an adequate description and corresponds to the simplified approach used in (3) with slight modification accounting reflection at the sea-floor interface \( V(m) \). However the variable \( \alpha \beta - \gamma^2 \) can vanish and pressure for that term becomes infinite. The physical reason for the divergence is that for these geometries some part of the rough surface is focusing energy at the receiver. The focusing properties of pressure release corrugated surfaces [8, 9] are known and intensively studied, thus we are following this approach in interpretation of backscattering from aquatic vegetation.

3. FOCUSING REGIONS

Here, we will examine one of the simplest model [8] and use a two dimensional rough surface with topography in x direction being that of the one-dimensional curve. In the y direction there is no surface variation. Most seagrasses, including Zostera, grow on sandy beds. The characteristic (near one dimensional) sand ripples patterns seen in the oceanic coastal zone which are formed under shoaling waves. Thus, we shall model these ripples by the one dimensional model [8].

Evaluating individual terms in the sum (6) for the one-dimensional model we obtain [8]:

\[
P_s(r, r_x, t) = e^{-i\omega t} \sum_{m=1}^{M} \frac{V(m)D(m)e^{i(kR_x + R_y)}}{R_x} \left( \frac{R_x}{(R_x + R_y)} \right)^{1/2} \frac{1}{(R_x + R_y)} \left( R_x + R_y \right)^{1/2},
\]

(7)
where $R_{(m)}$ and $\rho_{(m)}$ are the principal radii of curvature of the wavefront scattered from the vicinity of the $m$-th specular point. The variable $\rho_{(m)}$ can be negative (i.e. the wavefront is converging). Location of the points where focusing occurs is determined by the equation $R_{(m)} = |\rho_{(m)}|$ and these loci are caustics.

Numerical calculations of ray paths and caustic structure were performed for forward scattering from a single realization of pressure release corrugated surface with a one-dimensional Gaussian wave-number spectrum [8]. A tank experiment has been conducted to measure reflection of underwater sound from surface waves. Reflection from a wave crest leads to focusing and caustics [9]. These studies demonstrate the following. The rays are focused by the trough for sea-floor interface (and by chest for air-water interface) to converge at distances compared with $a_i$ – the radius of curvature of the interface. After passing through the focus the rays diverge to form a fan with well-defined boundaries. The number of rays that reach the receiver ($M$ in the sum of Eq. (7)) depends on geometry with a minimum of one ray. The boundary surfaces where the number of specular points (scattering rays) changes represent the position of caustics. The structure of these boundaries indicates the presence of casp caustics. The apex points (casp points) of each caustic would be the focal points.

Mathematically, the reason for the failure of the specular point analysis is that the quadratic term no longer dominates the behaviour of the integral of Eq. (5) in the vicinity of a stationary phase point and cubic or quartic terms should be accounted. “Focusing” (the $k$ dependence) of the pressure filed is strongest in the casp region where scattering field is described by Pearcey function.

4. DISCSSION AND CONCLUSIONS

The pressure, the bubbles deposited on the leaf blades (see Fig. 2 b) near caustics are exposed, will greatly exceed that of direct acoustic wave. The strength depends on the focusing factor and 'size' of the local concave mirror (the area over which the surface approximates the ideal mirror closely enough). The bubbles are effective contrast agents and thus, will decorate the structure and location of caustics in the backscattering.

Seagrasses, including (Zostera caulescens, Miki) were scanned with the multi-beam sonar SeaBat 9001 on the Sanriku Coast of Japan, facing the Northwestern Pacific [10]. The operating frequency of the ultrasound was 455 kHz. Sixty sonar beams within each swath (combined to form a 90°-wide by 1.5°- long geometrically correct cross-section) provide simultaneous sonar coverage equivalent to about two times the measured depth (Fig. 2a). The direct bubble scattering (the path (a) in Fig. 1) is unlikely responsible for the presence of the individual sparks in backscattering. As scatters, individual bubbles have no great effect at frequencies concerned (of the order of 450 kHz). Moreover, the number of sparks is small in comparison with probable bubble population (see Fig. 2 b). On contrary, the manifestation of caustics (through the paths (b), (c), and (d) in Fig. 1) is a plausible explanation of the observed sonar echoes. The shape of these echo signals reflects both algae population and topography of the bottom.
Fig. 2: Echoes of multi-beam sonar reflecting from Zostera seagrass (after Komatsu et al. [11]) and bubbles on Zostera sp (www.starfish.ch/underwater-foto/Leyte-photos.html).

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MEASUREMENTS AND MODELING
OF LOW-FREQUENCY ACOUSTIC BACKSCATTERING
FROM NEAR-SURFACE OCEANIC BUBBLES

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Abstract: Surface scattering is caused by the interaction of acoustic energy with environmental features at or near the ocean surface. With sufficient winds, air becomes entrained by breaking waves in the form of subsurface bubbles whose properties are governed by advective transport, gas dissolution, and buoyancy. At low frequencies (< 1500 Hz), acoustic scattering from bubbles depends primarily on the air-void fraction, and not on the details of the bubble distribution. A strong contributor to bubble entrainment may be related to turbulence. In this paper, a semi-empirical bistatic extension of the stochastic backscattering strength model of Gilbert, where the scatter from the turbulent microbubbles depends on the wavenumber spectrum of the bubble-induced sound-speed fluctuations, is presented. Here, broadband data from open-ocean experiments are used to provide estimates of physical parameters such as the rms sound-speed fluctuation at the surface, the horizontal and vertical correlation length scales of the sound-speed fluctuations, and the e-folding depth for the rms sound-speed fluctuation. To provide an operationally-useful predictive tool, these parameters are in turn prescribed in terms of their dependence on the wind speed. The resulting semi-empirical model is then used to explore the dependence of surface scattering strength on frequency, angle, and wind speed.

Keywords: Bubbles, surface scattering, turbulence, ocean acoustics
1. INTRODUCTION

Surface reverberation/clutter is caused by the interaction of acoustic energy with environmental features at (the rough air-sea interface) or very near (subsurface bubbles) the ocean surface, with a very strong dependence on surface conditions, waveguide properties, and sonar characteristics. As winds develop, surface waves are generated leading to roughened surfaces that become a source of loss and spread to propagating signals, thus having a major impact on reverberation, particularly in multiple-bounce environments. When winds continue to increase and wave breaking becomes significant, air becomes entrained in the form of subsurface bubbles. Under these conditions, both the rough air-sea interface and bubble clouds may contribute to the acoustic scattering.

This paper presents a semi-empirical bistatic extension of the stochastic effective-surface backscattering strength model of Gilbert [1], where the scatter from the turbulent microbubbles depends on the wavenumber spectrum of the bubble-induced sound-speed fluctuations. Here, low-frequency (< 1500 Hz) broadband data from open-ocean experiments are used to provide estimates of physical parameters of Gilbert’s model such as the rms sound-speed fluctuation at the surface, the horizontal and vertical correlation length scales of the sound-speed fluctuations, and the e-folding depth for the rms sound-speed fluctuation (that results from the decrease in bubbles with increasing depth). To provide an operationally-useful predictive tool, these parameters are in turn prescribed environmentally in terms of their dependence on the wind speed. The paper concludes with data-model comparisons and predictions of bistatic scattering.

2. EXPERIMENTAL DATA

The data for the model developed in this paper are broadband vertically-bistatic SUS data collected by the Naval Research Laboratory (NRL) in 7 open-ocean experiments under SPAWAR’s Critical Sea Test (CST) Program [2]. The CST data are particularly valuable in that the acoustic measurements were accompanied by complementary environmental measurements (such as wind speed, wave-directional spectrum, and subsurface bubble density) and, in view of the potential contribution that fish could make to the backscatter, dedicated measurements of fish scattering were also conducted. Because acoustic signatures that identify fish scattering have been developed [3], it has been possible to isolate their contribution from the CST data, crucial to determining the relative contributions of the air-water interface and sub-surface bubbles, especially at low-to-moderate wind speeds, and at low grazing angles and frequencies.

3. THEORY

Surface scattering strength ($SSS$) (in dB) is due in general to a combination of scattering from the rough air-sea interface and subsurface bubbles:

$$SSS = 10 \cdot \log_{10} \left( \alpha_{\text{hub}} \sigma_{\text{int}} + \sigma_{\text{hub}} \right) ,$$

(1)
where $\alpha_{\text{bub}}$ is a (dimensionless) attenuation factor due to bubbles, and $\sigma_{\text{int}}$ and $\sigma_{\text{bub}}$ are the scattering cross-sections per unit area (per unit solid angle) for the air-water interface and bubbles, respectively. A model describing the interface-scattering component ($\alpha_{\text{int}}\sigma_{\text{int}}$) has been described previously [4]; this paper concentrates on providing a more physical model for the bubble-scattering component ($\sigma_{\text{bub}}$) than in [4].

Breaking waves generate subsurface bubbles whose properties are determined by advection, gas dissolution and buoyancy [5]. At frequencies below ~5 kHz, acoustic scattering from bubbles is primarily due to Bragg scattering from heterogeneity in the sound-speed field, which in turn is proportional to the spatial spectrum of the index of refraction field [1, 5]. At these frequencies, the index of refraction depends on the air-void fraction but not on the details of the bubble-size distribution. Recent investigations [6] have demonstrated that, with high confidence, low-frequency scattering from bubble clouds of low air-void fraction (< 0.01%; microbubble clouds) are responsible for the high backscattering strengths observed in high sea states (as illustrated in Fig. 1).

![Fig. 1: CST SSS values vs. grazing angle at 85 and 1360 Hz for 6 wind-speed (U) bands.](image)

### 3.1. Model

Our closed-form, effective-surface, bubble-cloud scattering model derives from Gilbert [1], who models the mean backscattering cross-section $\sigma_{\text{bub}}$ as a product of a geometric factor $G$ and an effective horizontal wavenumber spectrum $P$:

$$\sigma_{\text{bub}} = G \cdot P,$$

(2)

The geometric factor describes how the backscatter intensity from a bubble cloud depends on two competing factors: 1) its average air-void fraction decreasing rapidly with depth, which suggests most scattering is very close to the surface; and 2) the total acoustic field vanishing at the surface and increasing to a maximum up to several meters below the surface because of interference between the incident and surface-reflected waves (Lloyd mirror pattern). Following Gilbert, we assume that a bubble cloud may be modelled as a vertical distribution.
of uncorrelated point scatterers. The extension of this geometric factor $G$ to bistatic geometries has previously been developed \cite{4} and is given by:

$$
G = \frac{dk_0^3 \gamma_i^2 \gamma_s^2 \left[ 6 + 3 \left( \gamma_i^2 + \gamma_s^2 \right) + \left( \gamma_i^2 - \gamma_s^2 \right)^2 \right]}{2\pi^2 c_0^2 \left( 1 + \gamma_i^2 \right) \left( 1 + \gamma_s^2 \right) \left[ 1 + \left( \gamma_i - \gamma_s \right)^2 \right] \left[ 1 + \left( \gamma_i + \gamma_s \right)^2 \right]},
$$

(3)

where $\gamma_i \equiv k_0 d \sin \theta_i$, and $\gamma_s \equiv k_0 d \sin \theta_s$, and where $k_0 = 2\pi f / c_0$ is the acoustic wavenumber with $f$ the acoustic frequency and $c_0$ the sound speed in bubble-free sea water at the sea surface, $\theta_i$ and $\theta_s$ are the incident and scattered grazing angles, and $d$ is the bubble-cloud air-void fraction e-folding depth.

This paper focuses on generalizing (and investigating the implications of) the spectral factor $P$ of (2). Following Gilbert, we define the effective horizontal wavenumber (power-law) spectrum $P$ for sound-speed fluctuations $C$ in the bubble layer as:

$$
P(q_r, q_z) \equiv \sigma_c^2(0) C(q_r^2, q_z^2) = \sigma_c^2(0) \frac{8\pi^{3/2} N(\beta) L_r^2 L_z}{\left( 1 + q_r^2 L_r^2 + \left\langle q_z^2 \right\rangle L_z^2 \right)^{\beta/2}},
$$

(4)

where $\sigma_c(0)$ is the rms sound-speed fluctuation at the surface, $N(\beta) = \Gamma(\beta/2) / \Gamma(\beta/2 - 3/2)$ is the ratio of two gamma functions, $L_r$ and $L_z$ are the horizontal (isotropic) and vertical correlation length scales of the sound-speed fluctuations, and $\beta$ is the spectral “roll-off” exponent. The horizontal component of the scattered wavevector $q$ is $q_r = k_0 \sqrt{\cos^2 \theta_i + \cos^2 \theta_s - 2 \cos \theta_i \cos \theta_s \cos \phi_b}$, where the bistatic angle $\phi_b$ is defined as the difference in azimuth between the incident and scattered directions. The average value of the square of the vertical component of $q$ over all depths is given by:

$$
\left\langle q_z^2 \right\rangle = \frac{\left( 2 + \gamma_i^2 + \gamma_s^2 \right) \left[ 1 + \left( \gamma_i - \gamma_s \right)^2 \right] \left[ 1 + \left( \gamma_i + \gamma_s \right)^2 \right]}{d^2 \left[ 6 + 3 \left( \gamma_i^2 + \gamma_s^2 \right) + \left( \gamma_i^2 - \gamma_s^2 \right)^2 \right]},
$$

(5)

3.2. Discussion

The expression (4) generalizes Gilbert’s form for $P$ by not assuming that $L_r = L_z$, while the expression (5) extends Gilbert’s form for $\left\langle q_z^2 \right\rangle$, and so $P$, from monostatic to 3D geometries. Together with (3), this gives a fully bistatic, physical model for $\sigma_{hub}$. However, as Gilbert noted, very little is presently known oceanographically about the 4 bubble parameters of $P$. Accordingly, in the next section we invert for these quantities via SSS low-frequency data-model fits.
4. SEMI-EMPIRICAL BUBBLE SCATTERING MODEL

In this section, the broadband CST data are used to estimate the physical parameters of (4). The goal is a semi-empirical form of (2) (and, so, of (1)) that depends environmentally on primarily just the wind speed. (It will also mildly depend on $c_0$.)

As a first step, we relate the bubble-cloud air-void fraction e-folding depth $d$ (m) to the wind speed at 10 m $U$ (m/s) via the following modification of the empirical formulas of Farmer and Vagle [7]: $d = 0, U \leq 3$; $d = 0.078 \cdot U, 3 < U \leq 8$; and $d = 0.51 - 0.1 \cdot U + 0.01 \cdot U^2, U > 8$. Using this expression for $d$ and assuming the semi-empirical model developed [4] for $\alpha_{int}, \sigma_{int}$, fits over grazing angle were then obtained to the SSS data for 12 frequency bands for each CST measurement. (The residual standard errors for these fits were typically < 3 dB.)

Fig. 2 displays the bubble parameter values of Eq. (4) resulting from these data fits as a

![Fig. 2: Estimates of $L_r$, $L_z$, $\sigma_c(0)$, and $\beta$ vs. wind speed for 12 CST frequency bands.](image)
function of wind speed and frequency. It can be seen that the values for $L_r$ and $L_z$ have a
discernable linear dependence on wind speed up to about 15 m/s, and a slightly decreasing
slope as frequency increases. While not apparent, this frequency dependence was found to be
log-linear for wind speeds below 15 m/s. It can be seen that $L_z \leq L_r$, one of the physical
constraints used in fitting the acoustic data. The values for $\sigma_c(0)$ indicate a consistent linear
dependence across all frequencies and wind speeds, while $\beta$ takes on a constant value of
about 4. It should be noted that the parameter values obtained through fitting the data were
strongly dependent on the initial value of the parameter $\beta$ used when applying the non-linear
least squares algorithm used in this study. However, through trial and error, it was found that
a value of $\beta = 4$ produced the smallest residual error when fitting the combined data set, and
resulted in the following relations for $L_r$, $L_z$, and $\sigma_c(0)$:

$$L_r = 1.0094 f^{-0.0887} (U - 3)$$
$$L_z = 1.6497 f^{-0.2043} (U - 3)$$
$$\sigma_c(0) = 25.1236 \cdot (U - 3).$$

Using a different set of open-ocean data, Gilbert estimated $\beta$ to be 3.86 and $\sigma_c(0)$ to be
in the range of 44 to 315 m/s for wind speeds of 10 to 15 m/s, both of which are consistent
with our findings. The spectral model $P$ used in this study is a generalization of that used by
Gilbert which assumes isotropic turbulence and defines correlation length in terms of the
horizontal extent of the bubble layer. While our correlation-length estimates are not directly
comparable, we found both mild frequency dependencies and a slight difference in vertical
vs. horizontal correlation lengths which is not inconsistent with isotropic turbulence.

In sum, this work extends Gilbert’s results by providing a bubble-scatter model
parameterized in terms of the wind speed which provides good fits to an extensive set of
open-ocean surface scatter data. In addition, a power spectrum was developed which provides
closed-form expressions for the rms sound-speed fluctuations and the associated horizontal
and vertical correlation length scales.

5. DATA-MODEL COMPARISONS AND BISTATIC PREDICTIONS

We now briefly exercise the model to explore the dependence of SSS on frequency, angle,
and wind speed. Fig. 3 displays data-model comparisons, while Fig. 4 shows bistatic model
predictions. Together, they show that bubble scattering has a strong dependence on both wind
speed and frequency, but generally a mild dependence on angle, except at low grazing angles
or bistatically near specular. In general, interface scattering dominates toward specular [4],
while bubbles (when wave breaking is significant) become an increasingly important driver
of SSS with decreasing grazing angle and with increasing frequency and wind speed.
Fig. 3: Comparisons of SSS data (symbols) and model predictions (curves) vs. (a) mean grazing angle at 1 kHz for 3 wind speeds, and for a mean grazing angle of 15 deg, SSS vs. (b) frequency for 3 wind speeds and (c) wind speed for 2 frequencies. The gray model curves represent just the bubble-scattering contribution $10 \cdot \log_{10} (\sigma_{bub})$ to SSS.

Fig. 4: Bistatic model predictions of bubble scattering strength $10 \cdot \log_{10} (\sigma_{bub})$ at 1 kHz for $\theta_i = 15^\circ$ vs. (a) scattered angle ($\phi_s = 180^\circ$) and (b) bistatic angle ($\theta_s = 15^\circ$).
6. SUMMARY AND FUTURE WORK

We have presented a new semi-empirical, closed-form, effective-surface, bistatic bubble-scattering model applicable to low frequencies (< 1500 Hz) that relies environmentally primarily on just the wind speed. It provides a more physical model than used in [4], and with its semi-empirical nature, the new model allows for new insights into the oceanographic mechanisms controlling acoustic scattering from assemblages of sub-surface oceanic microbubbles. One caveat is the lack of single-bounce, out-of-plane bistatic and forward-scatter data for vetting/refining the model’s predictions for these non-backscatter geometries.

Future work will include incorporating additional data sets and extending this formula to higher frequencies (as in [4]).

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REFERENCES

ACOUSTIC MEASUREMENTS OF BUBBLES IN THE WAKE OF SHIP MODELS

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Abstract: The interest to bubble generation by moving ships is connected with large area of bubble wake reaching several kilometres that can be used for ship detection. We developed acoustic system for the measurements of bubble density and conducted measurements in 100 m long towing tank. The developed system measured attenuation of ultrasound in wide frequency band from 50 to 800 kHz between two acoustic sensors placed on the distance 20cm. The attenuation of sound produced by bubbles was observed during several minutes after model of ship passed the point of measurement. The attenuation was recalculated to the bubble size distribution for bubbles from 4 to 65 microns using theory of resonance bubble attenuation. The measured bubble size distribution can be interpolated by power dependence $n(R) \sim R^{-3.5}$ that is typical for bubbles at sea subsurface layer. The dependencies of bubble concentration of model ship speed and type of propeller were investigated. The generation of bubbles was observed when the model speed exceeds definite threshold that can be connected with the cavitation threshold. The theory describing dynamics of wake turbulence based on the shear-free turbulent wake was developed. The measured bubble concentration was in good agreement with the developed theory.

Keywords: Ship wake, bubbles, sound attenuation
1. INTRODUCTION

Moving ships can generate bubbles by propeller cavitation, by the breaking of ship generated waves, and by air entrapment in the turbulent boundary layer under the ship hull. Interest in bubbles produced by moving ships is often connected with the bubble influence on sound propagation near the ship. The bubble layer can affect the parameters of sound propagation through the layer [1-5] and ship noise radiation [6]. The interest in bubble generation by moving ships is also connected with the opportunity of ship detection, since bubble wakes can reach lengths of several kilometres.

The first detailed tests of bubble measurements in wakes of ships and submarines were conducted during WWII [1] using measurements of sonar signal scattering and attenuation. These methods are still widely used for bubble measurements. High frequency multibeam sonars were used for measurements of spatial bubble distributions in wakes of ships [4,5]. The experiments demonstrated that the length of the bubble layer can reach 1500 m and its depth could be up to 10 m.

Even if it was known that bubble generation in the wake of a ship is connected with turbulence generated by the ship, there is no developed theory for prediction of bubble concentration. Such a theory could be used for estimation of bubble concentration for various ships, propellers and speeds. It could predict possible distances of bubble detection. Also, temporal variation of bubble density can be used for ship classification and its speed estimations.

The first step in the development of this theory is to connect known theory of ship turbulent wake generation with results of experimental research. It was shown [7] that a shear-free turbulent model is a good approach to describe the turbulent wake behind a self-propelled body, or ship-wake. Later [8,9] this approach was extended to allow the wake turbulence parameterization by characteristics of the wake source.

There are limited field data available, and more can be collected in controllable conditions in a laboratory tank. The bubble measurements using a ship model in a towing tank can be used for theory validation and for the determination of the theoretical model parameters.

For measurements of bubble density in the towing tank we applied a method based on the attenuation of acoustic waves in a wide frequency band (100-800 kHz), which allows detection of bubbles with radii 4-32 μm.

2. EXPERIMENTAL SETUP

A self-propelled ship model was used in experiments conducted at Stevens towing tank having length of 100 m, width of 6 m and depth of 3 m. The ship model has a length 1.60 m, maximum width 0.3 m, and height about 0.2 m. The model was moved in the tank using a special control system to set the propeller rotation so as to keep the propulsion reaction on a supporting strut near zero. Different sizes and shapes of propellers were tested. In this paper we present results for two of them: propeller #2 has diameter of 0.09 m, and blade width of 0.04 m; propeller #3 has diameter of 0.105 m and blade width 0.025 m (see Fig. 1).
The bubble-produced excess attenuation was measured over a wide frequency band using an acoustic radiation with a linear frequency sweep (Fig. 3(a)). In our experiments, the radiated signal was received by the receiver and sent to an electronic processing unit. To increase the signal-to-noise ratio and to cancel the influence of reflections, we cross-correlated the radiated and received signals. This cross correlation was filtered using band pass filters. In this paper, we present results of measurements in 7 frequency bands from 100 to 800 kHz having width of 100kHz.

The attenuation of sound was measured by comparison of the band pass filtered cross-correlation of received signals in clear water with amplitude of the same signal measured in water with bubbles. In the bubble wake, bubbles of various sizes are present. They are described by the bubble size distribution function $N(a)$, so $N(a)\,da$ is number of bubble in volume unit having radii from $a$ to $a+da$. For recalculation of the sound attenuation to the bubble density we can apply the widely used expression for sound attenuation of a signal with frequency that was derived in the assumption that sound attenuation was produced by the resonance bubbles [1,10]

$$n(a) = \frac{4.62 \times 10^{-12} \, f_a^2 \, I}{L}$$

Where $f_a$ is bubble resonance frequency, $L$ distance between transmitter and receiver, $I$ attenuation of sound between transmitter and receiver in dB. For bubbles near the surface, the resonance frequency of a bubble is connected with the bubble radius by relationship $f_a = 3250/a$, here frequency is presented in kHz and bubble radius in $\mu m$.

The temporal variation of acoustic signal attenuation connected with bubble presence was measured at 10 points, positions of which are shown in Fig. 3.
3. RESULTS OF MEASUREMENTS

The conducted experiments demonstrated that after the ship passing, the strong attenuation of an acoustic signal was observed that rapidly decreases to a relatively small level. The first strong attenuation is probably produced by relatively large bubbles that rapidly rise to the surface. Relatively lower longer time attenuation is probably produced by small bubbles. The concentration of small bubbles decays much more slowly than the concentration of large bubbles. This is probably due to the slow spreading and dissolving of small bubbles. The example of such attenuation variation is shown in Figs. 4 for two tests with the same experimental conditions.

![Fig. 3. Schema of acoustic sensor placement.](image)

![Fig. 4. Time dependence of acoustic wave attenuation (Fig. 5) presented in log/log scale. Solid line shows theoretical dependence $\alpha \sim t^{-2/3}$.](image)

![Fig. 5. Maximal sound attenuation produced by large bubble at various distances from the axis at depth 5.08 cm for various propellers and speeds. Band of filtering 200-300kHz.](image)

![Fig. 6. Acoustic attenuation temporal variation in three frequency bands 200-300kHz, 500-600kHz, 700-800kHz measured at the point at distance 20 cm from the axis and depth 5.08 cm. Model with propeller #2 was moved with speed 2.74m/s](image)

For estimation of large bubble spatial distributions, the maximum attenuation, which occurred a short time after ship passing, was presented for various spatial points for various propellers and model speeds. Fig.5 presents dependence of measured attenuation produced...
by large bubbles on distance from axis. The measurements were conducted at depth of 5.08 cm for various model speed and propellers.

Filtering in several frequency bands allows the estimation of bubble size distributions. Fig. 6 shows the time dependence of attenuation measured for the model with propeller #2 moving with speed 2.74m/s, for the point at distance 20 cm from the axis and depth 5.08 cm. It is seen that for time up to 60 s the attenuation of all bubbles is roughly the same. This means that according the formula (1) bubble size distribution is proportional to $\alpha^{-3}$.

The rough estimation of total gas volume (void fraction) can be made based on the assumption that the maximal size of bubbles is about 30 $\mu$m (resonance bubble frequency 110 kHz) and attenuation is about 0.5dB/m for higher frequency. In this case the estimation of void fraction of small bubbles gives the value $3 \times 10^{-9}$. This does not include possible contribution of relatively large bubbles, which was not investigated.

4. TURBULENT MODEL AND COMPARISON WITH EXPERIMENT

We have employed the turbulent-wake theory [7,8] for the data interpretation. This model can not be applied to describe behaviour of larger bubbles. Therefore, we will apply it for estimation of temporal variation of small bubble concentrations. Because the micro-bubbles with radii less than 30 $\mu$m have a rising speed less than 0.2 cm/s [5] the rate of bubble degradation is low enough so that the micro-bubbles in the wake turbulent field can be assumed as a passive admixture of the wake water body. Hence we have coupled the wake theory with the bubble turbulent diffusion in the wake as a passive admixture. In this case the total bubble mean concentration in a wake cross-section obeys the bubble mass-conservation law. That means that the mean bubble concentration decays with inverse proportionality to the area of wake cross-section. According to the wake theory the wake radius increases in time proportional to $tn$, where the power coefficient $n$ is in the range $0 < n < 0.5$ [7-9]. Then the cross-section area of turbulent wakes increases in proportionality to $t^{2n}$, and the mean bubble concentration decays as $t^{-2n}$. As is seen from Fig. 6, the bubble concentration decay in time for all collected data can be approximated by function that is proportional to $t^{-2/3}$, therefore the coefficient $n=0.33$. This result agrees with the theoretical prediction on limitations for the coefficient $n$ and known numerical values of $n$ for surface ships which are about $n = 0.2 – 0.3$. Hence, this result is qualitatively consistent with theoretical conclusion and quantitatively with known estimates of this coefficient.

5. CONCLUSION

The acoustic system, based on sound attenuation in a wide frequency band (100-800kHz) was used for measurements of bubble concentration in the wake of a self-propelled ship model in the Stevens towing tank. It was found that the ship model can generate two kinds of bubbles: one kind consists of large bubbles that rise rapidly to the surface and with a lifetime that does not exceed 15 seconds. The second kind, the smaller bubbles can be detected much longer and their size distribution is in a range from 4 to 32 microns. This was estimated from the sound attenuation data using bubble resonance theory. In many cases bubble size distribution was proposition to $\alpha^{-3}$. Bubbles with radii around 10-20 microns were observed for up to five minutes.

As observed in experiments the temporal variations of bubble concentration can be approximated as $t^{-2/3}$. This power law is qualitatively consistent with theoretical conclusion and quantitatively with known estimates of the model parameters.
The attenuation technique used in this work is a scientific tool for measurements of bubble concentrations in wide range of bubble sizes. From the perspective of best sensitivity of bubble detection, nonlinear acoustic methods look preferable, as they that can detect even single bubbles with radii around few microns [11]. Application of these methods can extend the bubble wake detection time.

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Spatial and Temporal Correlation Characteristics of Acoustic Path Differences across Surface-Ship Wakes

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Abstract: A series of three CW-pulsed signals were transmitted between two ships across a surface ship wake. These transmissions each contained a set of four 0.5-ms long pulses that were propagated across the wake every second. The twelve pulses ranged over frequencies from 30 to 140 kHz in 10-kHz steps. Each transmission was received on a 10-hydrophone string being towed on one side of the wake. The cross-correlation coefficients between pairs of hydrophones spaced 1 meter apart were calculated. The variability of these correlation coefficients, which are related to the changing acoustic paths to each of the ten hydrophones, were compared to the changing bubble number densities and bubble size distribution clusters as the wake aged. The correlation coefficients were obtained for ship speeds 12 and 15 knots. Work supported by the Naval Research Laboratory Program Element 62435N
OBSERVATIONS OF THE EFFECT OF INTERNAL WAVES ON SUBSURFACE LAYER OF AIR Bubbles IN THE SEA

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Abstract: During our study of currents and internal waves on the shelf of the Sea of Japan by the RIO GRANDE 600 KHZ Acoustic Doppler Current Profiler from a yacht, the effect of internal waves on the subsurface layer of air bubbles was detected. It was revealed that the subsurface layer of intense acoustic scattering in areas above internal wave trains changes its thickness, patterning profiles of internal waves propagating under it. The effect was observed both for the case of internal depression waves and for the case of internal elevation waves. In passing the frontal slope of the internal depression wave, a decrease in the thickness of the scattering layer takes place while, at passing the back slope, an essential decrease is observed. In contrast, internal elevation waves are accompanied by synchronously thinning and thickening the bubble layer during passing the front face and back slope of the internal wave, respectively.

Keywords: Internal waves, surface waves, air bubbles, subsurface layer, backscattering, sea surface, currents, acoustic profiler
1. INTRODUCTION

The subsurface sea layer is saturated by gas bubbles that cause enhanced sound scattering and absorption. The concentration of the bubbles is maximal near the surface and gradually decreases as the depth increases. If the picnocline is close to the sea surface, one can expect the influence of internal waves on the subsurface layer of the air bubbles. In [1], it was experimentally established that the outcome of internal waves to the sea surface leads to a periodical destruction of the gas-bubble layer, with simultaneous generation of slick stripes on the surface, which are separated by 200-300 m. Measurements performed in a shelf zone by using the 600-kHz Acoustic Doppler Current Profiler (ADCP) nearly always depict the subsurface layer as that of maximal sound backscattering. In measuring currents and internal waves with the ADCP on the shelf of the Sea of Japan, we faced interest specificities in the records of echo-signals in the subsurface sea bulk, which evidence for a strong effect of internal waves on the subsurface layer of air bubbles.

2. OBSERVATIONS OF THE EFFECTS OF INTERNAL WAVES IN THE SUBSURFACE PICNOCLINE

In October 2003, we performed measurements from a yacht equipped by the ADCP along a 10-km tack, southwards from the Vityaz’ bay. The weather was fair, the south wind was weak (2—3 m/s). Surface waves were not higher than 20—30 cm. The picnocline was at a depth of 12 m. The measurements coincided with the phase of the surface tide: the current of northern direction (towards the coast) prevailed in the upper water layer (from the surface to a depth of 25 m) while the countercurrent existed in the near-bottom layer (30 m and deeper). In the coastal part of the tack with sea depths of about 40 m, we observed groups of solitary internal waves, 4—4.5 m in height, that propagated towards the coast with velocities of about 0.25 m/s (Fig. 1). Because the internal waves were rather intense and the picnocline was close to the surface, the wave profiles had a characteristic shapes with sharp troughs and smooth crests. Such a feature is quite common for internal waves in sea shelf zones and is an evidence for their nonlinearity [2, 3].

Let us in more detail consider the records of the backscattering intensity obtained with the ADCP on that tack. Fig. 2 shows the averaged dependence of the volume backscattering coefficient, $m_V$, on the depth for each of four beams of the ADCP. The averaging is carried out for 100 successive measurements separated by 5—6 m on the tack. The profile exhibits the subsurface water layer, 2—2.5 m in thickness, where the backscattering intensity reaches its maximal value of 76 dB and the layer near the 12.5-m horizon where an intermediate maximum of 60 dB exists. The latter peak clearly displays the oscillations of layers under the influence of the internal waves. The measurements also showed that the subsurface layer of intense backscattering changes its thickness when the yacht passes over it, as if it replicates the profiles of internal waves propagating under the layer. Figure 3 presents a fragment of the record of the backscattering intensity for the moment of passing two internal waves with heights of 5 and 5.5 m. The formation of that sound scattering layer can be attributed both to the existence of air bubbles generated by breaking surface wave and to the existence of various suspended particles and marine organisms. The presented data show that the thickness of the layer varies according to the period of the internal wave.
Fig. 1: Records of the backscattering intensity on the tack of October 26, 2003. The record shows nonlinear internal waves propagating from the continental slope to the coast. Sea depth and the spatial scale (in m) are laid off the vertical and horizontal axes, respectively.

Fig. 2: Averaged depth dependence of the backscattering coefficient $m_V$ obtained on the tack (averaging over 100 successive measurements).

The cross-sections obtained at different phases of the internal wave (Fig. 4) allow one to draw more accurate conclusions. The thickness of the scattering layer increases and reaches 4 m (cross-sections 2 and 6) in passing the depression wave (the ADCP measures the $m_V$ value starting from a depth of 1.25 m). When the back slope of the internal wave passes, the
The aforementioned value decreases. In addition to measuring the depth dependence of $m_V$ at a frequency of 600 kHz, we also estimated the concentration of the resonant bubbles for that frequency. The radius of resonant bubbles at 600 kHz is about 5 mcm. By supposing that main sound scatterers in the subsurface layer are the resonant bubbles, we calculated their concentrations for different phases of the internal wave. The concentration at a depth of 1.25 m was close to 2300 1/m³ before the passage of the wave. At the moment of approaching the front face of the wave the concentration increased to 56000 1/m³ while it sharply decreased to 630 1/m³ after passing the back slope of the wave.

Let us briefly consider probable causes of the effect observed. Internal waves propagating over the subsurface pycnocline rather strongly interact with the sea surface because of their own orbital movements. The orbital velocities accompanying the observed internal waves...
reached values of 0.1—0.2 m/s. Such values proved to be sufficient for the successful areas of divergence and convergence at the sea surface to affect the surface waves that were flatten in one case and enhanced in another one. Actually, there was visually observed that the surface waves overturn when the front of the internal wave passes by. When the back slope of the internal wave passed, the “white horses” fully vanished. The overturning of surface waves leads to an increase in the quantity of the air bubbles that are probably the main sound scatterers. The lateral size of the areas of intense scattering was approximately 60—70 m, about one third of the internal wavelength. One more important aspect should be mentioned. On the one hand, internal waves cause an increase in the quantity of air bubbles in the subsurface sea layer because of their orbital movements while, on the other hand, internal waves carry bubbles thereby generating areas of decreased and increased concentration, the fact that was also observed in our measurements.

3. OBSERVATION OF THE EFFECT OF INTERNAL WAVES IN THE NEAR-BOTTOM PICNOCLINE

Let us now consider another example of the effect at hand. In September, 2008, we performed measurements of internal waves from an anchored yacht with the ADCP, also on the shelf of the Sea of Japan. The sea depth was 43 m. The measurements started at a nice weather. However, a few hours later, the sea state was gradually becoming higher. After 8 hrs of experimenting, the measurements were terminated because of the brewing storm. Nevertheless, we managed to record two trains of internal waves propagating towards the coast (Fig. 6). The specificity of the obtained record consists in that the first train was observed when the picnocline was in the middle of the water bulk while the second train corresponded to the near-bottom position of the picnocline. In both cases, the boundary of the subsurface layer of the air bubbles suffered from the influence of internal waves, in analogy with the situation observed in 2003 when the subsurface layer replicated the profile of

![Fig. 6: ADCP record of two trains of intense internal waves in September, 2008 (the backscattered signal).](image-url)
internal waves. The difference consisted in that the waves of the first train (the depression waves) caused the embedding of the air-bubble layer above the front face. However, in the case of internal elevation waves, the front face corresponded to thinning the layer. Such a feature can be also attributed to the action of orbital movements.

4. CONCLUSIONS

To conclude with, it is worth mentioning that the presented examples are not exclusive. The aforementioned behavior of the air-bubble subsurface layer influenced by internal waves has been repeatedly established in crossing both subsurface and near-bottom thermoclines by trains of internal waves and solitary internal waves. Such observations were performed by us during several summer seasons.

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REFERENCES

CHARACTERISTICS OF ACOUSTIC SCATTERING FROM HYDRATE SHELLED BUBBLES

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Abstract: Gas and oil seepages in shallow submarine environments produce buoyant hydrocarbon plumes that are detected as sonar targets within the water column. Within the gas-hydrate stability zone – for high hydrostatic pressures and low temperatures, methane-hydrate ice skins are formed on rising seep bubbles. To find correlations between the upper boundary of the gas flare and the predicted depth of the gas hydrate stability zone, the models of hydrate coating have been derived. In addition to the consolidated layer (ice), which rheological properties are defined by Kelvin–Voigt model, the shell has been considered as a layer of snow. The rheological properties of such media are described by the Burgers model. Based on these models, bubble scattering cross sections have been calculated. Decreasing of hydrostatic pressure upon a wall of the rising bubble leads to the rupture of the shell. The intense acoustic radiation accompanying shell fragmentation and pulsations of the realised bubble results in the significant increasing of the noise within an effective acoustic waveguide – gas flare, and can serve as an indicator of the hydrate shell formation on rising seep bubbles.

Keywords: Bubble, gas-hydrate, scattering,

1. INTRODUCTION

The persistence of methane gas bubble plumes rising as much as hundreds meters in the water column is remarkable because the ocean is undersaturated in methane, and the bubbles should quickly dissolve. A probable explanation was proposed by Merewether et al. [1] who inferred that the bubbles were protected by a coating of oil or gas hydrate. The latter was
suggested to cause the reduced shrinking rate of methane gas bubbles. Gas hydrate is an ice-like solid that results from the trapping of methane molecules – the main component of natural gas – within a lattice-like cage of water molecules.

Active methane gas seeps were observed on the Sakhalin slope in the Sea of Okhotsk. Echosounding systems using 12 and 18.5 kHz proved to be of use for imaging these bubble plumes in the water column, and aided in locating and sampling of vent sites. To find correlations between the upper boundary of the acoustic backscatter image of the gas flare and the predicted depth of the gas hydrate stability zone (GHSZ), the evaluation of the upper boundary of GHSZ has been carried out on the base of the model developed by [2] for pure methane gas using the temperature profiles taken during the cruise KOMEX-II. Figure 1 demonstrates this comparison for the Obzhirov structure.

![Figure 1: Relationship between the height of the acoustic backscatter images and the upper boundary of the gas hydrate stability zone for the Obzhirov seepage](Echosounding record courtesy of A. Salomatin).

The upper boundary of the acoustic backscattering images – the rapid disappearance of the acoustic signals was interpreted to be the result of dissolution of gas hydrate skin and subsequent enhanced shrinking of the residual gas bubbles soon after they rise above the stability zone. To provide a theoretical basis for this conjecture the scattering cross-section of the bubble coated by hydrate skin should be determined.

2. ACOUSTIC SCATTERING FROM SHELLED BUBBLE

The regimes of hydrate formation on methane bubbles and dynamics of dissolution of such encapsulated bubbles have been the subjects of laboratory [3], as well as field [2] experiments. A simple model can be proposed to characterize the rheological properties of hydrate shell. An analogy between acoustic manifestations of the hydrate shelled bubble and ultrasound contrast agent [4, 5] – microbubble whose surface is occupied by lipid or polymer molecules forming a shell and whose scattering signature provide a method of enhancing effectiveness of medical ultrasonic diagnostic – requires a marked correction. The approximation of the incompressible shell being satisfied for the rubber like media, where
sound speeds of the longitudinal and transversal waves are markedly different, has a limited applicability for the hydrate shelled bubble.

The linear problem of scattering of plane sound wave \( P_i(\mathbf{r}, t) = p_m \exp[(\mathbf{k} \cdot \mathbf{r}) - \omega t] \) (here \( \omega \) is the frequency, \( \mathbf{k} \) is the wave vector, \( c = \omega / k \) is the sound speed in liquid, and \( p_m \) is the amplitude of the wave) from the bubble coated by visco-elastic shell has an exact solution. The scattered wave at \( \mathbf{r} \) in response to an incident wave \( P_i(\mathbf{r}, t) \) can be represented in far field as

\[
P_s(\mathbf{r}, t) = f_s P_i(\mathbf{r}, t) \exp(ik|\mathbf{r} - \mathbf{r}_b|)/|\mathbf{r} - \mathbf{r}_b|.
\]

Here \( f_s \) is the scattering amplitude which can be considered isotropic as long as \( kR << 1 \) (noting particularly that \( (kR)^2 \) must be negligible [6]) and approximated by

\[
f_s = -R \frac{\omega^2}{\omega^2 - \Omega_0^2(1 - ikR)}, \quad \Omega_0^2 = \frac{3\gamma P_0 + 12\mu h}{1 + 4\mu/3K} \left( \rho R^2 \right)^{-1}.
\]

In derivation of this equation (1), we assumed that the behavior of the gas core is polytropic, the thickness of the shell is small \( h << R \), \( K \) and \( \mu \) are modules of compression and shear in the shell. Equation (1) is distinguished from those one, used for description of scattering from contrast agents [7] by co-factor \( (1 + 4\mu/3K) \). For the rubber like shell \( \mu / K << 1 \), and in this case equation (1) takes the usual form. However, for the hydrate shells, the elastic module of which are very close to ones of the common ice \((K \approx 7.5 \times 10^9 \text{ Pa}, \mu \approx 3.5 \times 10^9 \text{ Pa})\), the correction factor should be accounted.

In considering scattering from consolidated shell (1) we did not account dissipative effects in liquid, shell and gas. This approximation is justified, since the zone of stability for gas-hydrates is located at depths in hundreds meters and high hydrostatic pressure in tens atmospheres provides that the dominant mechanism of losses near the resonance \((|\omega - \Omega_0|/\Omega_0 << 1)\), where they should only be accounted, is the radiation damping. This type of losses has been accounted in derivation of equation (1).

Another peculiarity, that distinguishes cardinally the behavior of the scattering amplitude (1) from one for the conditions of the laboratory experiments with contrast agents, is the high equilibrium pressure in the bubble, which is near hydrostatic \( P_h(0) = P_h(0)(1 + z/H) \), \((P_h(0) \approx 10^5 \text{ Pa}, H \approx 10 \text{ m})\) and varies with the depth.

Figure 2 illustrates the variation of the back-scattering cross-section with the driving frequency and the bubble radius for the frequency range \( f = (\omega/2\pi) : 3.5 \pm 135 \text{ kHz} \) used at the field experiments on the Sakhalin slope in the Sea of Okhotsk, and for the interval of bubble sizes \( R: 0.1 \pm 0.5 \text{ mm} \). The values of physical parameters required to calculate the cross section were chosen as follows: the sound speed in water was set equal to \( c = 1453 \text{ m/s} \), the depth was taken to be \( z = 500 \text{ m} \), the shell thickness was set equal to \( h = 2 \text{ \mu m} \), the bulk modulus \( K \) and the shear modulus \( \mu \) were taken to be \( K = 7.5 \times 10^9 \text{ Pa} \) and \( \mu = 3.5 \times 10^9 \text{ Pa} \). These values of the elastic modules have been measured at laboratory conditions for the pure mono-crystal samples and are very close to the values for common ice. In natural conditions the gas-hydrate shell is formed in some minutes [3], however within several seconds a seep bubble formed at the seabed will become covered by a layer of surface active substances. This means that the shell will contain a large number of impurities and defects, thus, as in the case of a common sea ice, the values of the elastic modulus can decrease on an order of magnitude \( K \approx 6.5 \times 10^8 \text{ Pa}, \mu \approx 3 \times 10^8 \text{ Pa} \).
3. SHELL RUPTURE

In evaluating the scattering cross section we neglected the difference between hydrostatic pressure $P_h(z)$ and gas pressure in the bubble $P_g$. Decreasing of hydrostatic pressure acting upon a wall of the rising bubble will lead to the increasing pressure drop (decompression) across the shell. One can calculate the stress generated by decompression in the shell and evaluate a characteristics distance of rising at which the stretching stress will exceed the tensile strength of gas-hydrate. At this depth the shell will be fragmented.

Let at the depth $z_0$ the bubble is in equilibrium state: $P_{z0}=P_{g0}(z_0)$. The internal and external radiuses of the shell are $R_{10}$ and $R_{20}$, correspondingly. The shell is deformed at this state: $R_{2e}-R_{2e}\approx R_{10}-R_{e}=-R_{10}(P_{g}(z_0)/3K)$, where $R_{2e}$ and $R_{e}$ are the equilibrium radii of the nondeformed shell. Parameters of the state at the depth $z_1$ ($(z_0-z_1)/z_0<<1$) are defined by an equilibrium condition within phases and a continuity of pressure and displacement on interface surfaces. According to [7] the stretching stress in the shell at the depth $z_1$ will be defined by the following expression

$$
\sigma_{g1} = \sigma_{g2} \approx -P_{g}(z_1) + \frac{6\mu[P_{h}(z_0)-P_{g}(z_1)]}{12\mu(R_{20}-R_{10})R_{10}^{-1}+3\gamma P_{h}(z_0)/(1+4\mu/3K)}.
$$

(2)

Tensile strength of dirty ice is near $10^5$ Pa, therefore, rising over the distance

$$
(z_0-z_1) \geq 2(z_0+11H)\left[(R_{20}-R_{10})R_{10}^{-1}+(3\gamma P_{h}/12\mu)(1+z_0/H)(1+4\mu/3K)\right]
$$
the shell will rupture. The values of physical parameters required to evaluate (3) were chosen as follows: \( z_o = 500 \) m, \( h = 2 \) \( \mu \)m, \( R_{0} = 2.5 \) mm, \( K \approx 6.5 \times 10^8 \) Pa, and \( \mu \approx 3 \times 10^8 \) Pa (dirty ice). Substituting these into (3) yields \( z_0 - z_i \approx 9 \) m. Thus, during the main time on their way to the boundary of gas-hydrate stability zone, the shell represents pancake ice. Therefore, along with the model of the consolidated layer considered above, rheological properties of which are defined by Kelvin–Voigt model, there can be more appropriate to use the model of unconsolidated medium that is a layer of snow.

The deformation of snow and ice under nondestructive loading has long been described in terms of the four-element Burgers model. Its limiting cases are the Kelvin–Voigt model considered above and the Maxwell model which we used earlier to describe properties of gas-hydrate shell. For a considered case of linear oscillations and for periodic external forcing, calculations in the Burgers model are reduced to the expression (1) provided the shear modulus \( \mu \) is changed to renormalized one.

### 4. MICRO EXPLOSION

Along with the conventional (active) methods of gas plume sounding, the passive acoustic emission of bubbles on their way from the injection to the horizon of dissolution can be used to size their population and natural frequencies [8]. For the free bubbles (without shell), the noise spectrum emitted by marine hydrocarbon seeps was a subject both theoretical [9], and experimental studies [10]. The specific mechanism of acoustic emission attributed to the shelled bubble is realized at the moment of shell rupture. This mechanism is similar to one used in pneumatic seismic sources – air-gun and is realized at underwater explosions [11].

The thin shell can sustain only some bars (some tens meters of rising), therefore pulsations of the released cavity at the depth in hundreds meters will not be of a large amplitude. For the same reason the velocity of the bubble wall during the initial moment following occurrence of cracks expanding over the shell is essentially less than the sound speed in the surrounding liquid. As a result we can neglect nonlinearity and use the linear Rayleigh equation to analyze emission of a bubble with fragmented shell [11]

\[
R = R_{\infty} + ae^{-\delta t} \cos(\Omega_{\alpha}t - \alpha),
\]

\[
R_{\infty} = R_{00} \left[ 1 + \frac{P_g - P_h(z)}{3\gamma P_h(z)} \right], \quad a = R_{00} \frac{P_g - P_h(z)}{3\gamma P_h(z)}, \quad \alpha = \sqrt{\frac{3\gamma P_g}{\rho c^2}}, \quad \delta = \frac{R_{00} \Omega_{\alpha}^2}{2c}.
\]  

(4)

The acoustic signal radiated by a bubble after destruction of the shell (4), is much more intensive, than the sounds emitted from entrained bubbles after impacting of a rain drop (or sea splashes) a liquid surface [8, 12]. The acoustic trace following bubble injection from a nozzle in laboratory studies [8] or from the sea floor vents [10] has amplitude of orders magnitude smaller than the radiation arising after destruction of the shell (4). This effect – intense acoustic radiation accompanying shell fragmentation, – can lead to the significant growth of noise level within an effective acoustic waveguide – gas flare [9], and can serve as an evident indicator of hydrate shell formation on rising seep bubbles.
5. CONCLUSIONS

We have investigated the possibility of detecting the presence of methane-hydrate ice skin which is formed on rising seep bubbles at high hydrostatic pressures and low temperatures. Our approach is to formulate active and passive acoustic methods by exploiting the way the presence of elastic shell modifies the acoustic signature of bubbles. The model of hydrate coating has been derived. Based on this model bubble scattering cross section has been evaluated. The presence of the shell affects the character of the scattering only for the near resonant bubbles. For smaller bubbles the scattering follows to the Rayleigh law and they are irresolvable in back scattering. The scattering cross section of greater bubbles is defined by their geometrical sizes. It is shown that difference of hydrostatic pressure in some bars destroys the consolidated shell. Analytical expression for the shape of the acoustic signal radiated by a bubble after rupture of the shell has been derived. This radiation appears intensive enough, exceeding in orders of magnitude the noise of a rain or ‘birthing wails’ of the bubbles as they depart from the vent.

6. ACKNOWLEDGEMENTS

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Structured Session 18

Measurement Activities in the Baltic Sea

Organizer: Bo Lövgren
MULTISTATIC SONAR STUDIES IN SHALLOW WATER ENVIRONMENT IN THE GULF OF FINLAND

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Abstract: Reverberation is the major limitation for active sonar performance in shallow water environment. Broadband waveforms with good resolution in range and Doppler do suppress reverberation but their power is limited in shallow confined waters by time-spreading due to multipath propagation. The problem was tried to mitigate by using novel normalization and fusion techniques. Linear Kalman filter was used for tracking multiple simulated targets with variable target strength and velocity. The non-linearity problem related to target doppler information was solved by using separate linear filters for individual target doppler values. These values were then used for data association to update the tracks with the true positions and doppler values obtained from different sensors. The tracking was tested using measured and simulated signals. A three-dimensional model for active sonar was developed for the simulation. The model has a single transmitter and multiple receivers in arbitrary configuration and it accepts multiple underwater and surface targets. The active sonar program creates complex signals for each sensor hydrophone and it includes environmental parameters for water layer, ambient noise, bottom sediment and reverberation. The properties of several broadband waveforms were simulated and verified in multistatic sonar experiments carried out in shallow water environment in the Gulf of Finland. An echo repeater was used in the sea trials to emulate underwater targets. The FM/up-down waveform was superior in detecting static and slowly moving targets with good range accuracy. The Hamming windowed CW was better in detecting weak targets and it performed well even in the presence of strong static targets.

Keywords: Active sonar, multistatic sonar, underwater acoustics, Gulf of Finland
1. INTRODUCTION

The Finnish Navy and its defence industrial partners have been developing a rapidly deployable underwater surveillance system for future underwater threats. The main function of the experimental system SURA is passive detection [1]. The major shortcoming of the passive sonar, however, is limited performance against very silent underwater targets. The performance can be enhanced by integrating active sonar components in the system and operating them in multistatic configuration. The active performance is in turn limited by high reverberation in shallow coastal and archipelago environments. This has been demonstrated in active sonobuoy experiments carried out in the Gulf of Finland in 2003 [2]. Besides the higher reverberation, the shallow Baltic Sea environment has also lower sea water salinity which enables the use of somewhat higher frequencies due to lower chemical absorption in sea water. The hydroacoustic environment of the Gulf of Finland was modelled with a simple sonar model which provided optimized sonar parameters for this particular environment. The optimum frequency range obtained from the modelling was from 5 to 10 kHz with the pulse bandwidth of ca 2 kHz [3].

Reverberation is usually suppressed by using pulse compression where reverberation power is spread over the bandwidth of a broadband pulse, or, by target Doppler where the echo of a moving target is shifted away from the peak of the reverberation spectrum. An optimum waveform should have a good resolution both in range and Doppler, which can be visually verified using the ambiguity function. A severe limitation of using the pulse compression of broadband waveforms in shallow water environment is time-spreading due to multipath propagation.

The loss of detectability in shallow waters is tried to regain by combining normalization, fusion and tracking techniques. Linear Kalman filter is used for tracking multiple simulated targets with variable target strength and velocity. The performance of several broadband waveforms is compared on the basis of simulation and the results from the sea trials in shallow waters of the archipelago. A three-dimensional active sonar model is used to simulate the propagation of sonar signals in shallow water environment.

2. DESCRIPTION OF TEST ENVIRONMENT

The active sonar experiments were carried out on archipelago site where the depths varied from 15 to 20 m. The map of the test site is shown in Fig. 1. Bottom sediment on the test site is post glacial clay accompanied with areas of recent mud. The geoacoustic parameters for these sediments have been previously reported by Poikonen and Madekivi [3]. The tests were performed in June and September 2007. A typical configuration of the transmitter and sensor components in the latter test is depicted in Fig. 1. Sound speed profiles for the June and September conditions are shown in Fig. 2.

3. SONAR WAVEFORMS

Doppler approach is a straightforward way to reduce reverberation in shallow water environment for moving targets. Instead of using long CW pulses for improved Doppler reso-
Fig. 1: Map of the test site. Typical sensor configuration in September 2007 trials: Tx is sonar projector, R_ is receiving hydrophone array and E_ is echo repeater.

Fig. 2: Sound speed profiles on the test site.

<table>
<thead>
<tr>
<th>Waveform</th>
<th>$f_0$ Hz</th>
<th>$B$ Hz</th>
<th># freq.</th>
<th>Duration s</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>cw</td>
<td>8000</td>
<td>3.4</td>
<td>1</td>
<td>0.5</td>
<td>Hamming window</td>
</tr>
<tr>
<td>geoc</td>
<td>8000</td>
<td>800</td>
<td>12</td>
<td>0.4</td>
<td>geometric comb</td>
</tr>
<tr>
<td>gcomb</td>
<td>8000</td>
<td>900</td>
<td>12</td>
<td>0.25</td>
<td>triplet-pair</td>
</tr>
<tr>
<td>FM/up-down</td>
<td>8000</td>
<td>900</td>
<td>sweep</td>
<td>0.5</td>
<td>overlapping</td>
</tr>
</tbody>
</table>

Table 1: Waveforms used in active sonar tests.
olution we tried geometric comb waveforms [4] to obtain good resolution both in Doppler and range. The performance of the waveforms was simulated numerically prior to expensive and time consuming sea trials. Several alternatives were tested in the simulator and the best waveforms were selected for the sea trials. It was demonstrated that simulated performance predicted well the performance of the waveform in real environment. All the waveforms used in the tests are listed in Table 1.

4. ACTIVE SONAR MODEL

Acoustic signals needed in simulations were generated with the active sonar model. The model is three-dimensional in order to get more realistic signals for moving underwater targets and surface vessels. Target tracks are the combination of lines and circular arcs. The targets move at a given speed and direction and they scatter acoustic energy at a given target strength. The surface vessels in the model are passive noise emitters only with no backscattering properties. Constant sound-velocity profile is assumed.

The basic elements of the active mode simulator are depicted in Fig. 3. Underwater targets have omni-directional target strength patterns. All the objects in the model move horizontally meaning that their vertical positions remain unchanged during the model run. The model adds Doppler shift to the pulses reflecting from the targets moving along their tracks.

Reverberation is calculated by dividing the sea floor into small segments where the acoustic pulse is scattered according to the Lambert’s law [6]. The reverberation signal is the convolution of the pressure impulses generated by the transmitted pulse. Static targets, such as rocks and shorelines, can also be added to the model. Frequency dependent chemical absorption in sea water is taken into account in calculating transmission loss. Ambient noise and shipping noise are assumed to be isotropic. The acoustic signals from all sources are summed at each hydrophone using subsample signal delays. The signals are finally converted to baseband with the efficient method described by Lurton [9].
5. FUSION AND TRACKING IN SUPPRESSING REVERBERATION

5.1. Methods used in multistatic processing and target tracking

The signal processing sequence is shown in Fig. 4. The beam normalization was found to be the most important tool to extract weak targets from reverberation and to reduce false detections due to ambient noise. Two efficient normalization techniques were applied in the tests. The first was the SEP2D (separated two dimensional) and the second was the 2DMEAN (two dimensional mean) [10]. Especially the SEP2D technique seemed to effectively suppress the high reverberation in zero Doppler channel, see Figs. 5a – d. The technique is applicable both to narrow band and broad band pulses. The 2DMEAN method is in turn efficient in simultaneous static target tracking.

The third tool for reducing false detections is the sensor fusion and track association. It is based on using the measured radial speeds from Doppler processing only for associating individual sensor detections to target specified tracks and for calculating new target speeds for linear Kalman filter trackers. The association technique is the way to avoid the use of more complex, and sometimes ill-behaving, non-linear tracker which definitely would be needed if radial speeds were used as tracker variables [11]. Besides, the technique reduces false detections due to the stringent radial speed validation. The performance of the waveforms listed in Table 1 was evaluated in numerical simulations and in the sea trials.

![Fig. 4: Multistatic sonar signal processing sequence.](image)

5.2. System description

The projector with SL=185 dB is located in the middle of the configuration at the depth of 8 m. Four circular hydrophone arrays (SURA) are at 10 m depth on the corners of the square of 1200 by 1200 m. The gain of the circular array is 12 dB and the beam side lobe level -10dB. The pulse interval in the test was 6.144 seconds and the targets (TS = 10 dB) were moving at speeds varying from 2 to 3 m/s along trajectories shown in Fig. 5. Bottom backscattering strength of -15 dB, Sea State 2, isotropic ambient noise and light shipping noise were used to specify the environmental conditions in the simulations. The simulated ambient noise spectrum was almost equal to the real one measured at sea. The number of calculated beams and Doppler channels for each sensor data were 12 and 21,
Fig. 5: a) log2-CW pulse beam, b) normalized CW beam, c) 3-D and d) 2-D normalized FM/up-down beam (note the detection cross pattern with radial velocity point maxima). Horizontal axis: Doppler (-5 - 5 m/s), Depth/vertical axis: Range. respectively. The fine adjustment were performed for both values as well as for the range values using the relative side value method.

Fig. 6: Trajectories of 5 moving targets without static (left) and with static targets (+).

Fig. 7: CW pulse with 5 moving targets, static targets (TS=10 dB) added (right).
5.3. Results

The sensor configuration, target trajectories and the locations of 10 static targets (+) in the simulations are illustrated in Fig. 6. The results from the target tracking are shown in Figs. 7-10 so that they can be visually cross-compared. The tracking results in the real sea trial environment are shown in Fig. 11 where simulated moving target data were added to the true reverberation background. The trials configuration is depicted in Fig. 1. Static detections ($v = -0.5 \text{ to } 0.5 \text{ m/s}$) are marked with squares (□).

**Fig 8:** FM/up-down pulse with 5 moving targets, static targets (TS=10 dB) added (right)

**Fig 9:** GCOMB pulse with 5 moving targets, static targets (TS=10 dB) added (right).

The FM/up-down waveform was superior in detecting moving and static targets, Fig. 8. The detection of multiple targets falling in the same range slot is, however, poor due to the limitations in Doppler processing. This causes occasional fading in the tracks in Fig 8. Hamming weighted CW and GCOMB waveforms performed well in the presence of strong static targets. The latter gives better accuracy in locating the static targets.
Fig. 10: 5 moving targets \((TS = -3 \ldots +6dB)\) with 10 static targets, CW pulse (left), FM/up-down pulse (right).

Fig. 11: GCOMB pulse (left) with 5 moving targets \((TS = -3 \ldots +6dB)\) and 10 static targets, CW pulse with the sea trials data (right): \(TS = 10dB, v = 2\ m/s\), triangle denotes receding target \((v > 0.5\ m/s)\).

6. CONCLUSION

Reverberation is the major limitation for active sonar performance in shallow water environment. Broadband waveforms with good resolution in range and Doppler do suppress reverberation but their power is limited in shallow confined waters by time-spreading due to multipath propagation. The problem was tried to mitigate by using novel normalization and fusion techniques. The normalization turned out to be efficient in handling irregular envelopes of long pulses and in reducing false detections which both are the consequences of strong reverberation.

Linear Kalman filter was used for tracking multiple simulated targets with variable target strength and velocity. The non-linearity problem related to target Doppler information was solved by using separate linear filters for individual target Doppler values. These values were
then used for data association to update the tracks with the true positions and Doppler values obtained from different sensors. The performance of tracking was tested with several waveforms. The FM/up-down waveform was superior in detecting static and slowly moving targets with good range accuracy. The Hamming windowed CW was better in detecting weak targets and it performed well even in the presence of strong static targets.

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MEASUREMENTS AND SIMULATIONS OF ACOUSTIC PROPAGATION LOSS IN THE BALTIC SEA

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Abstract: Between 2002 and 2007 FWG has carried out four sea trials at different locations of the western Baltic Sea. These were BAROC 2002 near Landsort and Gotland and UCAC 2006, VOICE 2006 and UCAC 2007 east of Bornholm. Altogether quite a large number of runs with different configurations of source and receiver and different environmental conditions were analysed. Important processing outputs are transmission loss (TL) and channel impulse response as function of acoustic propagation distance. The data add to FWG's existing transmission loss data base which is largely based on experiments performed before 1990. Improvements of the experimental equipment as well as of the processing techniques allowed us to probe the acoustic channel more thoroughly than previously possible. This provides new opportunities to validate but also to pinpoint shortcomings of existing sound propagation models. In particular, our focus is on a possible improvement of the stochastic ray tracer SIPSI / MOCASSIN developed by FWG. In this paper we give an overview of the sea trials and compare measured and modelled transmission loss for a few situations.

Keywords: propagation loss, transmission loss, Baltic Sea, SIPSI, MOCASSIN
1. INTRODUCTION

Before 1990 FWG (Forschungsanstalt der Bundeswehr für Wasserschall und Geophysik) had conducted a long term programme on measuring sound propagation in the Baltic. The motivation for this programme is described in [1]. The standardised experiments used air-dropped explosives as sound source and yielded propagation loss (TL) in one-third octave bands using spectral analysis procedures. The vertical resolution, that is the hydrophone spacing, was about 10m and the horizontal spacing about 1 nautical mile. The data were applied to the validation of a range dependent acoustic propagation model which has become the core, called SIPSI, of the German Navy's sonar performance model MOCASSIN. A key feature of SIPSI is stochastic ray tracing. This method uses a Monte Carlo approach for the stochastic change of ray directions and was found to be a very effective way of accommodating the acoustic forward scattering which is caused by variations of the sound speed in the water column [1]. In line with the incoherent transmission loss data from the experiments, with the model's stochastic approach and with operational conditions and needs, the model calculates incoherent TL only and requires only one sound speed profile to be specified.

Using new equipment and new processing techniques FWG resumed TL measurements in 2001. In the Baltic Sea four sea trials have been carried out so far (Fig. 1). These were...
BAROC in May 2002 near the Swedish coast at Landsort and near Gotland [2, 3, 4, 5, 6], UCAC06 in August 2006, UCAC07 in August 2007 near Bornholm [8, 10] and VOICE06 in November 06 also near Bornholm [7]. The objectives of these sea trials were not identical. In fact, for UCAC obtaining TL was more a by-product rather than the primary goal as the main motivation of these trials was to develop and test underwater communication methods. Nevertheless, some portions of the trials were devoted to the measurement of TL and these used the same acoustic equipment (with the exception of VOICE06), followed the same measurement procedures and employed identical or at least comparable processing techniques. Here we will describe only the salient features of the TL experiments. Details may be found in [2, 3, 4, 5, 6].

Fig. 2: a) Towed transducer frame used during the UCAC trials with two omni-directional transducers for different ranges of frequency. b) Spectrogram of the sequence of twenty one second duration LFMs as used for the BAROC experiment. c) Schematic of the receiver systems deployed from the anchored WFS Planet during the UCAC sea trials. A NESSY hydrophone chain consisting of up to eight omni-directional hydrophones with a vertical spacing of 10m and/or the VAIII, a nested, vertical array of 128 hydrophones distributed over a length of 38m, were used. d) Schematic of the fixed range experiment VOICE06 with stationary source and receiver towers.

For the BAROC and UCAC trials one vessel was anchored deploying a NESSY hydrophone chain or the Vertical Array III (VAIII) or even both (Fig. 2c). A second vessel towed a frame holding two (UCAC) or three (BAROC) transducers. Towing was along a straight line away from the receiving vessel (outbound Run) or back on the same line towards it (inbound Run). Track length varied between 20km to 60km. For the TL measurements a sequence of about twenty “narrow band” LFMs (bandwidths were 200Hz or 400Hz) plus up to four “broad-band” LFMs (bandwidth of 3500Hz) was transmitted at intervals varying between 60 and 180 seconds. With a tow speed of about 5 knots this translates into a horizontal resolution of 150m to 450m. The received signals were analysed using matched-filter techniques which yielded both impulse responses and signal levels as function of distance between source and receiver and as a function of 20 centre frequencies between about 400Hz and 5000Hz. It was investigated whether the new TL data are comparable with the ones obtained by the previous method [1]. Differences do exist. However, being understood they cause little problems.

One issue with the TL versus distance data obtained using a towed sound source is that, because of the slow tow speed compared to the speed of sound, the effects of spatial and...
temporal variability cannot be separated. This difficulty becomes particularly troublesome in areas with high spatial variability of the environment and/or whenever a high sea-state caused movement of the transducer. Although this issue was recognized right from the beginning, tight cruise schedules and logistics allowed to perform only a few measurements with a stationary or drifting sound source (e.g. [5] for BAROC or Fig. 6 below). The trial VOICE06 is special as it was entirely focused on the temporal variability and therefore used completely stationary systems which were deployed on the sea-floor (Fig. 2d). It furthermore used a smaller number of LFMs [7].

It should be understood that if one wants to go beyond just measuring the temporal variability of acoustic propagation, that is if one wants to relate the acoustic to the environmental variability, one is forced to put significant effort in doing environmental measurements. For BAROC and VOICE06 extra vessels were tasked with accessing the environment and lots of extra environmental measurements were performed and data gathered [2, 3, 4, 7]. For the UCAC trials this was not possible.

Compared to the older TL measurements the new sea trials improved the quality of the data in many ways: the horizontal and vertical (for the VAIII) resolution is much improved, TL data are supplemented by impulse response functions and the amount of supplementing environmental data has increased greatly. This helps to discover and explain effects which previously were not accessible, and it may discover deficiencies in the modelling that can no longer be 'explained' with measurement uncertainty and/or lack of environmental input.

2. MEASUREMENTS AND SIMULATIONS

Of the four sea trials BAROC has so far been analysed most [2, 3, 4, 5, 6, 9]. The sound speed field at the Landsort site near the Swedish coast showed a well developed sound channel with strong spatial variability as consequence of a series of cold and warm water fronts [Fig. 3]. At the Gotland site the variability was less pronounced, but still significant, particularly at the depth of the sound channel (see [4, 9]). We investigated the effect of the spatial variability by analysing measured TL curves from four Runs at Landsort and two Runs at Gotland with different receiver and source depths and for frequencies of 400Hz, 1300Hz and 5000Hz. Altogether 144 measured TL curves were used. Simulations were performed with SIPSI and with MOCMULTI which is an extended version of SIPSI. MOCMULTI was developed by FOI to do 2D and 3D propagation modelling in the Baltic [6].

When forward scattering is the important process at work, then there is no difference between the two models. This is the case, for instance, when the sound source is located inside and the receiver above the sound channel (or vice versa). When the sound speed field has a strong deterministic component, MOCMULTI has got advantages over SIPSI. Fig. 3 shows such a case and another is described in [9].

We now turn to the UCAC sea trials. UCAC06 used the VAIII chain as receiver ranging from about 15m to 53m and UCAC07 the NESSY chain ranging from 10m to 60m. Other parameters as the transmitter configuration, calibration settings, the position of the receivers, the tracks and the LFMs were identical. We did not have a CTD-chain but carried out repeated CTD-casts from the receiving ship and also a few CTD's and XBTs along the tow track. For both trials the observed oceanographic conditions were typical for Baltic Sea summer conditions, but they were not identical (Fig. 4).
During UCAC06 there was a very sharp thermocline and the sound speed minimum was about 1427 m/s at a depth of 30m to 35m. During UCAC07 the sound speed minimum was only about 1435 m/s, it was about 10m deeper, and the thermocline change was more gradual. Compared to BAROC the observed variability of the sound speed profiles was small.

The TL curves are as expected with a large TL for the hydrophones in the surface layer and a much lower TL for those inside the sound channel (Fig. 4). Furthermore the UCAC06 and UCAC07 TL curves are quite similar. We note quite a bit of variability on the TL curves. One could think the movement of the transducer frame induced by the movement of the towing vessel to be a possible explanation. However, the same amount of variability is also observed for a nearly stationary vessel (Fig. 5). A similar observation is described in [5] for the BAROC experiment. We may therefore assume that the variability is caused by the natural variability of the sound speed profiles.

We performed simulations with SIPSI for several runs of UCAC06 and UCAC07 using four different frequencies. Since we did not have a 2D sound speed field we did not use MOCMULTI except to test whether a range-dependent sound speed field, which we derived from the CTD profiles and one or two XBT casts done along the track, would have a significant effect. It did not. The modelled TL curves for Run A05 (UCAC06) show good agreement with the measurements for 3400Hz (Fig. 5) and 5400Hz. We emphasize that a deterministic model would not be able to correctly predict the TL outside the sound channel with only one sound-speed profile given as input. SIPSI performs here very well. For lower frequencies (900Hz and 1300Hz, not shown here) the modelled TL was too small for
distances larger than 30km. Diffraction effects not included in the ray modelling might explain this (see also [6]).

Fig. 4: Top: Typical temperature, salinity, density and sound-speed profiles for UCAC06 (left) and UCAC07 (right). Bottom: TL curves for a centre frequency of 3400Hz. The source was towed at the centre of the sound channel [10]. The tow tracks of A05 (UCAC06) and A28 (UCAC07) are identical.

Surprisingly we were not able to achieve the same level of agreement for Run A28 of UCAC07 (Fig. 6) although we varied many parameters (sea-floor, sound speed profile, transducer depth, receiver depth, matched filter procedure). A calibration problem can be ruled out. One parameter which, for some runs, can be used to improve the agreement is the
value of SIPSIs's diffusion constant which determines the amount of forward scattering. However, we were not able to find one single optimum value which would improve the agreement for all combinations of source depth, receiver depth and frequency. It was already argued in [6] that a solution might be to extend SIPSIs's ray diffusion approach for situations with large thermocline steepness [11]. Furthermore, diffusion might be frequency dependent. Consequently, we might end up with a diffusion function rather than a constant.

![Graph](image)

Fig. 6: Measured (black) and modelled (red, blue, and cyan) TL curves for two hydrophone depths, one above and the other inside the sound channel. The centre frequency was 3400 Hz. Modelled curves are shown for deterministic mode and two different diffusion constants. Smaller values mean less forward scattering. The deterministic mode is without scattering. A value of 10e-7 is recommended for the Baltic.

3. SUMMARY

We have given an overview of four relatively recent transmission loss experiments that were carried out by FWG and partner institutes between 2002 and 2007 in the Baltic Sea. In contrast to FWG's earlier transmission loss experiments [1] the main processing method has been the matched filter method. The acoustic channel could be probed in greater detail and the variety of different measurement configurations with respect to source and receiver depths, frequency, and (summer) environmental conditions allows a comprehensive test of a sound propagation model. The comparison with SIPSIs yields a mixed bag of good, satisfactory and some unsatisfactory agreements. The last may, for example, be caused by an unsatisfactory representation of the environment or by the neglect of diffraction effects. The first problem is operationally unavoidable and the second inherent to the ray approach. Nevertheless, some improvement might also be possible by modifying SIPSIs's ray diffusion approach in cases with high gradient thermoclines [6, 11]. The charm of this approach would
be that the amount of environmental input to SIPS1 and hence also to the operational model MOCASSIN would not increase.

4. ACKNOWLEDGEMENTS

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REFERENCES


HIGH FREQUENCY SONAR PERFORMANCE IN THE STOCKHOLM ARCHIPELAGO

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Abstract: Surveillance with sonar in the Stockholm Archipelago is indeed a difficult task. The Archipelago has a crystalline basement, transacted by fracture zones and lineaments. As a result, the underwater environment is characterized by rapidly varying bottom topography, with seamounts, outcrop and sediment filled valleys which may contain gas. This influences heavily the performance of surveillance sonar, and great care must be taken when designing sonar for this environment. The Royal Swedish Navy is currently procuring a new helicopter which will be fitted with a dipping sonar. This has been tested in the Stockholm Archipelago, with good results. Calibrated targets were used, together with extra hydrophones for the monitoring of the sound field incident on the target.

As part of the test we have conducted 2D and 3D ray modelling of the transmission loss and the reverberation. The modelling has been performed with a new model, REV3D, devised for fast computation in connection with real-time inversion. REV3D avoids the demanding 3D eigenray problem by computing intensities as probe volume averages using the ray density. A particular feature is that time series envelopes can be simulated by positioning ray intensities along a time axis, according to traveltime and intensities for each ray.

The good agreement between measurements and modelling gives us confidence that our method for assessment of sonar performance works well and can be used for formal acceptance tests.

Keywords: Sonar, ray modelling, transmission loss, reverberation.
1. INTRODUCTION

1.1. Acoustic surveillance in the Baltic

As has been reported at previous UAM conferences, operating active sonar in the Baltic is difficult due to complex bottom conditions. The Baltic is very shallow, often with hard bottoms, which causes high levels of reverberation in active sonar. In addition, variations in the sound speed profile both with time and position can sometimes make it very difficult to operate active sonar in the optimum way.

A way to defeat high levels of reverberation is to use large bandwidth and narrow lobes. Fortunately, the Baltic has a very low salinity, typically 7 PSU, which means that active sonar can be operated at higher frequencies than usual. A higher operational frequency gives the advantage that for a given physical size of the array the lobes will be narrower. It is also easier to achieve a large bandwidth, if the operational frequency is high. Hence the Royal Swedish Navy (RSwN) uses active sonar with typically two to five times higher frequency than most other nations. As a consequence, RSwN sonars can not be bought “off-the-shelf” and have to be specially made.

1.2. The new helicopter sonar

The RSwN is presently acquiring a new helicopter, which will have Anti Submarine Warfare (ASW) capability, including a dipping sonar [1]. This sonar has been built according to specifications from the Swedish Defence Procurement Office (FMV). In this specification, it is described how the sonar shall perform in certain idealized conditions, such as isovelocity, flat sand bottoms with constant depth. Such an environment is useful for sonar performance calculations, in order to check different design alternatives. However, as it is impossible to find this environment in the real world, it will be difficult to perform acceptance tests. Therefore we have designed a procedure which uses a combination of measurements at sea and performance predictions. In short, this procedure is as follows:

- Measurements of environmental and underwater acoustical conditions (sound speed profile, bathymetry, weather, transmission loss, reverberation etc.)
- Performance predictions tuned to the experimentally obtained values.
- Calculation of detection distances in this environment.
- New measurements at the predicted detection distances to verify that the required performance is obtained.

In the autumn of 2007 a combined team from the manufacturer, FMV and FOI performed a very successful sea acceptance test according to these principles. In this paper we give some details from this test, with focus on the acoustic measurements and calculations.
2. MEASUREMENTS

2.1. Experiment setup

An important part of the experiment was to measure all relevant parameters as accurately as possible. The sonar was deployed from R/S Ocean Surveyor, and operated via the normal sonar handling system which is used onboard the helicopter, including a winch. Hence the depth of the sonar dome could be accurately adjusted.

The target was deployed at a depth of 76 m, 19 m above the bottom. In front of the target, at a distance of 24 m, we placed a vertical hydrophone chain with 5 hydrophones. This chain was connected to a recording system, with software for signal analysis. In this way we could monitor the transmitted signals and the target echo in real time.

![Fig. 1. The experiment setup with transmitter ship, recording ship, target and hydrophone chain.](image)

2.2. Environment

The experiment took place in Kanholms Bay in the Stockholm Archipelago, which is characterized by a rapidly varying bathymetry with a maximum depth of 100 m (Fig. 2, left). The sea bottom varies from soft sediments (postglacial clay) in the deeper parts to hard outcrop in the shallow areas.

The hydrography is typical for the time of the year (October). The top layers, which in the summer are filled with warm water, are in the autumn cooled and gradually mixed with deeper, colder water until isovelocity is reached in the end of December. Hence the October sound speed profile has a positive (upward refracting) sound speed gradient, in our case down to around 60 m depth. Below this layer the gradient is negative (downward refracting), and a small sound channel is formed near the bottom (Fig. 2, right). The result is a complicated variation of sound speed with depth, which strongly influences the sound propagation.
2.3. Calibration experiment

A new target was specially made for this experiment. It is in the form of a double cone, which has constant target strength (TS) independent of aspect angle. This is important since exact knowledge of the TS is crucial for the verification of the performance of the sonar. By using the setup as depicted in Fig. 1 we can obtain the TS by comparing the strength of the direct pulse with the strength of the echo from the target, after correction for transmission loss. Our result is that the TS of the double cone is $0 \pm 1$ dB.

2.4. Transmission loss and reverberation measurements

The transmission loss (TL) measurements were done by using the sonar as a transmitter, and the hydrophone chain as receivers. In all, 9 different transmission positions were used. The transmitting ship was anchored, and engines shut off, in order to avoid disturbing sounds. At most positions the sonar dome was placed at the most favourable depth, as determined by the sound speed profile. In a few positions the transmitter depth was varied in steps from top to bottom, and the effect of the complicated sound speed profile was clearly revealed.

Reverberation levels (RL) were measured by using the sonar as both transmitter and receiver. The recording system in the sonar was used to store the received signals after match filtering and beamforming.

In the measurements all available pulse shapes were tested, but for the performance measurements we preferred a hyperbolic frequency broad-band pulse, with bandwidth more than 4 kHz and centre frequency at about 20 kHz. The pulse length was 128 ms.
3. MODELLING

REV3D is a ray-tracing program for modelling transmission loss and reverberation in 3D environments. It avoids computing eigenrays by following an approach described by Piskarev [2]. The emitted energy from the source is viewed as confined to the infinitely thin rays. Intensities are determined using the ray density, and the dynamic ray-tracing problem need not be solved. The stochastic 2D ray-tracing model MOCASSIN [3] is based on similar ideas. It provided a useful point of departure for developing REV3D [4], although the stochastic parts have not been kept.

Three horizontal grids are used in REV3D. The first is a rectangular grid for specifying the sound speed in the water and the bathymetry. The sound speed is represented by range-independent profiles in the horizontal grid rectangles. In each rectangle, the variation of the sound speed $c$ with depth is assumed to be of the “$1/c^2$ linear” type. Hence, each ray is built up as a sequence of parabolic arcs and the ray tracing becomes highly efficient computationally. 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 quickly calculated by solving second-degree algebraic equations. Although no ray tracing is performed explicitly through the bottom, the subbottom can be taken into account by computing plane-wave reflection coefficients of stacks of fluid and/or solid layers [5]. Alternatively, reflection coefficients based on bottom porosity types [3] can be used.

The second and third horizontal grids are polar grids centred at the source (the “TL grid”) and at the receiver (the “RL grid”), respectively. Together with a division of the depth axis, the TL grid provides 3D box volumes for which average intensities are computed. Each ray contributes to the average for a particular 3D box according to the energy it carries and its arc-length within that box, and transmission loss curves can easily be produced. The RL grid provides bottom and surface patches, and reverberation time traces are formed by summing intensity contributions from each patch. It should be emphasized that Lloyd mirror and coherency effects are not modelled.

Being a ray-trace model, REV3D is best adapted to high-frequency computations. The applicability to low frequencies is improved by a frequency-dependent smoothing of the sound-speed profiles [3]. For example, penetration by pulses with large wavelengths is facilitated by smoothing out large sound speed gradients.

REV3D can also be used for computing transmission time series for 3D environments. Keeping track of the traveltimes for the different rays, the intensities from a source pulse can be positioned in time to obtain time traces for pressure magnitude averages in the different 3D boxes of the TL grid. An example is included in [6].

It is very convenient to do the modelling with a bathymetry data file for the area in question. Upon specifying source and receiver coordinates, and horizontal beam directions, REV3D selects the relevant depth values from the file. As alternatives to full 3D modelling, Nx2D and 2D options also exist, for which horizontal refraction is neglected.
4. RESULTS

4.1. Transmission loss and time series

Fig. 3 depicts measured and modelled TL as functions of transmitter depth. The results of the modelling agree quite well with the experimental results. The sound channel at 80 m depth has a significant influence on the TL. Please note that a small change in transmitter depth from 70 to 75 m results in a 12 dB difference in the TL.

Both measured and modelled time series (not depicted) show a double peak at short distances, where the second peak comes from rays reflected in the bottom. This second peak is sometimes stronger than the direct pulse, which can be understood by considering ray diagrams such as the one in Fig. 5, left. At longer distances the double pulse vanishes, i.e. there is only one strong arrival.

![Fig. 3: Experimental (left) and modelled (right) transmission loss as functions of transmitter depth for the chain with 5 hydrophones separated 1.5 m.](image)

4.2. Reverberation

As with the TL measurements, there are large variations in RL for different depths. In Fig. 4, left, recordings from the same beam, but at different depths, are depicted with time converted to one-way distances. At 2200 m distance the difference in RL is 40 dB, which is a good illustration of the difficulties a sonar operator may encounter in the Baltic.

The 40 dB difference can be understood if we consider Fig. 5, showing ray traces for the two depths (70 and 79 m). At 70 m most of the sound is downward refracted, causing high RL at short distances, and then a normal RL fall-off with distance. The 79 m case is quite different, as the sonar is then within the small sound channel close to the bottom. The result is low TL and the sound travels all the way to the harder, shallower parts of the experiment area (Fig. 5, right). Hence, very high RL is recorded from these parts.

The fall-off with time of the reverberation is typically controlled by the reflection coefficient at the bottom. The bottom in the area consists of soft postglacial clay, and an appropriate reflection-coefficient curve was selected for the modelling (porosity type 1
according to [3]). Indeed, the fall-offs with time (or range) for the measured and modelled 70 m reverberation traces (Fig. 4, blue) are very similar. The fit could of course be improved by formal inversion.

The level of the reverberation, on the other hand, is controlled by the back-scattering strength as specified by Lambert’s law, for example. A common objection to Lambert’s law is that it gives too small reverberation at low angles. For the computations shown, a variant was used with σ, the differential back-scattering cross section per unit area, given by

$$\sigma = \sqrt{\sin \Theta_i \sin \Theta_s}$$  \hspace{1cm} (1)

where Θ_i and Θ_s are the grazing angles for incident and back-scattered waves, respectively. Indeed, the square roots of the sines enhance reverberation at low angles, and it appears that this variant provides better agreement to the experimental data than the classical Lambert’s law. Physically, the additional back-scattering may be caused by volume reverberation from the sediment.

\[Fig. 4: \text{Experimental (left) and modelled (right) reverberation levels for two transmitter depths, 70 m (blue) and 79 m (red).}\]

\[Fig. 5: \text{Ray traces with transmitter at 70 m depth (left) and 79 m (right).}\]
5. CONCLUDING REMARKS

By tuning a 3D sonar performance prediction model we have been able to model transmission loss, reverberation and time series with good precision. Concerning reverberation measurements in areas with rapidly varying bottom topography, the variations between different bearing directions within the source and receiver horizontal lobes become important. Computations restricted to a single central bearing can be misleading. With a bathymetry data file for the area, REV3D conveniently includes a multitude of bearings to produce full 3D or Nx2D results. In the examples shown, horizontal refraction turned out to be small, and the 3D and Nx2D results were similar. Since differences in computation times are small, however, full 3D calculations can just as well be done.

In the sonar acceptance test, our model was used to calculate a theoretical detection distance of 5 km for the 0 dB target. And indeed, when this was tested by pinging at this distance, the mean signal-to-noise ratio was 17 dB, well above the value required in the specifications. Hence we find that the sonar did pass the acceptance test with a good margin.

6. ACKNOWLEDGEMENTS

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REFERENCES

Structured Session 19

Innovative Technologies and Techniques for Seafloor and Sea Surface Characterization

Organizer: Anthony P. Lyons
STATISTICAL PROPERTIES OF BACKSCATTER FROM SEAGRASS COLLECTED FROM A SIDESCAN SONAR SURVEY

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Abstract: As part of a sidescan sonar survey of a shallow water estuary, backscatter collected from seagrass was compared with uncovered areas to investigate the possibility of potential improvements to seafloor mapping. A high-frequency (250/550 kHz) EdgeTech 4200 sidescan sonar system was used in a survey of parts of the Swan-Canning estuary in Perth, Western Australia. Backscatter data were corrected for transmission loss and insonification area to derive the backscatter strength. The statistical distribution of backscatter intensity data collected from seagrass and sand were compared with Rayleigh and non-Rayleigh models. The K-distribution was found to be the best-fit model for variations of backscatter from both sand and seagrass. After processing, which includes averaging, the data no longer fit the moments of a K-distribution, but can be modelled by a gamma distribution. This is consistent with previously observed results and theoretical interpretation. However, in some instances, a log-normal distribution model fits the experimental data better than the gamma distribution.

Keywords: High-frequency backscatter, intensity statistics, seagrass, sidescan sonar
1. INTRODUCTION

This study investigated the statistics of acoustic backscatter collected over seagrass and bare sediment, which was constituted primarily of sand, using an EdgeTech 4200 MP sidescan sonar (SSS) system operating at 250 and 550 kHz. For a Gaussian scattering process, the distribution of instantaneous backscatter amplitudes (envelopes) is Rayleigh and, subsequently, the backscatter intensities are exponentially distributed. However, it has been demonstrated both theoretically [1] and experimentally [2] that, under certain conditions, the statistics of acoustic backscatter from the seafloor can be essentially non-Gaussian, when fluctuations of the backscatter envelope are not Rayleigh distributed. Recent studies have noted a non-Rayleigh character of backscatter statistics at high acoustic frequencies and suggested different models. One such model is the $K$-distribution, which has been shown to fit the distribution of backscatter fluctuations for certain types of the seafloor [2, 3]. The theoretical basis for the $K$-distribution approximating variations of the backscatter amplitude and intensity has been considered in [1] and [4] respectively. In this study, the distributions of backscatter fluctuations observed over seagrass and sand at 250 kHz and 550 kHz are compared with the exponential and $K$-distribution models to investigate the validity of these models at high frequencies.

The SSS data were collected as part of a habitat mapping project [5], which involved creating backscatter mosaics. For producing backscatter mosaics, backscatter data are gridded, which usually involves averaging the raw data within the grid cell and provides an improvement of the signal-to-noise ratio. Fluctuations of the average backscatter intensity were demonstrated theoretically in [4] to tend to be gamma distributed. This has also been found experimentally in the case with averaged backscatter intensity collected with a multibeam sonar system [6]. The SSS dataset used for this study provided a useful insight into the effect of averaging on backscatter intensity statistics, which has implications for habitat mapping.

2. METHODS

The tow-fish of the EdgeTech SSS was fixed mounted to the vessel to improve the navigation accuracy of backscatter measurements. Backscatter data were collected to a maximum range of 75 m either side of the tow-fish over sand and seagrass covered areas in the Swan River in Perth, Western Australia. The depth of the seafloor over the SSS transect was varying within 4-6 m.

Backscatter data were recorded in EdgeTech’s native format (.JSF). A program written in Matlab was used to read this into the Matlab environment and correct backscatter for transmission loss and insonification area (assuming a flat bottom). An additional routine was developed to produce the backscatter mosaics.

Backscatter intensity data used in the statistical analysis were resampled at a 5-time lower rate to exclude correlated samples received from the same insonification area on the seafloor. When calculating the probability false alarm $PFA = 1 - CDF$ (cumulative distribution function), the intensity backscatter data were normalised by the mean value.
3. RESULTS

The mean backscatter strength from sand and seagrass at 250 kHz and 550 kHz is shown in Fig 1 as a function of slant range to the bottom. A noticeable increase in backscatter strength with slant range at the higher frequency is likely due to inadequate correction for the insonification area at small grazing angles made for a flat-bottom model. At the lower frequency, the effective width of the transmitted pulse is nearly 3 times longer, so that errors in the insonification area estimates due to the actual bottom slope are not as critical as those at the higher frequency.

![Graph showing backscatter strength versus slant range for sand and seagrass at 250 and 550 kHz. The backscatter strength is shown in dB relative to an arbitrary but constant value, which is not corrected for unknown total gain of the sonar receive system.](image)

*Fig. 1: Backscatter strength versus slant range for sand and seagrass at 250 and 550 kHz. The backscatter strength is shown in dB relative to an arbitrary but constant value, which is not corrected for unknown total gain of the sonar receive system.*

The coefficient of variation (CV) of the backscatter intensity versus incidence angle from sand and seagrass is shown in Fig 2. For an exponential distribution of fluctuations the CV equals unity. The CV for both sand and seagrass at both frequencies remains close to unity between 45 to 75°, which is characteristic for a Rayleigh-like process. As incidence angle increases beyond 75° the CV becomes greater than unity, which indicates a deviation from a Rayleigh-like process. This deviation in CV is greatest in the 550-kHz backscatter data. Reasons for this change could relate to changes in the insonification area. The insonification area versus incidence angle is shown in Fig 3 for the two different seabeds. At large incidence angles (small grazing angles), the transverse width of the insonification area is as small as approximately 3 cm at 550 kHz and small variations in the seafloor slope could lead to considerable fluctuations of the insonification area, which might result in much larger fluctuations of the backscatter intensity.
Fig. 2: Coefficient of variation versus incidence angle for sand and seagrass at 250 and 550 kHz.

Fig. 3: Insonification area ($m^2$) versus incidence angle (deg.).

The PFA of normalised backscatter intensity collected from sand and seagrass at 250 and 550 kHz between 60 and 70° is shown in Fig. 4. Most of the data in Fig. 4 can be reasonably well approximated by the exponential model with the data only deviating at the tail. The deviation from the exponential model is more significant for backscatter collected at 550 kHz.
kHz. The $K$-distribution provides a better approximation for the tail of the data, although it is not perfect. For backscatter collected at greater incidence angles of 80 - 90° (Fig. 5), the approximation by exponential model is noticeably worse than that at smaller angles. In this angular domain, the $K$-distribution also provides a better approximation for the tail of the data than the exponential model.

The effect of averaging backscatter intensity on its distribution is shown in Fig. 6, where the PFA of backscatter intensity averaged over 50 samples collected over sand and seagrass at 250 and 550 kHz at 60 to 70° (left panels) and 80 to 90° (right panels) is compared with the exponential, gamma, log-normal and $K$-distribution models.

It was found that the more data samples were averaged over, the more the resulting distributions moved away from the exponential model. The averaged backscatter intensity collected between 60 and 70° was, in general, found to be approximated best by either the gamma or log-normal distributions. For larger angles of 80 - 90°, the best approximation models for backscatter fluctuations can be either the log-normal or $K$-distributions. While there is a theoretical premise for the gamma and $K$-distributions to fit the averaged intensity, no such reasons for the log-normal model are evident. However, the log-normal distribution has been found experimentally to fit a variety of backscatter data [6-9]. Trevorrow [9] observed a variation of backscatter distributions between the log-normal and Rayleigh distribution models, whereas Stanic and Kennedy [8] observed high-frequency shallow-water backscatter variations obeying the Gaussian distribution model at large grazing angles and the log-normal distribution at small grazing angles. Chotiros et al. [7] showed that high-frequency seafloor backscattering could depart from a Rayleigh distribution depending on the beamwidth. The wide-beam seafloor backscattering followed the Rayleigh distribution whereas the narrow-beam seafloor backscattering obeyed the log-normal distribution. Gavrilov and Parnum [6] found the gamma and log-normal distributions to approximate well the average backscatter intensity collected with multibeam sonar over a variety of seafloor habitats.

![Graph](image-url)

**Fig. 4:** The probability false alarm for backscatter intensity collected over sand and seagrass at 250 and 550 kHz for incidence angles between 60° to 70°.
Fig. 5: The probability false alarm for backscatter intensity collected over sand and seagrass at 250 and 550 kHz for incidence angles between 80° to 90°.
Fig 6: The probability of false alarm for backscatter intensity averaged over 50 samples collected over sand and seagrass at 250 and 550 kHz for incidence angles between 80° and 90°: data (-), exponential (--), gamma (---), log-normal (-.-) and K (--) distributions. The K-distribution is not shown, where its parameters cannot be estimated based on the method of moments.
4. CONCLUSIONS

Fluctuations of backscatter intensity collected over sand and seagrass at 250 kHz and 550 kHz using a sidescan sonar were found to be reasonably well approximated by an exponential distribution, when the grazing angle of backscatter is not small. This means that backscatter fluctuations can be modelled as a Rayleigh-like process. For small grazing angles, when the insonification area becomes strongly dependent on the local slope of the seafloor, the distribution of backscatter fluctuations deviate noticeably from a Rayleigh model and can be approximated much better by the $K$ - distribution. When backscatter intensity is averaged, the gamma distribution model provides a good approximation for the majority of incidence angles, except small grazing angles where the $K$ - distribution and log-normal distribution are much more accurate models for backscatter fluctuations.

REFERENCES


MEASUREMENTS AND MODELLING OF HIGH-FREQUENCY ACOUSTIC SCATTERING BY A ROUGH SEAFLoor AND SEA SURFACE

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Abstract: Acoustic scattering by a rough, possibly dynamic interface is experimentally studied by insonifying the seabed and the sea surface at high frequency at various incident angles. A directional source working at 300 kHz was placed at the top of a 3.5 m high tower deployed on the seabed. A vertical array of 3 omnidirectional hydrophones was suspended from a portable frame, which was deployed in bistatic configuration at a variable range between 30 and 70 m. A selection of the results is presented to evaluate the sea surface and seabed scattering amplitude in the nominal specular reflection direction. Scattering by the sea surface was measured during relatively long periods of time in order to correlate its value with the sea state. Model–data comparison was conducted between the scattering data and a time-domain, three-dimensional rough surface scattering model (BORIS-SSA). Model-based analysis allows for a better understanding of some aspects of high-frequency multipath reverberation in shallow water.

Keywords: Environmental Acoustics, High-Frequency Scattering from Rough Boundaries
1. INTRODUCTION

Multipath reverberation is recognized to be one of major factors that may degrade the performance of new-generation sonar systems, such as high frequency (>100 kHz), very-high-resolution SAS sonars, which are recently operated mainly in shallow water, where environmental constraints (in particular, sound interaction with a rough and dynamic sea surface) are much more severe than in deep water. The impact of multipaths on sonar imaging quality is not fully understood at the present. The main features that a ray propagation model based on pure specular reflection cannot predict are angle and time spread effects, which have been observed to characterize all the multipath arrivals that involve one or more “specular” bounces either on the sea surface or on the seafloor [1,2].

High frequency acoustic forward scattering measurements from rough seafloor and sea surface are presented in this work. The data were collected during the MARES’08 sea trial off the north coast of Elba Island, Italy, in March 2008. A 300 kHz bottom-mounted system was used, which is capable of isolating and measuring ray paths from single interactions with either the seafloor or the sea surface in bistatic configuration.

In the case of sea surface scattering, variability with time was measured through high-ping-rate, long-term monitoring. By pointing the sonar to the sea surface at normal incidence in monostatic configuration, it was also possible to use the same system to measure the wave height at a high ping rate and with the spatial resolution of about 6 cm. This resolution is substantially better compared to what can be obtained by a wave rider buoy or a static pressure sensor, and is only one order of magnitude larger than the sonar wavelength (equal to 5 mm). Wave height measurements were alternated to scattering measurements of the sea surface in order to be able to correlate the two datasets.

The acoustic datasets were correlated to environmental ground truth measurements, such as bathymetric swath measurements, geophysical measurements of the seabed, sea surface and seafloor large-scale (waves) and small-scale (surface ripples) measurements, CTD (conductivity-temperature-depth), wind and sea currents.

Model–data comparison was conducted between the scattering data and a time-domain, three-dimensional rough surface scattering model (BORIS-SSA). Model-based analysis allows for a better understanding of some aspects of high-frequency multipath reverberation in shallow water and is a prerequisite to high-frequency acoustic propagation and reverberation modeling in coastal waters. The data acquired have been used for model-data comparison against simulations obtained with the BORIS-SSA modelling tool. BORIS-SSA is a full 3-D time-domain model that simulates the echo received by an interface of known roughness and reflection coefficient, given the source pulse and the beam pattern and position of transmitter and receiver [3-5]. The surface scattering component is calculated using either the Kirchhoff approximation or a second/fourth order small slope approximation (SSA).

2. EXPERIMENTAL SET-UP

The experimental set-up was designed to collect acoustic scattering measurements from rough static and dynamic surfaces.

The Biodola bay at Elba Island is characterized by a wide area of relatively flat, homogeneous, fine sandy seabed. Sand ripples are ~3-4 cm high (peak to peak), have spatial period around 20-25 cm, and are mainly oriented with a heading of about 130° North, and with an angle of about 20° with respect to the transmission axis of the sonar. These values
were provided by divers. The sediment was classified as very fine sand, with compressional speed $c_{p,sed} = 1770$ m/s, density $\rho_{sed} = 2100$ kg/m$^3$, as derived from the analysis of a core. Figure 1(a) shows the photo of typical sand ripple structure in the measurement area; a compass indicates the North. All the selected measurements were collected under isospeed conditions: water sound speed $c_w = 1507$ m/s; water density $\rho_w = 1040$ g/m$^3$.

During the day when the data presented here were collected, weather conditions were good, with sea state 1. Figure 1(b) shows a photo of the sea surface. As estimated in the following, the roughness is characterized by a swell of small amplitude (hardly detectable from the photo) on which mid- and small-scale wavelets are superimposed. The main direction of the wind remained constant, blowing from 135° North with an average speed of 4 m/s.

In order to conduct scattering measurements from a fixed location, a directional source was placed at the top of a 3.5 m high tower deployed on the seabed. The transmitter was coupled with a backscattering hydrophone in quasi monostatic configuration. The sonar was rigidly connected to a rotation motor to tune its pan and tilt orientations, which were monitored by a motion reference unit (MRU).

The transmitter frequency is 300 kHz with 60 kHz bandwidth at -3 dB. Its beamwidth at -3 dB is approximately 60° on the horizontal plane and 0.7° on the vertical plane. A vertical array of 3 omnidirectional hydrophones (40 cm spacing) was suspended from a portable frame, which was deployed in bistatic configuration at a variable range between 30 and 70 m. From a rubber boat, it was possible to adjust the receiver height at a number of equally-spaced, fixed positions through an adjustment line connected to the receiver frame by a sliding and blocking mechanism. Changing the range and depth of the receive array enabled the receiver to be positioned at the nominal specular reflection direction once a certain sonar incident angle was selected. The system was connected to the NURC’s Coastal Research Vessel (CRV) Leonardo. The transmit/receive system was calibrated and identified using a 1ms LPM (linear period modulation) pulse in the bandwidth 270-330 kHz. For acoustic scattering measurements a 10 ms LPM pulse was used. While most of the parameters can be considered stable along the short period of measurement (1 minute), there are two exceptions: the Tx/Rx distance and the sea surface roughness. In post-processing, the scattered echoes are aligned with respect to the direct arrival in order to compensate for the receive array motion due to the sea waves action on the portable frame. The variability of the Tx/Rx distance is included in the simulations. Matched-filtering was applied to data and simulation to improve the signal-to-noise ratio and the time resolution.

Fig.1: (a) Photo of the seabed in the measurement area. (b) Photo of the sea surface.
The presented results were obtained under the following conditions (see Fig.2): the water depth was 10.1 m at the transmitter and 9.4 m at the receiver (29.7 m away), with a smooth up-slope of 1.3\(^{\circ}\).

The transmitter and the hydrophone, located in quasi-monostatic configuration, were also used to measure the wave height at high ping rate and with a relatively high spatial resolution. Wave height measurements were alternated to scattering measurements of the sea surface in order to be able to correlate the two datasets. The sonar was pointed to the sea surface at normal incidence and a burst of 50 \(\mu\)s was sent each 40 ms. Wave height was measured from the time of arrival of the echo from the surface with respect to the transmission time. As the wave displacement was estimated from the first arrival of the surface echo, its spatial resolution was approximately the minimum width of the sonar footprint on the sea surface, which was about 6 cm in the geometrical configuration selected. This resolution is only one order of magnitude bigger than the sonar wavelength (equal to 5 mm). Figure 3(a)-(c) shows the wave displacement vs. time acquired by a Seabird static pressure sensor, a Teledyne RDI ADCP1200, mounted such one of its transducers pointed to the sea surface at normal incidence, and the 300-kHz acoustic system described above, respectively. Also the two former instruments were mounted at the top of the 3.5m high tower. The data samples in Fig. 3(b) and (c) are synchronized; the data in (a) were recorded in the same day under the same weather conditions. The capability of the proposed acoustic system to catch higher frequency components is evident. In Fig. 4(a) the nondirectional spectrum of the time series plotted in Fig. 3(c) is computed in the bandwidth 0.1-5 Hz, and the corresponding wave height is estimated versus range (Fig. 4(b)) through the simplified characteristic equation linking time frequency \(f\) and spatial wavelength \(\lambda\) of the waves:

\[
\frac{1}{2\pi} \sqrt{\frac{gk}{\tan(hH)}} \tag{1}
\]

where \(g\) is the acceleration of gravity, \(k\) the wavenumber \((k = 2\pi/\lambda)\) and \(H\) the water depth. This estimate can be used to feed scattering, reverberation and propagation modeling tools working under the condition of rough sea surface.
3. EXPERIMENTAL RESULTS OF ACOUSTIC SCATTERING AND MODEL-DATA COMPARISON

A selection of single-bounce scattering measurements in bistatic configurations are presented in this section along with corresponding model-data comparison results. Figure 5 shows a time domain comparison of forward scattering from the sea surface. An example of experimental time aligned data is given in Fig. 5(a) and (b). The results shown are relative to 40 pings registered at a single receiver position (4.05 m of altitude). The sea surface arrivals are poorly aligned and also change in shape because the sea surface is changing in height and shape, ping after ping. Figures 5(c) and (d) show the corresponding results of BORIS-SSA simulations after matched filtering and realignment (to remove the receiver horizontal displacement which was included in the simulation), which agree well with the experimental data. Only surface scattering (second order SSA) is used in this case. The differences between real data and simulations are caused by the presence of noise in the experimental data and by the distribution of the arrival times, the experimental pings being more concentrated around the central arrival. The latter difference is probably caused by the imperfect sea surface model (especially for long and medium wavelengths).
The sea surface variability is simulated taking into account three wavelength scales. BORIS-SSA is used to produce stochastic realizations of small patches of sea surfaces that lie on a horizontal plane (2x2 m). These patches take into account the small and medium scale roughness by adding two surfaces together: (a) an isotropic Gaussian power law spectral density stochastic surface with rms height of 2 cm and correlation length of 33 cm (aimed to simulate coastal isotropic wave); and (b) a Gaussian power law spectral density stochastic surface translated in the wind direction, with rms height of 0.5 mm and wave correlation length of 1 cm (to simulate wind generated capillary wave) [5]. This surface model is preferred to other well established sea surface models (such as the Pierson-Moskowitz models included in BORIS-SSA) because it takes into account small scale roughness. This small patch is added to a large scale roughness (wavelength of the order of tens of meters) wave generated using a Gaussian power law spectral density stochastic surface translated in the wave direction, with rms height of 113 cm and wave correlation length of 16 m (these parameters are estimated from Figs. 3(c) and 4(b)).

Figure 6 shows measured and simulated specular reflection returns from the seabed acquired at a grazing angle of 12.4°, which is much lower than the critical angle, estimated...
here as 31.6°. The BORIS-SSA simulations are obtained using a seabed roughness model consisting of (a) a directional Gaussian power law spectral density stochastic surface with the same orientation as the sand ripples, rms height of 1.1 cm and correlation length of 25 cm; and (b) a Gaussian power law spectral density stochastic surface aimed to simulate the grain size, with rms height of 0.5 mm and wave correlation length of 3.3 cm. During this experiment the transmitter was randomly tilted ping to ping within a range of ±1° around the nominal grazing angle. The model (Fig. 6(c) and (d)) takes this system variability into account, as well as the variability in the transmit/receive distance. The peaks of the scattering amplitude in Fig. 6(a) and (b) are much lower than the theoretical reflection coefficient expected for a perfectly flat interface, due to the surface roughness (sand ripples and micro-roughness), which allows both penetration into the sediment and scattering. Fig. 6 shows a lower variability with respect to Fig. 5 because of the stationary state of the seafloor roughness.

4. CONCLUSIONS

An experimental system used to measure high frequency scattering from rough interfaces is presented. The spreading of sound bouncing off the sea surface and the seafloor is such,
that the maximum amplitude of the respective echoes are much lower than what one would expect from the theoretical reflection coefficient from perfectly flat interfaces.

Using BORIS-SSA, it is possible to confirm what one would expect from physical considerations: the main causes of time of arrival variability, time dispersion and amplitude variability appear to be the roughness of the interface. Variability of the boundary height, slope and fine scale roughness limits the applicability of simple flat interface specular-reflection ray approximation in such cases. The simulations presented here take into account the variability of the boundary parameters, which improves the model-data agreement. The large-scale roughness mostly influences the time of arrival, the variability of the peak shape and the dispersion of the energy in other directions than the specular one. The small and medium-scale roughness mostly influences the (maximum) amplitude and the single ping time dispersion. Scattering measurements from the seabed could not involve the same level of variability but confirmed the good agreement with simulations in terms of echo shape and time spreading.

The same experimental acoustic system was used in monostatic configuration to obtain relatively high-resolution temporal measurements of the sea surface wave height. However this system is not able to provide information of the directionality of the waves. Future activities will include a more accurate modelling of the sea surface. A more refined system to measure the sea surface wave height at high resolution, which is based on stereo photogrammetry [6] and provides also the estimation of wave directionality, is under development and is foreseen to provide realistic inputs to available scattering and propagation modelling tools.

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REFERENCES

A NEW SYSTEM FOR SEAFLOOR CHARACTERISATION: BRAD, THE BENTHIC ROUGHNESS ACOUSTIC DEVICE

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Small scale (10^{-2} to 10^{-1} m) roughness of the seafloor is of direct relevance to a range of interests, including boundary layer hydrodynamics, sediment transport and high frequency acoustic scattering. Methods for the quantification and characterisation of seafloor microtopography for a range of natural roughness scales are therefore essential.

A new acoustic system has been developed at the National Oceanography Centre, Southampton (NOCS), UK, the Benthic Roughness Acoustic Device (BRAD). This system is used to define micro-topographical roughness of the seafloor through high-resolution acoustic imaging. BRAD is composed of a high-frequency profiling sonar, the Sediment Imager Sonar (SIS) and a small driver motor, both mounted on a frame.

BRAD measures high-resolution relief of the seabed over an area of 1.7 m^2. A two-dimensional spectral analysis is then applied on the seabed elevation data in order to characterise seafloor roughness. Three field deployments in areas with different bottom types (seagrass canopy, bioturbated mud and sandy ripples) are presented and discussed here. The seagrass canopy was successfully differentiated from the underlying bed, which presented an isotropic spectrum. The bioturbated seabed was also characterised by an isotropic roughness spectrum, while the ripple field roughness spectrum exhibited a strong anisotropy in the direction of the ripples. At each site, the slope and intercept of the power-law regression line fitted to “slices” taken through the 2D roughness spectrum were also used to characterise topographical roughness.

Keywords: bed roughness, high-resolution acoustic imagery
1. Introduction

Small-scale (10^{-2} to 10^{-1} m) roughness of the seafloor is of direct relevance to a range of interests such as boundary layer dynamics [1], sediment transport [2] and acoustic scattering [3]. Seafloor topography is influenced by both the nature of the sediment, such as grain size and sorting (grain roughness), and the contributions of several roughness producing mechanisms. Under the action of waves and currents, bedforms such as sand ripples develop, creating topographical variations (bedform roughness). In the absence of sediment transport by waves or currents, the sediment surface is continuously being modified by the locomotion and/or feeding activity of animals that live on and within the sediment creating biogenic roughnesses. Furthermore, the animals and plants living on the seafloor themselves constitute a relief of the seabed. While bedform roughness is usually anisotropic, i.e. it presents a preferential direction (function of wave and current directions), biological roughness is generally isotropic [3].

Despite the importance of bed roughness on near-bottom flow and sediment transport, only a few studies have quantitatively resolved seafloor height at the relevant scales. Traditionally, stereo-photography has been used to compute 1D or 2D high-resolution seabed elevation data used to spectrally characterise bed roughness [3-6]. Acoustic systems have the potential to produce high-resolution images of the seabed, which can then be used to study topographical roughness. To date, however, high-resolution acoustic systems produce only profiles of seabed elevation (i.e. 1D) which have been used principally to study ripple dimensions (e.g. [7]). This can be limiting as roughness is a spatially varying parameter and the direction of the profile can have an important effect on the resulting analysis [8].

A new acoustic system has been developed at the National Oceanography Centre, Southampton (NOCS), UK, the Benthic Roughness Acoustic Device (BRAD). This apparatus provides high-resolution acoustic measurements of seabed elevation over an area of 1.7 m². In this study, we test the application of spectral analysis to characterise topographical roughness from seabed elevations computed using BRAD.

2. System Description and Methodology

2.1. The System

The Benthic Roughness Acoustic Device is composed of a profiling sonar – the Sediment Imager Sonar (SIS) - and a small driver motor which are mounted on a frame (1.7 m long, 1 m high and 1 m wide) in aluminium (Fig. 1). During deployments, the SIS is down-looking at a height of 0.88 m above the bed. It is advanced along the length of the frame by the motor at a velocity of 0.56 mm s^{-1}. The SIS is a single-beam profiling sonar with a rotating head manufactured by Marine Electronics Ltd. (Guernsey). It emits a pencil-beam sound wave (beam width angle 1.8°) at a frequency of 1.1 MHz. The sonar beam is swept at right angles to the sonar body and sweeps are made in an arc of 90° (beams every 0.9°) centred about nadir so that data are acquired 45° each side of the vertical (total of 101 beams per sweep). Echograms (backscatter intensity along the beams) are displayed, recorded and converted to ASCII image intensity by the Sediment Imager Control Software®. The data available after conversion are beam angle (degrees), distance from transducer (m) and backscatter intensity (0 - 255). With a range of 2 m, a transmit pulse of 10 µsec and a sample interval of 5 µsec, the along-beam resolution is about 3.75 mm (dependant on the sound velocity in water). The horizontal resolution is a function of beam angle and varies from 2 to 3 cm.
The SIS cannot be operated autonomously and so is connected to a power supply and a computer, which provides real-time display and recording of the data. BRAD can be deployed by 3 people from an anchored boat or a pontoon in water depths ranging from 1 to 6 m. The maximum area of the seabed that can be surveyed in a single deployment is 1.7 m² in a period of around 50 minutes.

2.2. DEM PRODUCTION

BRAD was deployed at three sites in Venice Lagoon, over a Zostera marina canopy (Site 1), bioturbated mud (Site 2) and rippled fine sand (Site 3). Digital elevation models (DEMs) of the seabed imaged with BRAD were created from the raw SIS data for each site. A threshold method was used to calculate the depth of the bed along each beam of each sweep. Seabed detection was initially tried using only the backscatter intensities along a beam. This method, however, often gave inaccurate results as the threshold values were found to be dependent on the intensity of the maximum backscatter and backscatter intensity in the water column, which vary from beam to beam (in particular with beam angle). To address this problem, a threshold using a quotient $Q$ was applied:

$$Q(Z) = \frac{B(Z) - W}{B_{\text{max}} - W}$$

where $B(Z)$ is the backscatter intensity at the depth ($Z$) under the SIS head and $W$ is the average backscatter 0.3 to 0.5 m from the transducer head (Fig. 2a). $Q$ has values of on average 0 in the water column and 1 at maximum. The first point along the beam where $Q(Z)$ was greater than 0.8 was defined as the depth of the seabed (Fig. 2a). Beam angle and depth of the seabed were used to compute the $x$ co-ordinates associated with the points satisfying the threshold condition. The positions along the frame ($y$) were calculated from the motor velocity, the time to complete a sweep and the beam number. Finally, the seabed elevation ($z$)
was computed as the deviation from the theoretical depth of the seabed under the SIS head \((Z = -0.88 \text{ m})\).

This processing was applied on datasets from Sites 2 and 3. Site 1, however, was undertaken over a seagrass (\textit{Zostera marina}) canopy. When a sound wave reaches submerged vegetation, some of the acoustic energy is scattered from the vegetation back to the source, creating strong backscatter above the seabed [9]. Along the beams collected at Site 1, a high backscatter response was seen from both the seabed and the seagrass canopy and could be differentiated from ambient noise (Fig. 2b). Several methods were tested to allow computation of both the height of the seabed and the height of the canopy. The method used here assumes that the bed under the canopy was relatively flat and situated no more than 4 cm above the theoretical depth of the bed \((Z = -0.88 \text{ m})\). Along a beam, the seabed depth was taken as the first point where \(Z < -0.84\) and \(Q(Z) > 0.8\) and the height of the seagrass was calculated as the first point where \(Z > -0.84\) and \(Q(Z) > 0.6\) (Fig. 2b). Seabed and canopy elevations of each site were thereafter interpolated over a regular grid (cell size of 0.01 m).

![Fig. 2: Values of relative intensity of the backscatter \((Q)\) of a beam along the depth under the SIS head \((Z)\). (a) Bare seabed. The maximum backscatter \((Q = 1)\) is showed as a black dot. The depth of the seabed is indicated by the red line and red dot (first point where \(Q > 0.8\)). The rectangle shows the extent of the backscatter intensities averaged to calculate \(W\). (b) Seagrass canopy. The canopy height \((CH)\) is showed by the green line and green point (first point where \(Q > 0.6\) and \(Z > 0.84 \text{ m}\)). The depth of the bed is indicated by the red line and the red dot (first point where \(Q > 0.8\) and \(Z < 0.84 \text{ m}\)).](image)

### 2.3. ROUGHNESS CHARACTERISATION

Topographical roughness can be described through statistical parameters such as the root mean square (RMS) of the elevation distribution (e.g. [10]). Spectral analysis, however, proved to be more efficient as it characterises the variance of seabed topography as a function of spatial frequency [6]. A power-law can be fitted to the spectrum of a 1D profile or to a “slice” of a 2D spectrum in log-log space. The slope and intercept values of the regression line are often used by acoustic modellers to predict bottom scattering [11]. Spectral analysis was applied to the DEM using the methodology detailed by Lyons et al. (2002) to
characterise the 2D roughness of a rippled bed imaged with digital photogrammetry. DEMs were first multiplied by a tapering function (Discrete Prolate Spheroidal Sequences) to reduce spectral leakage, which is caused by the finite size of data segments used for analysis [12]. While reducing bias, data tapering also causes a reduction of resolution, or a smoothing effect, in the spectral estimate. The seabed elevations were then transformed to the spatial domain using a 2D Fast Fourier Transform. “Slices” were taken through the 2D roughness spectrum in one degree steps from 0° to 180°. Each slice through the 2D roughness spectrum is a 1D representation of the 2D roughness spectrum in a particular orientation, but it is not the same as the 1D roughness spectrum estimated from a 1D profile (see [3] for more details). In case of rippled seafloors, a power-law regression was fitted to slices taken in directions along and at 90° to the peak in the spectrum caused by the ripples. At the other sites, a power-law regression was fitted to the average of all the slices. The slope and intercept (at a spatial wavenumber of 10°3.) of the power-law regression lines were used to characterise bed roughness.

3. Results

The first site imaged in Venice Lagoon consisted of a Zostera marina canopy. The seabed under the canopy was composed of fine sand with a small fraction of mud and shells. Fig. 3 presents the reconstructed seabed elevation at Site 2 and the 2D roughness spectrum computed from the seabed elevations. The spectrum was essentially isotropic; that is, the spectrum has a central peak without any pronounced directionality in 2D spectral frequency space [6]. The reconstructed canopy elevation at Site 1 is also shown on Fig. 3. The canopy had an average height above the bed of 19.5 cm (standard deviation of 4.9 cm). Its surface was found to be uneven and irregular (height varying between 5 and 35 cm above the bed).

Deployment at Site 2 was carried out inside Venice Lagoon over bioturbated mud. An important number of bioturbation holes, around 2 cm in diameter and deeper than 6 cm, were also seen. Unfortunately it was not possible to observe the organisms making these holes and hence their origin is unknown. The bioturbation holes could be recognised along certain beams as a strong backscatter below the bed. Seventeen bioturbation holes were counted within the 1.7 m² of seabed imaged. The 2D spectrum computed from seabed elevations at Site 2 was essentially isotropic (Fig. 4).

The isotropy of the spectra estimated from the seabed elevations at Sites 1 and 2 suggests that the seabed relief at these sites at the time of the imaging was weakly influenced by hydrodynamics, which usually creates directionality in the spectrum. Roughness at those sites was most likely created by biota activity, which is generally isotropic. Furthermore, the strong power contained in the high frequencies shows the importance of small scale roughness, whereas the low power contained by low frequencies highlights the lack of large wavelength features, further suggesting a predominance of biogenic roughness at these sites.

Site 3 was occupied outside Venice Lagoon over well-sorted fine sand. The seabed imaged was covered by symmetrical, sharp-crested, wave ripples (wavelength of approximately 10 cm and a height of around 1 cm) with numerous bifurcations. Seabed elevations reconstructed at Site 3 are presented on Fig. 5. The reconstructed ripple field shows sharp-crested and symmetrical bedforms, with occasional bifurcations, which agrees well with on-site description. The 2D spectrum computed from the seabed elevations at Site 3 clearly shows the anisotropy associated with the ripple field (Fig. 5) as well as the peak in power associated with the wavelength of the ripples.
Fig. 3: Top: seabed elevations at Site 1 and the isotropic 2D roughness spectrum estimated from the DEM (some spectral leakage can still be seen as horizontal lines along the spectrum). Bottom: 3D view of the canopy elevations over the seabed and slices through the 2D spectrum (45° and 135°) together with the power-law line fitted on the average of all the slices.

Fig. 4: Isotropic 2D roughness spectrum estimated from the seabed elevations at Site 2 and slices of the spectrum (45° and 135°) together with the power-law line fitted on the average of all the slices.
Fig. 5: Seabed elevations at Site 3 showing the ripple field and anisotropic 2D roughness spectrum estimated from the DEM.

Slope (γ) and intercept (ω) values calculated from the slices through the 2D spectra are summarised in Table 1, together with the median grain size diameter and RMS values of the seabed elevation distribution at each site. The highest slope was calculated from the slice taken perpendicular to the ripple crest at Site 3, showing the importance of the low frequencies compare to the high frequencies for this case of bedform roughness. The smaller slope, which was calculated from the average of the slices computed from Site 2 spectrum, shows that biogenic roughness is characterised by a relatively stronger contribution of the high-frequencies compare to the low-frequencies (i.e. roughness at the bioturbated site is characterised by high spatial frequency fluctuations in sediment height), as has been described in other studies [11].

<table>
<thead>
<tr>
<th>Site</th>
<th>d₅₀ (ø)</th>
<th>Type</th>
<th>RMS (cm)</th>
<th>Slope</th>
<th>Intercept (m⁴)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Site 1</td>
<td>2.54</td>
<td>fine sand</td>
<td>0.85</td>
<td>4.1</td>
<td>0.09022</td>
</tr>
<tr>
<td>Site 2</td>
<td>5.02</td>
<td>bioturbated silt</td>
<td>0.83</td>
<td>3.5</td>
<td>0.02750</td>
</tr>
<tr>
<td>Site 3 (c-c)</td>
<td>3.16</td>
<td>rippled v. fine sand</td>
<td>0.64</td>
<td>4.7</td>
<td>0.39884</td>
</tr>
<tr>
<td>Site 3 (a-s)</td>
<td>&quot;</td>
<td>&quot;</td>
<td>&quot;</td>
<td>4.1</td>
<td>0.09497</td>
</tr>
</tbody>
</table>

Table 1: Summary of median grain diameter (d₅₀), seabed type, RMS height and slope and intercept of the power spectrum at each site imaged with BRAD. At Sites 1 and 2, slope and intercept values were calculated from the average of all the slices; at Site 3, they were computed from slices taken perpendicular (c-c) and parallel (a-s) to the ripple crests.

Slope and intercept values calculated in this study were compared with slope and intercept values compiled in [3]. To enable comparison, the values calculated from 1D seabed elevations acquired with stereo-photography were transformed to 2D values in metre units following the method detailed in [3]. The slope and intercept values calculated from BRAD data were found to be higher than those of previous studies (3.5 and 0.0021 m⁴ on average for a variety of roughness types). The reasons for such a difference are not fully understood, but might come from the difference in the systems used to image the seabed (acoustical versus optical), the length of the seabed analysed and the systems specific resolution.
4. CONCLUSIONS

The development and first use of BRAD, a new high-resolution acoustic system designed specifically for seafloor roughness characterisation, are presented here. The system was deployed at 3 sites in Venice Lagoon over a variety of bottom types (Zostera marina canopy, bioturbated silt and ripple field). The DEMs were successfully produced from BRAD raw data and the 2D roughness spectra estimated from the seafloor elevations were used to characterise topographical roughness at each site. The roughness spectra clearly show the anisotropy of the ripple site and isotropy of the bioturbated seabeds. The slope and intercept values of slices through the spectra help in assessing the relative influence of small and high-frequencies on topographical roughness. A variety of sites should be imaged with BRAD and characterised through spectral analysis in order to construct a catalogue of natural roughnesses.

REFERENCES

OPTICAL AND ACOUSTIC SEAFLOOR ROUGHNESS MEASUREMENT USING SHAPE-FROM-SHADING TECHNIQUES

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Abstract: Systems for measuring seafloor height fields at scales essential for understanding and modeling acoustic scattering exist and include optical techniques such as stereo photogrammetry (scales of millimeters to less than a meter) and acoustic techniques such as interferometric sonar (scales of decimeters to hundreds of meters). These systems, however, can be expensive and complicated in terms of set up, calibration, and processing. This paper will focus on two applications (optical and acoustic) of a technique known as shape-from-shading (SFS) that allows simpler (and cheaper) systems to be used to obtain elevation maps of the seafloor. Particular issues that arise when going from the optical to the acoustic systems will also be discussed such as speckle noise and the inclusion of seafloor scattering models into the processing. Preliminary results of the SFS technique used on optical images of the seafloor and on high-frequency synthetic aperture sonar images will be shown.

Keywords: Shape from shading, seafloor roughness measurement, acoustic scattering.
1. INTRODUCTION

Small-scale seafloor topography (sometimes termed ‘micro-topography’ or just ‘roughness’) on the scale of millimeters to tens of meters is a fundamental seafloor property affecting a variety of physical phenomena including sediment transport and the interaction of acoustic energy with the seafloor. Sediment ripples are small-scale topographic features that have particularly strong impacts on acoustic interaction with the seafloor. Ripples in the appropriate orientation have been found to increase penetration of acoustic energy at sub-critical grazing angles allowing improved detection of buried objects. In modeling acoustic scattering from the seafloor, the power spectrum of height fields is of particular importance because it is this quantity which uniquely determines the first-order averaged acoustic scattered intensity (as a function of grazing angle) under perturbation theory.

Systems for measuring seafloor height fields at scales essential for understanding and modeling acoustic scattering exist and include optical techniques [1] such as stereo photogrammetry (scales of millimeters to less than a meter) and acoustic techniques [2,3] such as interferometric sonar (scales of decimeters to hundreds of meters). However, these systems can be expensive and complicated in terms of set up, calibration, and processing. The work presented in this paper focuses on two applications (optical and acoustic) of a technique known as shape-from-shading (SFS) that allows simpler (and cheaper) systems to be used to obtain elevation maps of the seafloor. Particular issues that arise when going from the optical to the acoustic systems, such as speckle noise and the inclusion of seafloor scattering models into the processing, are also addressed. The major advantages of the optical and acoustic techniques studied is their relative simplicity and robustness compared to other methods. Details and sample results of the SFS technique used on single optical images of the seafloor and on high-frequency synthetic aperture sonar (SAS) data are given below.

2. METHODS

The inversion method used in this work for both optical images and acoustic data makes use of the fact that shading encodes shape (hence the technique is commonly referred to as ‘shape-from-shading’ [4]). If the albedo (or surface reflectance) is constant over a region, changes of intensity correspond to changes in the surface normal of the scene (surface normals are perpendicular to the surface slope at any point). For the optical case shape-from-shading requires knowledge of the bi-directional reflectance distribution function (BRDF) and for the acoustic case the analogous bi-static scattering cross section is required. A commonly assumed BRDF is the Lambertian reflectance model. A Lambertian surface scatters light or sound equally in all directions and is commonly used in computer vision and graphics to represent the shading of natural surfaces. Under the Lambertian reflectance assumption, the observed intensity is proportional to the cosine of the incidence angle, i.e., the scalar product between the unit vectors in the directions of the illuminant and the surface normal. The optical and acoustic systems being studied for measuring small seafloor topography required use of single images with the light or acoustic energy coming from a single source direction. The basic problem for inverting for seafloor roughness is then: given a single image (acoustical or optical) of an object (the seafloor in our case) with known surface reflectance, taken with a source of known direction, can we recover the shape of the object?
In general the problem of single-view shape-from-shading is underdetermined - there are many 'solutions' or surface normals that can produce the same brightness in a single image. This problem can be addressed by adding constraints to the equations representing the relationship between the brightness of the pixels and the surface normal producing the brightness. Of the several techniques that have been studied in the past, the relaxation method was used in our case [5]. The relaxation method allows the surface orientation to be obtained by using the image irradiance equation and a smoothness constraint. Once the surface orientations are estimated the exact shape can be determined by a number of different methods. Direct integration is possible but errors accumulate quickly. Of the several methods for determining shape from surface orientation that are better than direct integration we chose to use a method that correlates shapelets (basis functions similar to wavelets or sinusoids) with surface normals to produce surfaces [6]. For both the optical and acoustic cases, correction is required for both a non-uniform illumination source and a non-constant source direction over the image.

3. OPTICAL RESULTS

An example of the shape-from-shading technique is given in Figures 1 and 2 below. The original intensity image in Fig. 1 is a photograph of a sandy sediment in the Gulf of Mexico off of Panama City, Florida, which has been raked by divers. The image was taken with an underwater camera as part of the ONR sponsored SAX99 sediment acoustics experiment. The surface slopes estimated with shape-from-shading are also shown in Fig. 1. Fig. 2 displays the final height field calculated from the surface slopes. The heights field matches closely that determined by both stereo-photogrammetric methods and diver measurements [7]. In terms of noise (measurement error) and aerial coverage, the result is actually better than that obtained using photogrammetry.

Fig. 1: Original optical image (left) and slope estimated using shape from shading techniques (right).
4. SYNTHETIC APERTURE SONAR RESULTS

SFS techniques developed for optical inversions can successfully be used on SAS data provided that, as for the optical images, corrections are applied for non-constant ensonification (due to sonar beam patterns) and non-constant source direction, with the additional corrections of speckle reduction, co-located source and receiver, and for the fact that seafloor scattering is not Lambertian. Adaptive mean filtering has been shown to be optimal for speckle reduction [8,9] and was applied to our sample SAS data. It is exceedingly difficult (perhaps impossible) to reformulate the inversion scheme presented in [5] using the acoustic bistatic scattering cross section. It is feasible, however, to use the apparatus already developed for the Lambertian case by simply adjusting the SAS image using the known relationship between Lambertian scattering and perturbation scattering theory (this relationship is shown in Fig. 3). By doing this, we are simply accounting for the fact that the Lambertian assumption overestimates scattered levels at low angles and underestimates scattered levels at higher angles.

Fig. 2: Estimated height field calculated from the slopes (units are in cm).

Fig. 3: Relation between the Lambertian scattering model (with a level set by the coefficient, \( \mu \)) and the perturbation approximation model at a given angle.
Figure 4 shows a SAS image produced from data collected by the Naval Surface Warfare Center - Panama City Division during a demonstration experiment which took place near La Spezia, Italy off the Ligurian coast and was obtained on a homogeneous rippled sand seafloor [10]. This data set consisted of broadband transmissions from 105-135 kHz and the resulting images had a resolution of approximately 2 x 5 cm. The sand ripples of almost 1 m wavelength are easily seen in the image. The 3-D shape and a height profile of the ripples determined with the SFS technique are displayed in Fig. 5.

Fig. 4: Image of a rippled sandy seafloor produced using synthetic aperture sonar data (data provided by NSWC-Panama City).

Fig. 5: Height field obtained from SFS using the data of Fig. 4 (note the scales of the 3-D image have been modified to highlight the topography and are different for each axis.)
5. SUMMARY

The work presented in this paper focused on the optical and acoustic application of the shape-from-shading technique to the problem of obtaining elevation maps of the seafloor. Initial results of the technique applied to seafloor photographs and SAS data are encouraging. The major advantages of the optical and acoustic techniques studied is their relative simplicity and robustness compared to other methods such as stereo photogrammetry or interferometric SAS. In spite of the fact that the corrections applied to the data were not perfect (we lacked some knowledge of system and seafloor parameters), the capability of converting 2-D information to 3-D for acoustic data using SFS is obvious. The SFS methods presented here will be refined in the future and ways will be explored of inserting these techniques into operational systems.

6. ACKNOWLEDGEMENTS

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REFERENCES

INVERSION FOR THE OCEAN SURFACE WAVE DIRECTIONAL SPECTRUM USING THE DOPPLER-SHIFTED, BRAGG-SCATTERED SIDEBANDS OF LOW-FREQUENCY NARROWBAND ACOUSTIC TONES

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Abstract: The frequency and directional content of ocean surface wave fields can be estimated from high-resolution beamformer measurements in the vertical plane between an acoustic source and a 2-dimensional, horizontal receiving array in a shallow ocean waveguide. The acoustic tones transmitted in the experiment are low frequency - 280, 370, 535 and 695 Hz. From first-order perturbation theory, the frequency shift of a narrowband acoustic tone scattered from the ocean surface is dependent on the surface wave frequency, while the arrival angle deviation depends on the acoustic and ocean surface wavenumber vectors through the Bragg condition. All ocean surface waves are assumed to be linear. The inversion results are compared to the directional ocean surface wave spectrum measured independently using an ocean wave vector sensor (i.e., one that measures ambient pressure and the two horizontal components of flow) near the acoustic source. Complications arising from, and additional information contained within, multi-path acoustic propagation, higher-order scattering from the surface waves, and multiple interactions of the acoustic signal with the surface are addressed. [Work supported by ONR, Code 321(US)].

Keywords: Doppler, scattering, ocean surface waves, inversion
INTRODUCTION

In shallow water environments acoustic signals interact with the surface multiple times between the source and receiver, and the surface waves distort the acoustic signals through scattering. Systems that utilize these distortions to gain information such as surface gravity wave directional spectra are not new, but the vast majority of them operate at high acoustic frequencies (>1000 Hz). There are considerably fewer inversion results using measured low frequency acoustic signals.

Omnidirectional spectra of received low-frequency narrowband tones have been observed with Doppler-shifted sidebands whose frequency content and amplitudes relative to the non-shifted center band depend on the prevailing surface waves. Sidebands in beamformer output have shown angle-of-arrival and Doppler-shifted frequency dependence that suggests Bragg scattering by ocean surface waves, and the angles of the peaks of these bands have been used to compute surface wave propagation angles at the dominant surface wave frequency [1].

The goal of this paper is to use narrowband tones recorded in a fixed source/fixed receiver set-up along with first-order perturbation scattering theory to invert for the directional surface wave spectrum. Most previous efforts to do this have been largely based in numerical modelling [2]. Results presented in this paper mimic those modeled set-ups, using signals scattered in the vertical plane between source and receiving array.

![Surface wave directional spectrum](image)

**Fig.1:** Surface wave spectrum and direction (solid lines) computed from PUV measurements, and surface wave propagation direction inverted from acoustic vertical beamformer output.

Figure 1 shows the results from an effort to use scattered narrowband tones at frequencies 280, 370, 535 and 695 Hz measured on a fixed broad-aperture horizontal array deployed on the bottom to invert for the directional wavenumber content of the surface waves.
Background and Methods

1.1. First order perturbation and scattering

A common mathematical method of treating the distortion of acoustic signals by interaction with a rough moving surface is to expand the pressure field $\psi$ in an asymptotic series in terms of a small expansion parameter such as $kh$, where $k$ is the acoustic wavenumber amplitude and $h$ is the RMS amplitude of the surface roughness. The upper pressure-release boundary condition is maintained by a Taylor expansion of $\psi$ about the mean surface $z=0$ [3]. The Fourier transform with respect to time of the first order upper boundary condition is

$$\tilde{\psi}^{(1)}(\mathbf{r}, \omega) = \tilde{\eta}(\mathbf{r}, \omega) \otimes \frac{\partial}{\partial z} \tilde{\psi}^{(0)}(\mathbf{r}, \omega) \bigg|_{z=0}$$

Here the convolution is over temporal frequency $\omega$, $\eta$ is the surface roughness, the tilde denotes Fourier transform in time, and $\tilde{\psi}^{(0)}$. Thus by Eq. (1) the amplitude of the first order correction to the acoustic field ($\tilde{\psi}^{(1)}$) depends on the surface wave field and the unperturbed pressure $\tilde{\psi}^{(0)}$.

It is easy to show by finding a mode solution to the Helmholtz equation for the first-order acoustic field term $\tilde{\psi}^{(1)}$ with the upper boundary condition (1) and Fourier transforming in space that each propagating mode is scattered by the surface waves. The first-order scattered field consists of Doppler-shifted components propagating in different directions. Scattered acoustic frequencies are $\omega = \omega_o \pm \sigma$, where $\sigma$ is surface wave temporal frequency, and scattered acoustic directions of propagation are described by the Bragg condition, $\mathbf{k} = \mathbf{k}_n \pm \mathbf{k}_\kappa$, where $\mathbf{k}_n = [k_{x_n}, k_{z_n}]$ is the acoustic mode wavenumber vector (in an iso-speed waveguide, $\mathbf{k}_n = \mathbf{k}$), and $\mathbf{k}_\kappa$ is the surface roughness wavenumber vector. The surface wave dispersion relation describes the relationship between frequency and wavenumber:

$$\sigma^2 = g\kappa \tanh(\kappa H)$$

In Eq. (2) $g$ is the gravitational constant ($9.8 \text{ m s}^{-2}$) and $H$ is the depth of the ocean bottom.
1.2. Scattering in the vertical plane

The surface wavenumber vector $\mathbf{\kappa}$ is purely horizontal, and thus the change to the acoustic wavenumber resulting from scattering is purely in the horizontal component $\kappa_r$. If $\mathbf{\kappa}$ is decomposed into components that are parallel and perpendicular to the vertical plane between the acoustic source and receiving array, $\mathbf{\kappa} = [\kappa_r, \kappa_\perp]$, where $\kappa_r = \kappa \cos \phi$ and $\phi$ is the azimuthal angle of propagation of the surface wave with frequency $\sigma$ (Fig. 2). Then vertical angles $\theta_n^{\pm}$ of arrival of first-order scattered sidebands for the $n^{th}$ propagating acoustic mode in an iso-speed waveguide are described by

$$\tan \theta_n^{\pm} = \frac{k_{n_r}^{\pm}}{k_{n_z}}$$

In Eq. (3) the frequency of the sideband scattered with angle $\theta_n^{\pm}$ is $\omega_0 \pm \sigma$. Note also that the non-Doppler-shifted angle of arrival for mode $n$ is related to the vertical and horizontal acoustic wavenumber components through $\tan \theta_n = \frac{k_{n_z}}{k_{n_r}}$, and $k^2 = k_{n_r}^2 + k_{n_z}^2$.

Fig. 2: Plan view of the source/receiver configuration and the surface wavenumber vector $\mathbf{\kappa}$

After rearranging Eq. (3) for $\kappa_\perp$ in terms of the measured angle of arrival $\theta_n^{\pm}$ and finding wavenumber $\kappa$ from measured frequency shift $\sigma$ (Eq. (2)), it is easy to compute the angle of propagation of surface waves from the output of an acoustic beamformer:

$$\phi = \cos^{-1} \frac{\kappa_\perp}{\kappa}$$

(4)
Note that, for first order scattering, there can be two simultaneous measurements of $\phi$, one each for the positively and negatively shifted sidebands.

1.3. Experiment and processing methods

The measurements in this study were made in an experiment off the coast of southern California in approximately 10 m of water. The 64-element hydrophone array, which had 1.875 m inter-element spacing, was deployed on the bottom just outside the surf zone approximately 1.25 km away from the low frequency (J-15) source. Over the course of the experiment waves and currents buried the hydrophone array under a thin layer of sediments that was effective in keeping the array from moving under the influence of water motions, but remained sufficiently thin so as to allow for acoustic signals to reach the hydrophones. Throughout the experiment conductivity and temperature vs. depth (CTD) profiles were measured, which showed that the waveguide was iso-speed with sound speed $1510 \text{ m s}^{-1}$. Also deployed at the experiment site was a surface wave vector sensor, which recorded pressure and two horizontal components of fluid motion (PUV). From these measurements a full directional surface wave spectrum can be computed.

Beamforming was performed using a white noise-constrained (WNC) data-adaptive beamformer with the constraint set to 3 dB below 10 times \(\log_{10}(N)\), where \(N\) is the number of hydrophones used in the analysis. The array was deployed in such a way that it had aperture in two horizontal dimensions, and therefore searches in beamformer output could be performed over vertical angle with fixed azimuth without worry about side lobe contamination.

![Fig.3: Normalized beamformer output sweeping through vertical angle at a fixed azimuth for frequencies ranging from 534.5 – 535.5 Hz.](image)

Dominant acoustic angles of arrival are found as peaks in the non-Doppler shifted beamformer output (Fig. 3), and the associated mode numbers are computed from them using the measured sound speed. Scattered angles of arrival and Doppler shift values are inferred
from the beamformer output, as well. Though there is a left-right ambiguity in Eq. (4), the experiment’s proximity to shore removed this issue with the assumption that surface waves are not generated in-shore of the receiving array. While the array’s geometry easily allows for a full search in vertical angle at all azimuths, a simple scenario was desired for this study.

**DISCUSSION**

The more reliable inversion possibilities arise from the lower frequency acoustic tones (≤535) Hz in this study. This is due in part to there being fewer propagating modes at these lower frequencies, and therefore fewer scattered sidebands that can be mistakenly be associated with the wrong acoustic mode. However, due to the frequency dependence of the amplitudes of scattered sidebands on acoustic carrier frequency (from the derivative with respect to depth in Eq. (1)), the scattered sidebands are not as easily observed.

Higher order scattering complicates the process of inversion discussed in this paper by complicating the sideband structure and rendering the identification of sidebands associated with specific modes more difficult, if not impossible, especially when the surface wave field is comprised of multiple distinct systems. Multiple scattering of the acoustic waves by the rough, moving surface can lead to the same difficulty. Many mathematical treatments of scattering assume only a single interaction with the surface [3], but multiple scattering is possible, particularly in shallow water such as during the experiment discussed here.

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Towards Large Scale, Integrated Optical and Acoustic AUV Survey

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Abstract: The applications of underwater 3D reconstructions are wide-ranging from scientific use such as environmental surveys and underwater archeology, to commercial use such as drilling and pipelaying. Modern benthic surveying relies on a variety of different tools and sensors to produce quality seafloor maps. With the advances in Autonomous Underwater Vehicles (AUVs) and improved sonar technologies, very large detailed maps are being created. AUVs are also adept at collecting high resolution, near-bottom imagery. Data covering a particular survey area may now be generated from several sources each with its own strengths and weakness, tradeoffs in range and resolution. In this paper we present the integration of data from a ship based multibeam along with multibeam and stereo images gathered from an AUV. Work has been done of such integration on small scales, attempting to fuse a 2D mosaic with the acoustic bathymetry by Singh et al [3] but does not cover the area or volume of data described in this work. The novelty of the proposed technique is the ability to integrate the three datasets each at different resolution to produce one consistent model which captures not only high resolution images of the seafloor but sweeping geological features that span kilometers. The combination of scales provides a wealth of information and context to the viewer and allows for the synthesis of data sources that are traditionally viewed in isolation. We propose a technique for integrating and visualizing large scale 3D models consisting of optical and acoustic data from heterogenous sensors and platforms. For the purposes of this work we assume all the sources are georeferenced with a bounded amount of error. The georeferencing of these various platforms is a well studied open problem within the underwater community[1]. We use the visual Simultaneous Localization and Mapping system proposed by Mahon[2]. We present real data gathered by an AUV equipped with a high-resolution stereo imaging system and and low cost multibeam, profiling Sonar. This paper will show several visually consistent three-dimensional reconstruction from field deployments. REFERENCES [1] R. Eustice, H. Singh, J. Leonard, M. Walter, and R. Ballard. Visually navigating the RMS titanic with SLAM information filters. Proceedings of Robotics: Science and Systems, 2005. [2] I. Mahon,

APPLICATION OF LASER SCANNER TO SEAFLOOR ROUGHNESS MEASUREMENT AND FEATURE DETECTION

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Abstract: Structural light or laser line scanning has been widely used in robotics to measure features in the field for its simplicity and efficiency. The performance and accuracy of this methodology depends mainly on the calibration of the optical device which captures the image of the laser reflection. CCD camera is generally adopted as the input device. Conventional CCD camera calibration involves finding the optical and geometrical parameters of the camera and the environment in which it situates. A set of points with known location are needed as references for the calibration procedure. Whenever the scanner is moved or the CCD scanning head is ever disassembled for various reasons, this cumbersome calibration process needs to be done again. In other words, the calibration might not be difficult for indoor applications but it is very challenging for field works. In this paper a modified coordinate mapping calibration procedure is used. This approach, borrowing the idea from map projection, is easy to implement such that field operation is feasible. An underwater laser scanning system, based on this new calibration scheme, was developed and integrated with a deep water instrument frame. The combined system was deployed to a site 80 m deep to acquire the seafloor micro bathymetry in Shallow Water Experiment 2006 (SW06). The resolution of the laser scanner was high such that we can easily identify that the seafloor was full of shell hash and other benthic animals. The reflection of the laser line also changes its intensity as the texture of the target does. With post processing of the intensity, different features on the seafloor were detected. These functions of the laser scanner offer a new practical tool for field 3D measurement. The coverage ratio for SW06 was estimated to be 7-10%.

Keywords: Laser scanning, field measurement, seafloor shell identification

1. INTRODUCTION

There are mainly two different approaches to measure 3D relative coordinate in the field, namely point ranging and multi-perspective referencing. Laser point time-of-fly ranging is a typical example of the former approach, and stereo photography is a representative case of the
latter approach. Stereo photography technique does not need an auxiliary light source and is classified as passive vision. It works well unless the images acquired have only smoothly textured areas, repetitive structures or unclear images. In these cases, registration of target features is hard to achieve. It also suffers from poor illumination. To overcome this limitation and increase the image signal-to-noise ratio, another simple technique used in machine vision is to project structured light onto the scene and infer detailed information of various features from the distortion of the structured light in the image [1, 2, 3]. Active light sources are used, so it is classified as active vision. One of the structured light patterns commonly used is a light stripe generated by placing a cylindrical lens or a narrow slit in front of a laser source. The light stripe projected on the target leaves a trace on the surface. This trace is the cross-section of the at the scanned location. A CCD (couple charged device) camera can be placed at a proper distance with an oblique perspective angle to observe the deformed light stripe. The laser stripe seen in the CCD camera is similar to the silhouette of the target. The high contrast of the laser scan line in the image can be extracted by proper thresholding. It should be noted that the offset of the laser line is not necessarily proportional to the height of the target because the camera is pointed at an oblique angle to the target such that the image generally has some distortion. Therefore a calibration is needed to convert the laser scan line described in pixel coordinates into the actual dimension of the profile [4]. Applying this idea in the water, many systems have been developed to measure seafloor micro bathymetry [5, 6, 7, 8]. To have a calibration method which is simple enough to perform in the field is crucial for the feasibility of the system. In this paper, we propose not to treat the calibration as a problem of finding intrinsic and extrinsic parameters for the camera model, but instead as an empirical method in which the view seen by the CCD camera is split into smaller regions and local linear maps are built for each region. The details of this approach are described in [8]. We adopted this idea to develop a laser line scanner called Seafloor Laser Scanner (SLS) which was used in Shallow Water Experiment 2006 (SW06) to acquire seafloor roughness for modeling acoustic backscatter [9]. Data collected in this experiment demonstrate the performance of SLS in characterizing the detail dimension of the features on the seafloor. Moreover, post processing of the laser intensity reveals the difference in the texture of the features. In other words, it provides another piece of information to identify the location of different features on the seafloor. In this paper, we report the operation of SLS in SW06, and the data acquired in the experiment in Section 2. The post processing of the data to construct the 3D micro bathymetry and the intensity maps are presented in Section 3 and 4 respectively. Conclusions are made in Section 5.

2. DEPLOYMENT

The SLS consists of a water-proof laser line projector, CCD camera and PC control unit [9]. The scanning head assembly maintains the relative position and orientation between the camera and the line laser source. So during scanning, the laser sheet is always kept at the same location with respect to the camera image frame. The camera is a Basler A102fc CCD with a resolution 1388×1038 pixels. The wavelength of the laser is 650 nm, and a cylindrical lens is placed at its tip to generate a 60° fan angle.
In SW06, the scanning head assembly was mounted on the linear table of the \textit{In-situ Measurement Porosity 2} (IMP2). IMP2 was developed by APL, University of Washington to measure the roughness of the seafloor by a servo-control conductivity probe [10]. It has a linear table which provides a stable 4-meter translation track for SLS. The laser scanning assembly was kept roughly 75 cm above the seafloor surface to have an effective scanning swath about 30 cm. The CCD was tilted down to look at the laser reflection with a $30^\circ$ grazing angle. The linear velocity of the linear table is 2.3 mm/sec when it returns to its home position. The camera acquires images about 5 to 6 frames per second. With this frame rate, the scanning interval is about 0.5 mm along track. On R/V KNORR cruise August 5 to August 25, 2006, the integrated system was deployed three times. Two sets of seafloor micro bathymetry were retrieved successfully. The third trial failed due to the early termination of IMP2 probing such that the laser scanning system was not triggered. In the first deployment on August 14, 2006, the camera optical axis was aligned with the rail. With this setup, the CCD image was symmetrical about the centerline of the image frame. The quality of the image could be assured but the laser line swath did not cover the region where conductivity probe penetrated the seafloor. In the second trial on August 16, 2006, the camera optical axis was panned $30^\circ$ to the right intentionally. By doing this, the laser scanning swath covered the IMP2 probing marks but image were skewed. However, the results obtained by the two approaches could be compared.

3. MICRO BATHYMETRY

By extracting the bright laser reflection in the images and converting them from pixel coordinate to 3D coordinate in engineering unit, we can have dense 3D points to describe the seafloor with resolution 0.5 mm or finer. The shaded relief maps of the two scanning sites are shown in Fig. 1. In these two photo-like figures, we can find that the seafloors were abundant of features like starfishes, shells fragments and sediment mounds and pits. The seafloor, well below the surface wave, was free from the influence of wave motion. The seafloor showed no signs of disturbance by the surface waves, and no apparent ripple pattern can be found. On the other hand, the shell fragments were evidently lying on most part of the scanned seafloor. SLS's
ability to portray the details of the features on the seafloor can be demonstrated by a closeup of the SLS scanning on the first site shown as a 3D view in Fig. 2. In the figure, we can see that on the surface of three shells, there existed some form of barnacle encrustation. The dimension of the encrustation is as small as a couple of mm's. Another observation is the IMP2 probing marks on the relief map of the second site. The marks were captured by the laser scanning faithfully like a seam line in lower panel of Fig. 1. With a closeup of the line as shown in Fig. 3, we can find that each probing mark looked just like a stitch on the seafloor. When the probe lowered down, it created a pit with a diameter about 4 to 5 mm. As the probe retrieved, the sediment around the probe was piled up to form a mound around the pit like a crater. It was about 8 to 9 mm in diameter and a couple of mm in height. Center-to-center distance between pit holes was about 1 cm which was sampling interval set to IMP2. The roughness power spectra of the two sites were estimated and reported in [12].

4. REFLECTION INTENSITY MAP

The intensity of the laser reflection actually is not everywhere even. By cross-referring to the micro bathymetry, it can be found that the reflectivity of shell pieces is very different from that of the sediment. Therefore, it motivates us to use the differences in reflection intensity to identify and assess the number of shell pieces on the seafloor. In Fig. 4 an example of the relief map and its corresponding reflectivity image derived from the SLS data are shown. Constructing a reflectivity image is similar to that of a relief map but replacing the relief by the laser reflection intensity. We can find that it is easier to identify in the reflectivity image the
existence of starfish, shells and shell fragments scattered on the seafloor. The shell pieces in the figure were shown as having fine texture from increased brightness against the sandy background. Shell pieces as small as 2 mm in linear dimension are easily identifiable on the reflectivity image. The strong contrast in reflectivity offers the possibility of estimating the size and number distributions of shell pieces, both important inputs to model mid- to high-frequency acoustic backscatter.

To quantify the viability of using the laser images to estimate shell distributions, a laboratory experiment was performed to simulate the field experiment with controlled number and size distributions of shell fragments on the seafloor. A collection of shells and shell pieces were gathered from a beach along with sands having grain sizes between 100 to 200 μm. The shell fragments were sorted with sieves into six bins. A subset of the shells, proportional to the size distribution of all the shells collected, were used in the experiment. Details of the shell pieces are provided in Table 1.

<table>
<thead>
<tr>
<th>Size (mm)</th>
<th>&lt;1.4</th>
<th>1.4 - 2.36</th>
<th>2.36 - 2.8</th>
<th>2.8 - 3.35</th>
<th>16 - 20</th>
<th>21 - 25</th>
</tr>
</thead>
<tbody>
<tr>
<td>No. of shells</td>
<td>20</td>
<td>10</td>
<td>15</td>
<td>16</td>
<td>5</td>
<td>6</td>
</tr>
</tbody>
</table>

Table 1: Number and size distribution of shells used in the controlled laboratory experiment

The sample sediment was prepared in the following manner. First, the sand was poured into a 30 cm by 36 cm tray, and then smoothed flat. Next shell fragments from each size bin described in Table 1 were in turn placed on the sand surface. Their ordering and location were determined by the following procedure. Each shell was assigned with a unique number, and the
sand tray was divided into nine equal regions. One random number sequence was generated to be associated with the shell fragments, and another random number sequence was generated to decide which region on the tray to place each of the shell fragments, mimicking a Poisson process. Finally the shell fragments were tossed onto the designated regions randomly. Thus the orientation and the sides of the fragment were also random. This procedure was repeated until all the shells were allocated. Altogether, 67 shell pieces were properly placed on the sandy surface in the tray.

A small underwater linear track was specifically made for the laboratory measurement. The laser scanner together with the linear track was set with the same configuration as in SW06. The scanning head was kept about 75 cm above the seafloor such that it covered a swath 30 cm in width. The optical resolution in this configuration is about 0.3 mm, and the linear track was commanded to move at 2.0 mm/sec to have a compatible resolution along the track with the fastest frame rate (6-7 frame/sec) of the camera. The water depth on the SW06 site is 80 meters, a very dark environment even during day time. The lab scanning was carried out at night to achieve conditions similar to that at the field. In Figure 5, a reflectivity image from the lab scan is shown where the spatial resolution in horizontal dimensions is 0.3 mm. It is found from the lab measurement that the reflectivity of shell fragments is much greater than that from the sandy background. Shell fragments have several different colors, ranging from limestone gray, to orange, to dark charcoal. The laboratory experiment shows that colors of shells have no effect on the reflectivity for the green laser diode source used.

From the laboratory data, two scanning images were selected: one with and the other without shells in the scene. Then the reflectivity histograms from the two scanning were calculated and compared to choose an intensity threshold to distinguish the shells from the sand background.
The shell fragments have a narrowly distributed reflectivity around 250, which is much greater than that of sand with a mean at 50 and a maximum value less than 150. So intensity at 200 is chosen as the threshold to separate shells from sand. The intensity images after applying the threshold becomes a binary image as shown on the right panel of Fig. 5. A contiguous area where the pixels are above the threshold was designated as a shell, and its size was calculated by multiplying the number of pixels with unit area of a pixel, which is 0.25 mm$^2$. Only those shell pieces which have an area greater than 3 mm$^2$ can be reliably identified. Thus, smaller pieces were neglected. This simple procedure yielded a 100% correct identification of all 67 shell pieces in the laboratory test.

The same methodology was used to process the SW06 data. The smallest dimension of the shell fragment detected in the indoor experiment was 4 mm$^2$. Therefore, in addition to applying the intensity threshold as in the laboratory case, another condition was imposed – retain only those shell fragments of size > 4 mm$^2$ and treat smaller fragments as sand particles. It was estimated that the overall area coverage by shell fragments > 4 mm$^2$ was 8.53% for Site I and 6.10% for Site II. Goff et al. collected nearly 100 grab samples in an area approximately 8 km northwest of our experiment sites [11]. They report bottom samples collected in locations where shells predominate having a coarse fraction (> 4 mm) weight percentage of 5-18%. Though it is difficult to convert between coarse fraction weight percentage and the area coverage, their results agree qualitatively with those reported here.

5. CONCLUSION

In this paper, we report the development of an underwater laser scanner to acquire seafloor micro bathymetry. The CCD camera calibration algorithm is simplified substantially by incorporating the idea borrowed from map projection. With this improvement, the calibration can be done in the field before or after the system modules are assembled, and the operational cost of the system is reduced. The laser scanning head was integrated with an underwater linear stage which provided a stable linear track of 4 m long. The effective coverage of the seafloor was about 360 cm by 30 cm wide swath. Two micro bathymetry maps with mm-level resolution were acquired. On both maps, abundant shell fragments were discovered on the seafloor. The high resolution laser scanning was proved to be an excellent tool to depict the environment in

![Fig. 6: Shell coverage maps for Site I and II. Dark pixels are areas covered by shell pieces.](image_url)
great details. Moreover, laser reflectivity from shells was found to have higher values than that from sandy background. Therefore, it offers a straightforward method to find out the number and size distributions of shells on the seafloor by applying a simple threshold. This approach is validated by a controlled laboratory measurement.

we further and the shell fragments were identified. , the coverage ratio by shells was found to range from 7-10% for the SW06 seafloor, having a potential impact on bottom reflectivity and backscatter for certain sound frequencies.

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REFERENCE


Structured Session 20

Acoustic Data Fusion

Organizer: Eric Maillard
SONAR DATA SIMULATION & PERFORMANCE ASSESSMENT THROUGH TUBE RAY TRACING

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Abstract: Simulating realistic sonar data is crucial for tuning detection and classification algorithms according to environment and acquisition characteristics. Moreover, robustness of performances estimation and prediction applications can be greatly enhanced as soon as such a simulation tool provides both a modular underwater world representation (multiple sensors, environments and acquisition conditions) and a selection of several computational engines (ray theory, parabolic equation ...). Therefore, we developed such a framework for simulators, allowing both scene design and computational engine choice. Within it, a tube engine has been successfully implemented and realistic simulations obtained. Moreover, the engine keeps track of the full propagation history of the simulated acoustic wave allowing mission planning & debriefing to take benefit from this knowledge for environment assessment applications (mine hunting, for instance). Indeed, sea bottom regions non-observed during a survey (either planned or performed) can be computed, outlining areas of potential threats, for instance. Thus, bringing that non-statistical information to CAD/CAC systems may enhance the reliability of their outputs.

Keywords: sonar, simulation, acoustic propagation, minesweeping, mission planning, mission debriefing,
1. INTRODUCTION

Due to operational constraints for underwater data acquisition, simulating realistic sonar data, like images, swath bathymetry profiles and now interferometric signals, is crucial for tuning detection and classification algorithms according to sensors settings and sea-bottom nature and topography. Moreover, the robustness of any performances estimation or prediction can be improved, as soon as such a simulation tool provides a modular and flexible underwater world representation (multiple sensors, environments and acquisition conditions) along with various computational engines that may be used depending on simulation cases (ray theory, normal mode or parabolic equation, all solving the Helmholtz equation with specific approximations). Therefore, we developed such a framework for simulators, allowing both scene design and computational engine choice. In the presented work, the underwater virtual environment is composed of various items like environmental characteristics, sonar configurations and trajectories, seabed topography and nature. The computational engine is based on acoustic tubes launched from the sonar transmitting antenna; this engine preserves the full history of acoustic tube propagation paths within the environment including all the interactions with scene elements. Keeping all this propagation history allows building backscattered signals recorded by the receiving antennas. Depending on the sensors characteristics and the building algorithm, several sensors can be simulated (sidescan sonar, multibeam echosounders, front looking sonar, ...) and the results (images, sea-bottom profiles, ...) compared with reality show realistic simulated outputs.

As the propagation history gathers all the information about the acoustic wave interaction with the scene, mission planning and debriefing may use this knowledge for environment assessment applications (minesweeping, for instance). Thus, we propose an extra simulation tool that computes for a survey (planned or actually performed), the sea-bottom areas observed by the sensors. Indeed, determining non-observed regions during the survey, outlines areas with possible undetected threats. Furthermore, a classification process, either human or automatic, may take benefit from knowing the analyzed area has been observed several times and considering the resulting multiple views. Finally, it also brings non-statistical effective information within CAD/CAC systems, enhancing the reliability of their outputs.

2. A SONAR DATA SIMULATION ENVIRONMENT

2.1. Simulation framework & software components

The main goal of this simulation framework is to propose a common software environment for developing modular simulation tools. The keyword “modular” stands for both the simulation objects describing an underwater scene and the algorithms developed to deal with acoustic propagation and object interactions (reflexion, refraction ...). Thus, the user picks among several families of components to tune its simulation environment. As shown in Fig. 1, some of these software components are relative to static properties of the underwater scene while others are meant to represent dynamic phenomena. The “scene” components family gathers all the objects (wrecks, containers, sea bottom elevation maps ...) belonging to
the scene, with their geometric descriptions either through mathematical definitions or lists of facets. It also includes all the characteristics of the propagation media (sound velocity profile ...), the list of sensors involved in the simulation with their intrinsic properties and their dynamic behaviors (trajectories, attitudes ...). The “propagation” package keeps all the models available for simulating acoustic wave propagation within a medium, while all the engines from the “engine” package rely on both a scene description and a propagation model to propose a full acoustic simulation. An engine’s main goal is to generate a propagation history from the transmitting to the receiving sensor, gathering along the propagation path, all the interactions between the acoustic wave and the scene elements. Preserving such a propagation history allows each receiving sensor to aggregate all the incoming contributions to produce its specific sonar simulated data [1].

Fig. 1: Simulation software components.

2.2. Tube tracing engine

Within the previous framework, a tube engine has been developed and used for simulating sidescan sonar images. This engine represents acoustic wave propagation as a series of rays always orthogonal to the current local wave front. The solid angular range of such a wave, also called sector, depends on the aperture properties of the simulated antenna. Within this sector, tubes are built on the top of neighboring rays using four of them as borders, as shown in Fig. 2. The intersection between a tube and a scene element defines a tube footprint. If this footprint is totally included in one facet of an element, it becomes an elementary footprint $E_C$. With these definitions, the basic tube engine algorithm consists in a 3-step processing:

- The transmitting sector, defined by the Tx sensor properties, is split as long as it contains non-elementary tubes
- The same splitting process is performed for the receiving Rx sensor.
- For each intersection between an elementary Rx tube and an elementary Tx tube, the corresponding footprint is computed allowing the receiver to properly aggregate the transmitting contribution after its interaction with the intersected facet.

In case of multiple paths simulation, all the footprints belonging to a facet are gathered in order for this facet to become a new transmitter. The amount of levels in this recursive loop defines the lengths of multiple paths to consider.
The strong key feature of this algorithm is to independently compute the full acoustic wave propagation history so that each receiving sensor uses this history to output data. In this paper, only images resulting from a simulated sidescan sonar will be exhibited, but starting from the computed history, any sensor output can be produced assuming a sensor-specific aggregation algorithm has been provided.

Fig. 2: Tube tracing mechanism with scene interaction (left) and full propagation from the transmitting to the receiving antenna where tube aggregations are performed (right).

2.3. Simulation results

This engine has already been used for several dedicated simulations including front-looking sonar and multibeam echosounders. However as the application proposed in this paper concerns minesweeping mission planning, results will focus on sonar image simulations.

Fig. 3: Simulated sidescan sonar images using an oil well head 3D model (left) and a Digital Terrain Model built from bathymetric data (right).

Fig. 3 presents such simulated images showing the capabilities of the simulator either for imaging complex objects or elevation described sea bottoms. On the left part, the scene consists in an oil well head lying on a flat sandy bottom and has been observed through 3
different viewpoints. The oil well head geometry comes from a “Wavefront” file giving its edges and vertices. On the right part, the scene only consists in a Digital Terrain Model built from multibeam echosounders bathymetric measurements in the Sidney harbour [2]. The features (small objects and wrecks) occurring on the bathymetric data also appear on the simulated image.

In order to check the quality of our simulations, simulated images have been compared with real sonar images using the same acquisition conditions (sensor trajectory & attitude), with the help of data collected for the “Shallow Survey ‘99” conference in Sydney harbour: a Klein 5400 for the sonar images and a RESON 8101 for the bathymetric data. The left part of Fig. 4 describes the methodology that consists in comparing real sonar images with those produced by our simulator fed with all the recorded survey information but sonar images. The final expectation is to find on the simulated images, all the features appearing on the real ones.

![Fig. 4: Comparing simulated images with effective sidescan images.](image)

That is the case as confirmed by a quick look at the right part of Fig. 4: both wrecks can be identified, two other spots corresponding to impacts (purple plain circle) or objects (orange dotted circle) also match, even fine sea-bottom marks appear on the simulated images confirming the relevancy of such a simulation. The features smoothing appearing on the simulated data can be explained by the 1.5m x 1.5m grid resolution of the built DTM, compared to the KLEIN 5400 pixel resolution (a few centimetres). Indeed, for our simulated sidescan sonar, the smallest observed object is one DTM cell. Some ripples artefacts also appear on the simulated images because of a DTM building process performed on not very well-calibrated MBES data. That introduced a few centimetres height difference between XYZ points coming from end-of-range overlapping tracks, leading to false ripples [3].

### 3. APPLICATION TO MISSION PLANNING & DEBRIEFING

One criterion for estimating the quality of any survey is to determine the proportion of sea bottom effectively observed by the sensors, during the survey, in relation the theoretical sensors coverage. We propose to use the propagation history built for the sonar simulation in order to compute the effective sensors coverage within the simulated environment that represents the survey to plan or to debrief. Indeed, depending on the source of the trajectories feeding the simulation, either mission planning (with planned trajectories) or mission debriefing (with effective trajectories) can be performed.
3.1. Effective coverage computation

The propagation history from one transmitting sensor to one receiving sensor preserves all the elementary tube footprints that correspond to the effectively observed portions of the scene. When aggregating all these polygonal intersections, the proposed algorithm produces a vector description of the observed areas.

Fig. 5: Multi-sensor mission planning (2 sidescan & 1 front looking sonars).

Fig. 5 shows an example of the graphical user interface developed for the algorithm. On the left side, it shows all the entities involved in the simulation: the used sensors with their characteristics and their trajectories, the sea bottom elevation map and the sound velocity profile. On the right side, a geographical representation of the surveyed area is proposed with the trajectories to follow (grey lines starting by a grey cross) and the color-coded sea bottom elevation map, from blue for the deeper areas to red for the shallower regions. The current status of the computation is also displayed with green lines for the portion of trajectories already performed and a semi-transparent overlay of gray and purple surfaces. The grey surfaces correspond to the regions effectively observed by the sensors while the purple ones indicate the sensor coverage holes i.e. the regions not observed by the sensors. We observed on the figure that these holes logically appear when higher elevation seabed elements create acoustic shadows in relation to the sonar location.

As the sonar follows its path, these gray and purple surfaces are updated in order to give at the end, a full report on the sensor coverage efficiency including the theoretical coverage surface, the effectively observed surface and the induced coverage ratio along with the description of all the coverage holes. The left side of Fig. 6 presents such a report.

The right side of Fig. 6 illustrates the impact of the sound velocity profile in a survey effective coverage. The same survey simulation has been performed with two different sound velocity profiles (shown at the right bottom of the figures): the first one with smooth variations, the other one with gradient brutal changes. The “thermocline” phenomenon of the second profile logically involves an effective coverage decrease, for the same survey, compared to the first profile [4].
3.2. Multi-tracks aggregation

As a survey consists of several tracks, the proposed algorithm also permits to merge the coverage computed for each track into a global survey coverage. Depending on the final goal, several merging operators are proposed. The “or” operator computes the regions observed at least once during the survey; this is the basic operator. Indeed, when planning a mission, the surfaces of the remaining coverage holes have to be minimized, and when performing a survey debriefing, it permits to quantify the non-observed areas that may contain potential threats. The “and” operator allows the user to compute the regions being observed by all the selected tracks. This tool may be useful when dealing with difficult sea bottoms where classifying objects based only on one point of view, is not enough and knowing several shots are available can greatly enhance classification reliability. As for individual tracks, coverage statistics are also computed for the global survey taking into account the chosen merging operator.

Fig. 8 proposes such a scenario with a survey made of two tracks A and B. The upper images show each track’s individual coverage. The lower images give the aggregated coverage for the “or” and “and” merging operators. About the “or” operator, the grey surface represent the areas observed at least once while all the remaining purple areas are full survey coverage holes. Concerning the “and” operator, the gray surfaces indicate where the sea bottom has been observed by the two tracks: in these areas, objects to classify have been seen twice.
4. CONCLUSION & PERSPECTIVES

This paper proposes an extension to our sonar data simulation for mission planning and debriefing purposes. Indeed, the propagation history preservation, key feature of our simulator, allows accurate measures of sea bottom sensors footprints and their associated coverage holes geometries giving us one useful criterion for measuring a survey quality. By embedding this measure into a multi-criteria optimization process like genetic algorithms with variable length chromosomes representing portions of trajectories [5], optimal survey paths may be proposed in order to minimize coverage holes while respecting specific navigation constraints for trajectories, systems energy autonomy capabilities...

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Automated acoustic data fusion combining neural networks and Dempster-Shafer theory

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Abstract: Seabed classification is a key issue for civilian and military underwater applications, from offshore exploitation to mine counter measure. The current automated classification techniques mostly rely on the analysis of the data provided by a single sensor, supposed to unambiguously separate the different classes of seabed. In this paper we present a different approach which considers that, even if a sensor cannot tell the differences between two classes, classification will improve by considering that the seabed ambiguously belongs to one of these two classes, and, further, that the analysis of the data from another sensor can resolve the ambiguity. For each sensor, the classification is achieved in a conventional way by feature extraction and neural network based supervised classification. The fusion of the neural network results is achieved using evidential reasoning. After a description of the method, the paper discusses the experimental results from the fusion of information delivered by towed acoustic sensors: imaging sidescan sonar, vertical echo sounder and interferometric bathymetric sidescan sonar. The final part of the paper deals with the extension of this concept to an autonomous platform called DAURADE designed by DGA and SHOM to simultaneously operate five acoustic sensors: dual frequency echo sounder, sub bottom profiler, sidescan sonar, multibeam echo sounder, high frequency sector scan sonar.
Gaining insight to Acoustic Measurements through the fusion of multisource data

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Abstract: Scientists and engineers alike are faced with the challenge of verifying the results of their measurements through either visual groundtruthing (i.e. cameras, net-tows, etc), calibration and/or by comparing measurements taken by multiple devices (i.e. imaging sonars, split beam echosounders, multibeam echosounders, etc). Typically the temporal, visual and analytical components of data integration are not handled well within a single software application. Eonfusion is four-dimensional (4D) visualization and analysis software which significantly enhances the ease with which scientists can integrate complex ecological and environmental data, and share methods across disciplines. It handles large volumes of data in a variety of formats, and supports the fusion of different data types. Eonfusion bridges specialist domains such as bioacoustics and multibeam acoustics and enables the analyst to rapidly visualize trends and patterns and pose novel questions regarding feature relationships in an integrated context. By supporting coincident visualization and analysis of, for example time- and geo-referenced sensor data, bathymetry, video and bioacoustics data, Eonfusion facilitates the validation of hypotheses surrounding multidisciplinary/multisource observations. Multiple data sets can be fused into a single set with shared coordinates, enabling the discovery of topological relationships between coincident data items. These relationships also allow data attributes to be directly compared and transferred between data sets. In this paper we present real-world examples of fishery, habitat mapping and oceanographic data which are integrated and visualized in Eonfusion to provide insight to multi-source measurements as they relate to marine species and their environment.

Keywords: multisource, fusion, 4-dimensional, analysis, visualization, interactive

INTRODUCTION

The technological evolution of instrumentation and its associated platforms is quickly improving the quality and detail of the data which engineers and scientists use to study the oceans [1]. Hardware design improvements have resulted in increased sampling rates, lower power requirements and higher data volumes with finer resolutions. While these advances in
hardware have expanded the spatial context in which data may be acquired, the ability to integrate data from multiple sources as well as analyze and visualize data has limited many users to using a combination of 2-dimensional Geographical Information Systems with signal processing packages and, on occasion, 3-dimensional visualization solutions. While these methods have led to some amazing discoveries, they are time-consuming and often result in only a subset of an entire dataset being analyzed. This has led many to ask “How do we quickly and easily find the moments in our multi-variant data sets that represent the important events and quickly get them into publications?”

In 2003 researchers and engineers from Myriax began the Eonfusion project whose primary goal was to provide a closely coupled data integration, analysis and visualization research tool designed for people working with large volumes of spatially and temporally variant data sets. The result of this effort is now available as a commercial-off-the-shelf software solution. Eonfusion provides data fusion of complex environmental data sets originating from a wide variety of formats and displays them in a 4-dimensional graphical space. It allows users to perform intuitive multi-dimensional time-series analysis without limitations based upon scale or source, and offers spectacular data visualization to communicate research outcomes.

**CORE TECHNOLOGY**

Integrating data from different sources required Eonfusion to seamlessly incorporate data that varies both temporally & spatially into a scale neutral, high graphics environment. It also required the integration of data sets with different sampling rates, different sampling times and a variety of geospatial coordinate systems. Additional tools were needed to allow the user to logically keep track of their work, quickly perform standard data manipulation & transformation routines, transfer attributes between data sets. And finally the option for the more technical user to incorporate their own algorithms was highly desirable. Out of these requirements, five main product feature requirements evolved [1].

**Evolving the 4th-Dimension**

In Eonfusion, time data is seamlessly treated as a vertex attribute in the data structure and as a fourth coordinate in its visualization space. This allows the user to integrate, quantify and explore data across temporal space as easily as one would across the geospatial (x, y and z). An adjustable ‘time-slider’ feature enables unrestricted time-scaling, allowing the user to quickly move forward or backward through time and adjust the visible time scale from picoseconds to eons.

The four dimensional capabilities of Eonfusion's visualizations are not limited to x, y, z and time. They readily accommodate any combination of four attributes that exist within data, and any parameter may be mapped to the time slider. This flexibility yields a simple yet powerful means of cycling dynamically between upper and lower parameter ranges in order to resolve dependencies between variables.

**Scale-neutrality**

Eonfusion supports data sampled over a continuum of scales. For example, a full world-scale bathymetry at multi-kilometer resolution can be visualized simultaneously with sub-meter scale data acquired at high and low frequency sampling rates, while maintaining the resolution of the original data sets.

**Data fusion**
Eonfusion not only enables multiple data sets to be viewed coincidently in space and time but also provides the ability to migrate attributes from one dataset to another. Eonfusion provides this through vector fusion and raster attribute transfer. Vector fusion allows vector data sets to be fused together, resulting in a combined superset of their vertices which are linked to their original dataset and copied forward to a new dataset. Raster attribute transfer allows attributes to be migrated from rasters onto the vertices of vector data. This transfer applies in space and time, so that an attribute from a time series of rasters can be transferred to time series data points.

**Intuitive ‘data flow’ environment**

In Eonfusion, users can design and build a flow chart of graphical elements defining data inputs, transformations, fusion algorithms, 4D output scenes and exported data sets. Attributes can be transferred between data sets so that a single fused data set contains all of the critical attributes of interest. Completed data flows can be exported as data processing methodologies in XML in order to ensure proper quality control and/or shared with other Eonfusion users.

![Fig. 1. Data flow operators are intuitively linked in a graphical interface to develop methods accessing sophisticated data integration capabilities. Rugosity, seafloor backscatter and bathymetric (raster) data files are elegantly integrated with track (vector) data using Eonfusion’s graphical data flow methods.](image)

**User customization**

While Eonfusion possesses a series of operators for data transformation, etc, it can also be readily customized to fit the needs of any user. This customization is achieved through an integrated C# coding environment (the Expression Evaluator) for algorithm implementation and an application programming interface (API) that facilities the development of modular extensions.

**INTERACTIVE GROUNDTRUTHING**

Eonfusion’s approach to dramatically reducing the effort through the aforementioned features provides an interactive and comprehensive quality control across sampling platforms. To assess the differences between existing methodology and improvements structured within Eonfusion, media, raster and vector based data supplied by NOAA’s Biogeographical Branch was integrated into the software (Fig 1). Raster data included;
multibeam derived bathymetry, calculated surface roughness (Rugosity) data and seabed backscatter. Vector data sets included ArcView shape files containing habitat classified polygons as well as tabular text data containing ROV positions, times and corresponding habitat classification data from the Buck Island Reef National Monument [2]. To quantify the effectiveness of the rugosity calculation and the relationship between acoustic backscatter and habitat the raster information was transferred to the vertices of the ROV track. Within the visualization space, a 2-dimensional graph showing both raster values as a function of time are displayed while the data points & track of the ROV may be colored to match that of a seabed classification scheme. Furthermore, through the Expression Evaluator users can perform a statistical analysis between the two data sets to further evaluate the effectiveness of satellite based algorithms or, if desired, directly incorporate their algorithms into the dataflow to derive additional parameters. During the ground-truthing process, the user attaches a “double-ended probe” to the ROV track (Fig 2) which is dynamically link to a media view of the corresponding video. As either end of the probe is dragged along the track, the video is automatically updated to that point in time and space of the ROV position. The user can then define the substrate between two points and publish the quantified information to a database. Once completed, since this information has been fused with the backscatter, rugosity and habitat classified information via the vertices of the ROV track, all data can then be used to reclassify the polygons or simply written out to a table for further analysis (Table 1).

Figure 2. Multibeam Digital Terrain Model (DTM) color-coded based on surface roughness (Rugosity). The trackline of a video equipped ROV is fused with the raster data along with the media. “Tags” have been attached to the vector data which provides a visual quantification of the habitat which is then linked back to the database (data provided by NOAA NCCOS CCMA - Biogeography Branch)
Table 1. Output results from ground-truthing of the multibeam derived products. Information includes vertex information for position, user classified habitat information as well as the raster rugosity and backscatter information that intersect the vertices.

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BRIDGING DISCIPLINES

The accessibility of bathymetric and associated benthic habitat data through multidisciplinary projects promotes the bridging between the biological and physical disciplines. For example, Dr. Hugh Pederson of the Tasmanian Aquaculture and Fisheries Institute (TAFI), University of Tasmania, was posed with the challenged to address frequent anecdotal reports that fishing activity had altered the behavior of rock lobster stocks in eastern Tasmania. His research sought to examine differences in the movement, behavior and habitat utilization of rock lobsters exposed to fishing activity and those protected inside a Marine Protected Area (MPA). Through the use of the habitat information provided to the MPA, the study also tested hypothesis as to the absence of large rock lobsters in an adjacent fishery [3].

During the study over 180 lobsters were tagged with acoustic transmitters and their movements were monitored using a Vemco Radio Acoustic Positioning (VRAP) system. Over the course of 18 months of study more than 3 million positions were recorded. This information along with habitat polygons, meteorological data bathymetric data was integrated into Eonfusion. Clear population differences were observed between the marine reserve and fished areas including variation in patterns of movement, population density and size structure. These differences manifest in the timing of lobster movement, visualized using Eonfusion’s unique time slider control.

To quantify patterns of movement based on light, filtering of day and night time movement data was performed within the software and a “sun-moon widget” was be used to visualize solar and lunar phases. Preliminary results show that within the no-take marine reserves, researchers observed that lobsters are more active during daylight hours and demonstrate a greater range of movement than lobsters in fished zones, where the population is more active during the night. Research will now focus on understanding the patterns of habitat utilization and the impact of environmental variables on lobster behavior.

FUTURE PROOF

As technology evolves of methodology, our insight into the oceans will only become more comprehensive. However the requirements for integrating, analyses & managing the vast amounts of archived and near-real time data will become more demanding. Eonfusion is a solution already in place to help minimize the complexity of data management integration while opening the doors for analysis and visualization. While the user may choose to use it as a stand-alone solution, Myriax provides the option to directly integrate Eonfusion with
your existing GIS and/or signal processing package. Utilizing its API, users may further expand the functionality of their current solutions by providing a direct interface to virtually any data format currently in use. Furthermore, its customized memory management system is designed to take advantage of multi-core processing, high-end graphics cards and 64-bit operating systems not only allowing for fusion of data sets from multiple sources but also for integration of high volume data sets such as multibeam bathymetry, LiDAR and multi-year time series data. The visualizers themselves allow for attributes to be expressed by not only color but by sphere/line radius, translucent halos, surfaces, etc. As we move forward, users will discover an immersive experience discovering relationships that will evolve along with the technology.

**ACKNOWLEDGEMENTS**

Data from Buck Island Reef National Monument courtesy of provided by NOAA NCCOS CCMA - Biogeography Branch, http://ccma.nos.noaa.gov/about/biogeography/

Data for Lobster Tracking Study Courtesy of Hugh Pederson, Tasmanian Aquaculture and Fisheries Institute

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Semi-automated classification of multibeam SoNAR data in the U.S. Virgin Islands

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Abstract: Benthic habitat maps of coral reef ecosystems support multiple resource management objectives, including understanding and predicting the spatial distribution of resources, detecting environmental change, and supporting spatially-explicit decision making. Marine habitats deeper than 30 meters have been successfully characterized by conducting heads-up digitizing and interpretation of acoustic imagery acquired using multibeam echosounders (MBES). These resulting maps, however, are subjective and ultimately irreproducible because they depend on the accuracy and interpretation of the person that is digitizing. Here, we semi-automate the seafloor feature extraction and classification process using underwater video, bathymetry, backscatter and a suite of morphometrics derived from high-resolution MBES datasets collected off the coast of St. John in the U.S. Virgin Islands. A new hybrid classification approach was developed, combining object-based segmentation with pixel-based classification and regression trees (CART), to identify and extract seafloor features at high spatial (2 x 2 m) and thematic (≤32 unique classes) resolutions. Our analysis suggests that this new technique is also robust in dealing with dataset errors and artifacts. The ability to quickly and objectively create benthic habitat maps would allow scientists and resource managers to better quantify and assess the changing health of mid to deep-water coral reef ecosystems. http://www.ccma.nos.noaa.gov/ecosystems/coralreef/benthic_usvi.html
COMBINATION OF BATHYMETRY AND CALIBRATED IMAGERY IN MULTIBEAM ECHOSOUNDER

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Abstract: The scientific community and advanced users have a need for accurately calibrated measurements of acoustic properties of the sea bed. The predominant sensors for underwater survey are based on sonar technology. In this category of remote sensing devices, the latest generation of multibeam echosounder provides high-resolution bathymetry and imagery. This combination offers a unique opportunity to improve the accuracy of the final measurement by the inherent co-localization of the two components. In this paper we present an implementation of backscatter computation that includes a calibration procedure and takes advantage of bathymetry knowledge. It demonstrates that, while a variation of imagery intensity can result either from the topology of the seabed or from a change in the sediment composition, the bathymetry provides a means to separate these causes and compensate for topology. Once all the sonar parameters are carefully measured during a calibration routine and accounted for during the backscatter estimation, only the characteristics of the sediment are present in the resulting data set. This calibration procedure is a critical step of the process as any error during that operation will translate in a loss of accuracy in the final results. The work presented in this paper targets a high-resolution high-frequency system. After the necessary information about this system has been collected during calibration, calibrated backscatter is computed in post-processing on raw data collected during the survey. A test is performed to verify the validity of the implementation and a sample survey is conducted for final validation.

Keywords: multibeam echosounder, bathymetry, imagery, calibrated backscatter, DTM, mosaic
1. INTRODUCTION

Calibrated backscatter is an important source of information to identify the type of sediment present on the sea floor. Single beam echo sounder and sidescan sonar were the primary sensors used to collect acoustic data that can be processed to determine the sea bed type [1]. Lately, the MultiBeam EchoSounder (MBES) has been added to this set of sensor. With its wide swath and imaging capabilities, a MBES combines the benefit of both above mentioned sensors. It is also faced with the same set of challenges.

The strength of the acoustic signal reverberated by the sea floor greatly depends on the type of material constituting the upper layer of the sediment. Unfortunately this strength also depends on the angle at which the signal hits the seabed, the frequency of the signal and many other parameters. Last but not least the propagation of the signal through the medium is subject to various losses proportional to the travel time.

The goal of calibrating the imagery data is to remove as many influences as possible from the signal strength. This results in a simpler relationship between this signal and the type of sediment causing the reverberation. Automatic tools can then be applied reliably to perform seabed classification.

In the remaining of this paper we detail how the calibration of the output signal is performed though careful accounting of the various sonar parameters affecting the signal levels. The effect of the bottom topology on the signal is described. Finally some experimental data collected with a high-resolution MBES is presented to illustrate the application of this processing technique.

2. SONAR CALIBRATION

The goal of this calibration is to guarantee that the sound pressure at the face of the receiver array is linearly translated into an output signal level. The absolute calibration further ensures that the actual sound pressure is truly represented in the digital format of the output signal.

First the arrays responses are measured in a calibration facility to determine their characteristics such as open circuit responses of the receiver and transmitting voltage responses of the projector. For example, a typical beampattern for the type of projector used in the experimental part of this study is presented in Fig. 1. One benefit of the small size of this type of projector is that the actual response of the transmitting array can be measure in a tank. For larger system operating at lower frequencies this array beampattern must be modelled from the actual response of individual elements of the array. While not presented here, the same procedure is also performed for the receiver array [2].
The dynamic response of the electronic transceiver must be linear to preserve the relationship between acoustic sound pressure and signal levels. The characteristics of each channel are measure separately. A typical channel dynamic response is presented in Fig. 2. Receiver saturation is presented by the black line. It corresponds to a combination of very high input signal with large gain. This situation is easily avoided in survey conditions.
The relative variations in amplitude and phase of the response must also be compensated. These variations can affect the results of the beamforming process. A complex weighting of the channel signals during beamforming is applied to equalize these channel responses. In a nominal situation all channels are quite similar so this equalization is not critical. Fig. 3 presents an example of channel failure and severe characteristics variations. If this system were to be used in a survey operation, failure to account for the deficiencies of the receiver would result in degraded performances.

![Channel complex response](image)

**Fig. 3: Channel complex response**

3. FROM EQUALIZED SIGNAL TO BACKSCATTER STRENGTH

The first step in estimating the sea bed backscattering strength consists in computing the reverberation level according to [3]:

\[
EL = SL - 2TL + TS
\]

where \( SL \) is source level, \( TL \) is the one way transmission loss, and \( TS \) is the target strength.

From this reverberation, the echo level can be expressed as:

\[
EL_\theta = BL_\theta - BG - D_{RH}(\theta + \theta_{Roll}) - D_{RV}(\theta + \theta_{Pitch}) - OCR - G(g)
\]

where \( \theta \) is the bearing of the beam, \( BL_\theta \) is the magnitude for each beam waveform, \( OCR \) is the mean open circuit response, \( G(g) \) is the applied gain as a function of the operator selected...
gain $g$, $D_{RH}(\theta)$ and $D_{RV}(\theta)$ are the normalized beam pattern in the horizontal (across-track) and vertical (along-track) directions respectively, $BG$ is the beamformer gain.

The backscatter is related to the target strength according to:

$$BS = TS - 10 \log A$$

where $A$ is the scattering area that can be approximated as:

$$A = \min \left( \frac{c.T}{2 \sin \phi} \cdot \frac{\varphi \cdot R^2}{\cos \phi} \right)$$

where: $\varphi_V$ is the transmitter along track opening angle, $\varphi_H$ is the receiver across track opening angle, $\phi$ is the incidence angle, $c$ is the speed of sound and $R$ is the range to the seabed.

Finally, the backscatter strength must be corrected for the incidence angle. One simple approach consists in applying a Lambert law:

$$BS = BS_o + 20 \log(\cos \phi)$$

where $BS_o$ is the backscattering strength at Nadir.

4. INFLUENCE OF TOPOLOGY ON BACKSCATTER STRENGTH

From the previous section we can see that the topology contributes greatly to variations of the signal levels that result in backscatter strength [4]. The reverberation area (4) is a function of the incidence angle. This incidence angle is a combination of beam steering, array attitude and slope of the seabed at the location of the sounding. The Lambert law is also a function of the incidence angle. The variation of the signal level as a function of incidence angle is presented in Fig. 4. Once a proper grazing angle correction is applied the signal variations are due to seabed scattering properties (see Fig. 5). It should be noted that the data presented in Fig. 4 and Fig. 5 is affected by large vessel motion as visible on the ping-to-ping variation of the swath location.

Fig. 4: Un-corrected imagery data
The correction presented above illustrates the effect of the incidence angle on the signal strength. By nature MBES systems perfectly co-locate the bottom topology and imagery. It is thus possible to apply a perfect correction. It should be noted that this correction is only possible in post-processing or with a short delay as the topology estimation involves multiple pings.

5. EXPERIMENTAL RESULTS

Experimental data was collected off the coast of Santa Barbara on a dramatic feature of the seabed. The topology of the “One Mile Reef” is presented in Fig. 6.
Snippet imagery was collected simultaneously with the bathymetry. Each snippet corresponds to a short time series of the signal magnitude centred on the bottom detection sample. From these snippets and the topology of the seabed, a mosaic of the backscatter is built. This mosaic is presented in Fig. 7.

![Fig. 7: backscatter mosaic for the One Mile Reef](image)

The various levels of backscattering strength are clearly visible in this figure. While the variations on the reef are to be expected since it is likely that the sediment on this feature is different from the surrounding sand, some unexpected weaker spots are also present in this data. A bottom classification algorithm would have to be applied to identify the nature of this sediment.

6. CONCLUSION

The MBES offers an invaluable combination of bathymetry and imagery data from a single source. These two data sets are naturally co-located so processing of the imagery to generate mosaics of backscatter strength can achieve a high level of accuracy by precise estimation of the acoustic wave incident angle. The absolute backscatter strength measurement does require additional parameters such as detailed sonar acoustic and signal processing characteristics.
While this processing adds to the computation of a standard mosaic, it is an important operation for applications such as habitat mapping. When the overall process is performed to the highest standards of operation, the final calibrated backscatter can achieve a level of absolute accuracy on the order of ±1dB.

7. ACKNOWLEDGEMENTS

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REFERENCES


Structured Session 21

Advances in Visualization and Quantification of Acoustic Beamformed Data

Organizers: Chris Malzone & Mike Mutchler
Atlantic Herring Low Frequency Target Strength Estimation from OAWRS Data in the Gulf of Maine over 10 days of Observation


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Abstract: During the Gulf of Maine Acoustic Experiment in Fall 2006, massive shoals of Atlantic herring were instantaneously imaged over wide areas using an ocean acoustics waveguide remote sensing (OAWRS) system during evening to midnight hours over a period of 10 days from Sep 26 to Oct 5. The low frequency target strength (TS) and abundances of the herring population have been estimated by correlating the OAWRS data with localized measurements from a conventional fish finding sonar (CFFS) for 3 days from Oct 1-3 [Gong et. al. JASA, Vol 124, p 2586]. Here we provide an analysis of the low frequency TS of the herring population in the frequency range from 300 to 1500 Hz and abundances for the remaining 7 days of the experiment. The acoustic scattering from herring populations is highly frequency dependent and is well modelled using a resonant scattering model for swimbladder bearing fish. Here we compare the TS estimates and the neutral buoyancy depth for herring over the observation period of 10 days.

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Moving beyond the 4th Dimension of Quantifying, Analyzing and Visualizing Acoustic Beamformed Data

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Abstract: A significant challenge facing researchers is the bridging of domains between the natural and physical sciences through the integration, analysis & visualization of multivariate, multisource data. For example integrating scientific echosounder data with that of an imaging sonar and relating those measurements with tracking information from an animal in a high graphics space is a challenge for even the most talented of analyst. This is due to the fact that quantifying the temporal, visual and analytical components on an integrated dataset has not been possible within single software tool suite, and the final products are often difficult to share with collaborating researchers and managers. To address these issues, Myriax launched the Eonfusion project that has resulted in an intuitive and readily extensible software solution that provides researchers a state-of-the-art 4D-analysis environment. This software solution significantly enhances the ease with which scientists can integrate diverse data types (raster, vector & media), quantify trends and share methods across disciplines. The powerful 4D-visualization interface also incorporates a range of analysis and plotting tools as well as a time-stepping feature that allows users to explore the evolution of spatial structure and resolved relationships over time. Its revolutionary fusion operator allows data at different temporal or spatial scales to be reconciled, attributes merged and topological relationships identified. The software is customizable through an integrated coding environment using C# for simple formula implementation or an API facilitating the development of advanced modules and algorithms. It is ideal for both undertaking complex analyses and communicating syntheses. In this paper we present real-world examples that show the integration of bioacoustics, swath echosounder and biological tracking data which are integrated and visualized in Eonfusion to reveal integrated relationships.

Keywords: fusion, 4-dimensional, analysis, visualization

INTRODUCTION

In modern time, the ability to explore the oceans has been inextricably linked to the evolution and development of marine technology. Over the past 20 years, the steady
evolution of hardware design has resulted in increased sampling rates, data volumes and resolutions. Furthermore, improvements in the manufacturing of marine technology have decreased the overall cost of hardware which has expanded the user base of higher end technology. However, while these improvements have resulted in higher quality oceanographic data being collected over much larger spatial areas, the ability to analyze & quantify this data is limited to a combination of 2-D static geospatial software packages and/or command line driven signal processing software packages. Time series analysis of these complex data sets is essentially limited to a single point in time. Furthermore, even with recent improvements in 3-dimensional visualization, users are still limited to a single point in time within this higher end graphical space. The net results of these limitations often leave researchers to rely upon a "best guess" approach in determining the important events in their time series and visually bland graphics to deliver the message [1].

Over the past 3 years, The Myriax Group has taken upon themselves to find a solution to these limitations through the Eonfusion Project. The result has been a software based solution that provides data fusion of a wide variety of environmental data formats into a revolutionary 4-dimensional graphical space. Users now possess the ability to perform an intuitive, multi-dimensional time-series analysis without limitations based upon scale or source.

THE SOFTWARE SOLUTION

Eonfusion represents a unique software solution that combines cutting-edge processing with state-of-the-art visualization and analysis capabilities while providing connectivity to a wide range of data sources. Eonfusion supports effortless assimilation and integration of multi-variable data while providing interoperability with existing geospatial applications and Web-based data providers. The software offers a powerful 4D visualization interface together with a range of 2D-plotting tools for flexible graphical analysis of data. Its time stepping feature allows users to explore the temporal evolution of spatial structure and changes in resolved relationships between variables over time. It incorporates a visual dataflow interface (Fig 1) that simplifies application development, data manipulation and exploration. Eonfusion also offers a range of capabilities for quantitative analysis, including data filtering and spatial querying tools. Its revolutionary fusion operator allows both vector and raster data sampled on fundamentally different scales to be reconciled. The software is also readily customizable, possessing an integrated coding environment (the Expression Evaluator) for algorithm implementation and a programming interface (API) that facilities the development of modular extensions. Furthermore, it presents an optimum environment for users to integrate, fuse and visualize measured and modeled environmental data sets, simultaneously explore relationships between multiple variables, formulate and test hypotheses and to visually communicate results [1].

HABITAT MAPPING

Eonfusion's ability to fuse different data sets together and to visualize multiple data types in the same view makes it an excellent tool for visualizing and verifying habitat information. Depth values from a DEM (Digital Elevation Model) raster can easily be fused onto 2D habitat class polygons, yielding a 3D habitat class map. Eonfusion can also synchronize video data with a 4D (x, y, z, time) track. A probe tool is used to query the time attribute on a track and also to provide input to a video display. When the track is displayed in a scene along with bathymetry and habitat information the user can quickly and easily use their video display to ground-truth their data. They can also manually attribute sections of
the track according to what is observed in the video. The user's interpretation becomes a permanent part of the dataflow and can either be re-attributed within Eonfusion for further analysis or exported for incorporation into a secondary geospatial database.

**Fig. 5.** Multibeam Digital Terrain Model (DTM) color-coded based on surface roughness (Rugosity). The trackline of a video equipped ROV is fused with the raster data along with the video. “Tags” have been attached to the vector data which provides a visual quantification of the habitat which is then linked back to the database (data provided by NOAA NCCOS CCMA - Biogeography Branch), [http://ccma.nos.noaa.gov/about/biogeography/](http://ccma.nos.noaa.gov/about/biogeography/).

**CHANGE DETECTION**

Eonfusion is also an ideal tool for historical change detection analysis. For instance, multibeam bathymetric surveys were performed over the head of Monterey Canyon in the Spring of 2003, Fall of 2006 and Winter of 2006 to provide data in quantifying the sediment transport regime [5]. Morphological changes were detected by transferring the depth attribute from an earlier survey onto a more recent one. Sedimentation or erosion was then quantified by calculating the difference between each depth bin throughout the coincident space of the rasters and then color-coded accordingly. The time slider is used to provide an easy transition between survey and difference grids for rapid exploration of change. This same approach can be applied by utilizing modeled data that can either be loaded directly into Eonfusion or interfaced directly using the API.
Multi-survey/multi-year bioacoustical fish abundance information can seamlessly be combined with digital elevation data, habitat information and any other fisheries oceanographic cruise data for investigations on the spatial dynamics of fish populations. Furthermore, with the introduction of multibeam acoustic technology into fisheries community, its verification with split beam technology can be seamlessly fused to verify multisource measurements (Fig 7) [6]. Once the data has been visualized in a scene it's simply a matter of moving the time slider to see how parameters interrelate and vary over space and time. The scale-independent nature Eonfusion means that the original high-resolution raw data can be assimilated and viewed along with the spatial summary plots. This provides an intuitive and dynamic environment within which outliers and anomalies can quickly and easy be investigated within the source data. Furthermore, by assigning different attributes to the diameters of spheres/bubbles, the color as well as “halos” surrounding the spheres, the user can simultaneously quantify multiple parameters of a data set including target strength, target length and target width.
CONCLUSION

As technology continues to drive forward our ability to collect vast amounts of multidisciplinary data, so will the demand for generic solutions to some of the problems of large scale geospatial data integration, visualization, analysis and communication in support of environmental research and ecosystem based management. With an intuitive user interface for processing data, and exploring and manipulating 4D scenes, Eonfusion reduces the complexity of data integration and processing challenges that are faced by researchers. Regardless of application, this solution provides opportunities for discovery of multivariate relationships required for a better understanding of environmental systems, and improves communication of results and methods with colleagues and stakeholders.

ACKNOWLEDGEMENTS

Data from Buck Island Reef National Monument courtesy of provided by NOAA NCCOS CCMA - Biogeography Branch, http://ccma.nos.noaa.gov/about/biogeography/

Data from the Monterey Canyon Head courtesy of the Seafloor Mapping Lab at CSU Monterey Bay http://seafloor.csumb.edu/SFMLwebDATA_mb.htm#CANYON
Integrated multibeam and split-beam data courtesy of Tom Weber, University of New Hampshire.

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VISUALIZATION OF UNCERTAINTY IN BATHYMETRIC MEASUREMENTS

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Abstract: Assessing the quality of a survey has long been a need of the hydrographer. The overall error or Total Propagated Uncertainty (TPU) affecting each sounding results from the cumulative error contribution of all the elements of a survey operation from the physical mounting offsets to the positioning system and the sound speed profile. One of the tools of choice for bathymetric survey is the MultiBeam EchoSounder (MBES). The early attempt to quantify the uncertainty attached to each sounding measured by this sensor was based on a highly simplified model of the sonar; latter attempts refined this modelling by including specific details of the bottom detection algorithm. An inherent weakness of these approaches lies with the need for detailed knowledge of the inner workings of the MBES and of the environment. In this paper we present the implementation and validation of a new approach based on statistics collected directly on the raw sonar data recorded during normal multibeam survey operation. A test survey was conducted using the latest generation of multibeam sonar. The estimation of the uncertainty was computed using the most advanced of the previously published models adapted to the specifics of the sonar and using the new approach. The results of these methods are compared. Online visualization of the uncertainty greatly improves the ability to interpret the quality of the survey. Some options of visualization of the MBES uncertainty are mentioned.

Keywords: Multibeam echosounder, uncertainty, sonar, bathymetry, visualization
1. INTRODUCTION

While TPU estimation has preoccupied the scientific community of and bathymetric survey operators for quite a while [1], the real push for a complete analysis, understanding and measurement comes from the publication of a tool called CUBE [2]. This processing is based on an estimation model that depends on modelling of the error affecting the quantities being estimated or measured. Since CUBE is now widely accepted as the tool of choice to process bathymetry data, the availability of a reliable method to measure the uncertainty affecting all sensors used during a bathymetric survey has been exacerbated.

The first published model of the uncertainty affecting bathymetric soundings was based on work by Erik Hammerstad[1]. Rob Hare has published a series of reports and papers detailing the contribution of all aspects of the survey to the final TPU [1][3]. The early models of the sounder uncertainty were based on specific features of the MBES class for which it was designed. As such this model was not easy to adapt to other systems using different sonar techniques.

More recently, Xavier Lurton from IFREMER published a set of new equations based on a more advanced modelling of the sonar from beamforming process to bottom detection algorithm [4]. One major difference between earlier models and IFREMER approach is that the former are purely theoretical while the latter depends on actual survey conditions through a Signal-to-Noise Ratio (SNR). This model has been validated using theoretical analysis and extensive Monte Carlo simulation. While the agreement between the theory and the simulation is good, a major drawback of this model lies in its dependency on the system SNR.

A more robust approach has since then been proposed by IFREMER [5]. It consists of direct measurements of the uncertainty associated with the bottom detection algorithm. Since this approach does not rely on any model, it is universal in its application to all MBES. It is also easy to compute in real-time as a complement to the bottom detection process.

In the following of the paper, the different approaches are detailed and some experimental results are presented to compare the relative merit of each approach. Some visualization options for online presentation of this crucial information are proposed.

2. MODELS

Before we present two popular models of MBES uncertainty, it is important to state the assumption underlying TPE computation. The various sensors used during the survey are all contributing to some extent to the total uncertainty associated with each sounding. In order to make the overall analysis tractable it is assumed that the uncertainties are statistically independent. This hypothesis is quite reasonable as, for example, the uncertainty associated with a positioning system using GPS information is not affected by the uncertainty associated with measuring the speed of sound at the sonar location using a CTD probe.

This assumption allows us to concentrate on the uncertainty generated by the MBES. In so doing, we assume that the information required to operate the sonar properly are perfectly
measured. This applies in particular to sound speed at the location of the sonar and in the whole water column. So its effect on the sounding uncertainty is not discussed here. The combination of the contributions of all sensors (GPS, Vertical Reference Unit, Compass...) to the final overall uncertainty is performed using a forward error propagation principle (see [3]) usually implemented in the survey software or post-processing tool.

The first model of uncertainty associated with sounding measurement dedicated to MBES sonar identifies two components of the sounder uncertainty. The uncertainty in range measurement is [3]:

\[
\sigma_r = \sqrt{\left(\frac{\Delta r}{2\sqrt{2}}\right)^2 + \left[\frac{c \cdot \tau}{4\sqrt{2}}\right]^2} \tag{1}
\]

where \(\Delta r\) is the range resolution of the sonar, \(c\) is the speed of the sound at the sonar location, and \(\tau\) is the pulse length.

The error in beam angle measurement must be expressed differently for each bottom detection type. For amplitude detection it reads:

\[
\sigma_\phi(\text{amplitude}) = \frac{\Psi_y}{12} \tag{2}
\]

where \(\Psi_y\) is the beamwidth of the receiver and \(\sigma_\phi(\text{amplitude})\) represent the uncertainty when the bottom detection is performed on the signal amplitude.

When the bottom detection is applied to the split-array phase, the uncertainty is expressed as:

\[
\sigma_\phi(\text{phase}) = \frac{0.2\Psi_y}{\sqrt{np}} \tag{3}
\]

where \(np\) is the number of point processed during the zero-phase crossing detection.

The number of point can be computed as:

\[
np = \frac{r \Psi_y \tan(\theta - s)}{\Delta r} \tag{4}
\]

where \(\theta\) is the steering angle, \(s\) is the across-swath slope, and \(r\) is the range.
The MBES used to collect the experimental data presented in this paper uses a third approach to bottom detection. This approach combines the results from the amplitude and phase detection in a weighted average. It requires another estimation of the uncertainty when this blending occurs:

\[
\sigma_\theta(\text{blending}) = \sqrt{\alpha^2 \sigma_\theta^2(\text{amplitude}) + (1 - \alpha)^2 \sigma_\theta^2(\text{phase})}
\]  

where \( \alpha \) is a weighting factor.

A more advanced model was introduced by Xavier Lurton in [4]. Assuming a Rayleigh distribution, the error when applying split-array phase processing is defined as:

\[
\sigma_{\text{split}}^2 = \frac{\Delta \tau^2 \cos^2 \theta + \psi^2}{12} \left[ \frac{1}{(n_p - 1) \text{snr}} + \frac{1}{2(n_p - 1)(n_p - 2) \text{snr}^2} \right]
\]  

where \( \text{snr} \) is the output signal to noise ratio, \( L \) is the length of the receiver antenna, \( \mu \) is the proportionality factor, \( \delta \) is the steering angle including any mounting offset, and \( \psi \) is the across track distance. In our case, \( \mu \) is equal to 0.5.

When the bottom detection is performed using the amplitude signal the uncertainty is expressed as:

\[
\sigma_{\text{amp}}^2 = \frac{0.0262}{n_a} \left[ (\nabla \Psi)^2 + \left( \frac{c \tau}{2 \cos \theta} \right)^2 \right]
\]  

where \( n_a \) is the number of samples used for the centre of gravity estimation of the bottom location.

During our analysis of the model we refined the model of equation (6) to include a factor related to the relationship between the slope of the split-array phase and the off-axis receive direction [6]. We compared the validity of the three models with some simple statistical estimation of the uncertainty defined as the standard deviation of the depth measurement.
This first experiment was conducted on a flat sea floor while the vessel was idle. Unfortunately, the sound speed profile was not collected during the experimentation so refraction artefacts are affecting the experimental data. Fig. 2 shows that while the models are within an order of magnitude of the uncertainty when the phase detection is assumed, all models for amplitude detection grossly overstate the uncertainty.

3. DIRECT MEASUREMENT OF THE UNCERTAINTY

Recently, a new approach was proposed to measure the uncertainty directly on the sonar signal. While the models presented above rely on some level of knowledge of the sonar characteristics and environment conditions, the new approach does not. It is a direct estimation of uncertainty as a statistical measure applied to the complex signals.

When the amplitude signal is processed using a centre of gravity approach, the uncertainty is measured as [5]:

$$\frac{\delta \tilde{e}}{z} = \frac{\delta \tilde{e}_D}{t_D} = \frac{\sqrt{2\Delta t_D}}{Bt_D\sqrt{f_s}}$$

where $t_D$ is the range to the bottom, $z$ is the depth, $\Delta t_D$ is the envelop width for that range, $f_s$ is the sampling frequency of the signal, and $B$ equals 5 for the centre of gravity method.

The only knowledge required to estimate this uncertainty is the number of samples processed. There are at least two choices to compute the envelop width, one correspond to the
actual receiver beamwidth translated in a bottom footprint and the other corresponds to the beam spacing. In this paper we use the former option.

The second measure applies to the soundings resulting from phase processing. In this case the relative uncertainty is expressed as:

$$\frac{\delta x}{z} = \frac{\delta t_D}{t_D} = \frac{\delta \Delta \phi}{A t_D N}$$

(9)

where $A$ is the slope of the phase ramp, and $\frac{\delta \Delta \phi}{\sqrt{N}}$ is the phase standard deviation.

A comparison on real data of IFREMER model and their latest approach to uncertainty estimation is presented in Fig. 3. The uncertainty figures are average over 50 pings.

4. VISUALIZATION OF THE UNCERTAINTY

The direct estimation of the sounding uncertainty is performed online. It is thus available for display during the survey. The simplest display of the uncertainty has been presented in the previous paragraph. It is a simple graph presenting the relative uncertainty as a function of the beam incidence angle.

During the survey operation the evolution of the uncertainty is better appreciated as a function of time. This provides a means to assess the effects of bottom topology and sonar settings on the uncertainty. An example of this option is presented in Fig. 3 left.
It can be seen around ping 200 that the uncertainty increases across all beams. This is caused by the presence of a pipeline in the path of the survey line (see Fig. 3 right)).

This representation can be further improved by referencing the sounding on a geographic grid. The UTM coordinates of the soundings are computed for each ping and the colour-coding of the uncertainty scales the usual range of this measure. It becomes a valuable tool to plan the next line of the survey. It can be seen in Fig. 4 that while most of the current line is valid, there is a patch of the outer swath that presents a much higher level of uncertainty and could require an additional survey line if the survey standard were set extremely high.
5. CONCLUSIONS

The new approach proposed by IFREMER to measure the uncertainty associated with the bottom detection algorithm directly on the raw sonar data provides a more reliable solution to the crucial task of quantifying this information. This method does not require detailed knowledge of the sonar operation or of the environment conditions. As such it is inherently more robust than the previous models. The implementation of the algorithm is straightforward and it does not consume much resources. It can operate while conducting the survey and thus become an additional quality control tool for the surveyor. This information is directly related to hydrographic survey standards and it is possible to apply a simple threshold to raise an alarm if the survey operation does not meet requirements. However, a finer analysis is possible if the data is presented with more details. We have proposed a series of visualization option that can be easily implemented in real-time. A more advanced display adding a time dimension to the 3D data could be used in post-processing.

6. ACKNOWLEDGEMENTS

The author wishes to acknowledge Justin Friesner and Matthew Hayes for collecting the experimental data used in this study.

REFERENCES

Structured Session 22

Ambient noise and its applications to monitor physical, biological and anthropogenic processes and activities in the ocean

Organizers: Jeffrey Nystuen, Chi-fang Chen & Hsiang-Chih Chan
Objective classification of physical, biological and anthropogenic underwater sound sources in the marine environment

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Abstract: Passive measurements of underwater sound can be used to quantitatively measure the physical processes that produce or modify the detected sound. This includes wind, rain and ambient bubbles. However, in order to reliably quantify these processes, objective acoustic classification of the sound source is required. A low duty cycle recorder (PAL) is used to provide long-term monitoring of up to one year. The sampling strategy for the PAL allows for the detection and identification of sound-producing marine animals, especially whales, and human-generated noises, e.g., ships. This allows different sampling strategies to be implemented when different sound sources dominate. Thus, a low-duty cycle instrument can provide quantitative assessment of the physical marine environment (wind speed, rainfall rate and type, and sea state) as well as detect and quantify biological and anthropogenic activities.
Operational ocean monitoring and forecasting in the Eastern Mediterranean


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Abstract: The POSEIDON integrated ocean monitoring and forecasting system for the Aegean Sea was developed in 1997-2000 and has been operational since then, delivering information products and services for the state of the marine environment (www.poseidon.hcmr.gr). A major upgrade was recently carried out during aiming to: a) extend the buoy monitoring network to the Ionian and Levantine Seas b) upgrade the observing capacity with new in-situ physical and biochemical data as well with relevant satellite observations and c) enhance the forecasting skill and extend it to the whole Mediterranean Sea. Apart of the standard buoy network that provides meteorological and surface oceanographic data, the monitoring component now includes two multi-parametric open sea observatories: a) the Cretan Sea E1-M3A station focused on biochemical observations of the euphotic zone (0-100m) and physical observations of the upper 1000m and b) the SE Ionian Sea Pylos station designed for multi-disciplinary observations of the whole water column with emphasis on near bottom measurements. This is achieved through an autonomous bottom platform (deployed at 1670m) that hosts a variety of sensors and transmits data to the surface buoy using acoustic modems. Apart of the standard CTD sensors, the mooring line also hosts a Passive Aquatic Listener (PAL) that provides for the first time in operational oceanography online observations of ambient noise in the Mediterranean Sea. The PAL system identifies and classifies the sound sources into geophysical (i.e., rain accumulation, wind speed), biological (i.e., marine mammals, crabs, shrimps, etc.) and anthropogenetic (i.e., ships, fishing boats, sonar) and then quantifies them. The forecasting component includes high resolution weather, wave, hydrodynamic and ecosystem modeling for the whole Mediterranean and the Aegean Seas based on the a) the POM model for hydrodynamic forecasts in the Aegean (1/30o resolution) and Mediterranean Seas (1/10o) also using data assimilation of satellite and in situ observations b) the WAM-Cycle4 model for wave forecasts at the same resolution/configurations c) the ERSEM model for ecosystem forecasts at a single Mediterranean-scale configuration (1/10 o) and d) the
non-hydrostatic ETA model for meteorological forecasts in the Mediterranean and Black Sea areas at a 1/20° resolution. POSEIDON is the national contribution of Greece to the ocean components of GMES and GEOSS and is coordinated with relevant EU funded projects as well as the regional developments of GOOS (MedGOOS & MOON).
AN INTEGRATED OCEAN OBSERVING SYSTEM IN IONIAN SEA

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Abstract: During the last decade, operational oceanography had to face the challenge of building integrated observational systems capable of real-time recording and transmitting parameters not only from the atmosphere and the upper sea, but also from the deep sea layers. The Hellenic Centre for Marine Research in the frame work of the Poseidon II project, managed to deploy a Seabed observatory in the Southern part of Ionian Sea, a few miles outside Pylos bay at 1674m depth. The real-time measurements are transmitted to operational centre through the existing «Poseidon» telecommunication network. The communication between the platform and the surface buoy is achieved through hydro-acoustic modems. An inductive modem is also used for the functioning of a Passive Aquatic Listener (PAL) which is attached on the buoy’s mooring line. The PAL has an embedded algorithm which allows the noise classification in order to provide timeseries of precipitation, waves, whale’s signals etc. The tsunami detection module consists of a high resolution pressure sensor which records changes in water column pressure in high frequency time steps. The bottom observatory is also equipped with a CTD instrument for temperature and salinity records of the deep water environment. Furthermore, additional sensors can also be hosted to the seabed platform covering possible future needs.

Keywords: tsunami alert systems, multi-sensor observatories, sea level
1. Introduction

Since 1999 a real-time monitoring, forecasting and information system for the Hellenic Seas called “POSEIDON” has been implemented, contributing to the efforts of GOOS and its Mediterranean component MedGOOS. This integrated Operational Oceanography system is typically composed by three components: a) the data collection system operating in real time, b) the data analysis and forecasts production system, and c) the products dissemination to end-users system. A network of monitoring buoys that collect atmospheric and upper ocean oceanographic data at different sites of the Aegean and Ionian Seas is the backbone of the “POSEIDON” system. As shown at the next figure (fig.1), two different types of monitoring platforms are used depending on the depth and the data needed from its mooring site. Data are transmitted in real time to an operational centre at HCMR where they are analyzed and used for validation of the operational forecasts (Nittis et al. 2008). Both observed data and model results are used for generation of synthetic visual products that are disseminated to end users through internet or other means (e.g. mobile telephony).

Pylos mooring site was integrated into the “POSEIDON” buoy network since 2007 covering the South Ionian Sea region. Together with the E1M3A station, Pylos station is equipped with different kinds of sensors capable of recording a variety of meteorological, physical and biochemical parameters. These two stations have also been a part of EuroSITES network of open sea multi-sensor moored arrays which operates at European level and contributes to OcenaSITES global network of open sea Eulerian observatories (Petihakis et al. 2008). During November of 2008 the first oceanographic deep sea bottom platform in the Mediterranean Sea was deployed at a depth of 1674 meters and started to transmit real time data from the sea bottom through an acoustic modem to the acoustic receiver of the surface Pylos buoy. The collected data are then transmitted through the “POSEIDON” telecommunication system to operational centre of HCMR.

The SDSM (SeaWatch Deep Sea Module) is a platform that lay on the sea bottom and it has built-in sensors capable of recording at a high resolution the water column pressure, temperature and salinity. The module was set up and provided by Fugro OCEANOR Co. The coupling of a sea bottom observing platform with a multi-parametric mooring, creates new opportunities of monitoring the ocean not only through air-sea interaction related parameters or the first few hundred meters of the water column data, but also through geo-physical and bio-chemical data of the deep sea basin that are now becoming available. This work will present the architecture and a short technical description of the system as such selected timeseries, raw data, emerging problems and issues need to be addressed in the real future.
Fig. 1: The POSEIDON buoy network. White dots represent the SeaWatch type buoys which are multi-parameter buoys that can be used to collect directional wave data as well as meteorological, oceanographic and water quality parameters. The other three are Wavescan type which are multi-parameter measuring platforms well suited for deep offshore locations or areas of strong current forces (Furgo OCEANOR www.oceanor.no). The blue dot indicates the position of Pylos mooring and the Sea-bed platform.

2. System architecture and instruments

2.1 Architecture and data transmission

The system consists of an autonomous platform that lay on the sea bottom and a surface Buoy which is part of the Poseidon monitoring network. In normal mode the platform is controlled by the acquisition unit of the Buoy and like all the other sensors data are acquired every three hours. The communication link between Buoy and platform is implemented by hydro acoustic modems at 1200 baud rate. In that way the acquired data are packed and transmitted to the central receiving station using the communication systems of the surface Buoy. In case of tsunami detection it is the platform that initiates transmission. In this mode all the pressure data (1 sample per 15 seconds) are continuously transmitted to the surface Buoy and immediately to the central receiving station.

The SDSM seabed observatory lies on the sea bottom at 1674m depth in a range of 1km approximately from the buoy’s mooring line. It is especially designed for tsunami surveillance. The SDSM comprises of a high-resolution pressure sensor interfaced, via a processor, to a combined acoustic modem / release. The processor reads pressure data and transmits it, together with temperature and battery voltage, acoustically to the surface WAVESCAN buoy. The pressure samples are compared to the calculated values using the DART algorithm (Gonzalez et al. 1998). In case the difference exceeds the user defined threshold the unit’s state changes to tsunami mode. Onboard alkaline batteries supply its power. The equipment is mounted on a frame, anchored by concrete, with floatation spheres that enable it to be retrieved by issuing a release command to the acoustic modem / release and waiting for it to surface (Furgo OCEANOR www.oceanor.no).
The buoy used at Pylos site is a Seawatch-Wavescan type which is a multi-parametric instrumentation platform and suitable for deployment in deep offshore locations. Using an inductive mooring cable, CTD instruments are attached on the mooring line providing salinity, temperature and pressure data down to 1000m depth. Biochemical parameters such as oxygen and chlorophyll can also be measured from the sea surface down to 100m depth. ADCP profilers collect current data every 5m from the sea surface to the depth of 50m. At the end of the inductive cable an acoustic modem which receives data from the sea-bed platform is attached. The upper part of the buoy has a mast with the meteorological instrumentation adjusted on it such as wind, temperature and air pressure sensors. Except from the basic meteorological parameters, additional parameters can be measured such as rainfall, radioactivity, radiance and irradiance. On the top of the mast there are also a GPS system and the telecommunication antenna. Two systems are used for transmitting the collected data. GSM cellular networking and INMARSAT satellite networking both transmitting data to a special receiving station at HCMR premises. The transmission is near real time with a three hours cycle.

![Diagram of the observing system in South Ionian Sea](image)

**Fig.2, 3:** Schematic presentation of integrated observing system in South Ionian Sea. Left figure shows a 3-D representation of the acoustic telecommunication between SDSM and the buoy as such the satellite data transmission. The right figure shows a more detailed representation of the instrumentation and the anchoring of the system.

### 2.2 Instruments and sensors

The pressure sensor of the seabed platform is a transducer of type 43K-101 from Paroscientific. The sensor uses a precision quartz crystal resonator whose frequency of oscillation varies with pressure induced stress. The pressure range is 0-3000 psia. Pressure signal is a frequency output with a 10% frequency change within the frequency band 30 KHz to 42 KHz. Temperature signal is a frequency output with a 45 ppm/°C sensitivity within the band 168 KHz to 172 KHz. A seabird 16plus equipped with conductivity temperature and depth sensors is connected to the SDSM unit. The hydroacoustic modem that combines also releaser functions is the Smart release from Benthos. The following table presents the Pylos buoy instrumentation for the 11 of November 2008 deployment.
Table 1: Sensors used in Pyllos Wavescan buoy during November 2008 deployment. The left columns present the measured parameters and the corresponding depths.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Depths measured (m)</th>
<th>Sensor(s) used</th>
<th>Accuracy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wind speed/dir.,</td>
<td>Surface</td>
<td>Young 04106</td>
<td>1m/sec, 10 deg</td>
</tr>
<tr>
<td>Air Pressure,</td>
<td>Surface</td>
<td>Vaisala PTB 220A</td>
<td>-/+0.15hPa</td>
</tr>
<tr>
<td>Air temperature,</td>
<td>Surface</td>
<td>Omega</td>
<td>-/+0.1oC</td>
</tr>
<tr>
<td>Wave Height, direction, period</td>
<td>Surface</td>
<td>Fugro OCEANOR Wavesense</td>
<td>0.1m, 0.5 deg, 0.5 sec</td>
</tr>
<tr>
<td>SST, SSS surface,</td>
<td>Surface (1m)</td>
<td>Seabird sip</td>
<td>-/+0.1 oC, 0.05 mS/cm</td>
</tr>
<tr>
<td>Currents</td>
<td>Surface (1m)</td>
<td>Nortek Aquadopp current meter</td>
<td>-/+0.5 cm/sec</td>
</tr>
<tr>
<td>Temperature</td>
<td>20, 50, 75, 100, 250, 400, 500m</td>
<td>Seabird 37-IM C-T</td>
<td>0.005 oC</td>
</tr>
<tr>
<td>Salinity</td>
<td>20, 50, 75, 100, 250, 400, 500m</td>
<td>Seabird 37-IM C-T</td>
<td>0.0005 S/m</td>
</tr>
<tr>
<td>Pressure</td>
<td>250m</td>
<td>Seabird 37-IM C-T-D</td>
<td>0.1% FS</td>
</tr>
<tr>
<td>Precipitation, wind speed</td>
<td>500m</td>
<td>PAL (passive aquatic listener)</td>
<td></td>
</tr>
</tbody>
</table>

3. First deployment experience, preliminary results

3.1. System deployment

The selection of the appropriate position for the deployment of the platform was made after taking into consideration the seismicity of the territory and the plateau topography of the sea basin. The East part of the Ionian Sea is a territory of high seismic activity where in the past tsunami incidents had been reported. Furthermore the depth in this region reaches the 3000m and is presenting a rough terrain with an increased possibility of underwater shifts. This requires, though, a detailed recording of the sea bottom topography since the platform has to be located in a safe and stable position with a clear sight of the buoy’s mooring line without steep slopes nearby. For this reason multi-beam recordings had been made above this sea region with R/V Aegeo the day before the deployment. Despite the topography survey, the platform’s position could not be precisely defined during the deployment due to weather conditions and ship’s drift. This fact can produce miss-functioning problems especially with the acoustic telecommunication part. Figure 4 presents the combined picture derived from the multi-beam recordings. The triangle and the square where supposed to be the deployment positions of the platform and the buoy respectively. The cross indicates the final SDSM position after the deployment, while the buoy has a drift from its initial position of 1000-2000m.
3.2 Presentation of selected diagrams

The 3-hourly data are transmitted to “POSEIDON” operational centre in real-time and the produced timeseries are used for monitoring, forecasting but also for data assimilation and climatology studies in delayed mode. The next figures present a set of selected timeseries covering the time period of 40 days after the deployment at Pylos. In figure 5 the air pressure variation is presented and the seasonal lows of the South Ionian are visible while next plot (fig.6) shows the significant wave height. As expected the correlation between low air pressure systems and increased waves is strong. In figure 7 the water temperature for several depths is presented. These timeseries describe the seasonal change of the upper thermocline which is at its lowest depths at early autumn period due to seasonal heating and then reaches the deeper layers due to gradual mixing at mid-winter period. In this case the change of the stratification starts at the middle of December as shown from the surface and the 100m depth temperature. The last figure (fig.8) presents the high frequency samples (every 15 sec) of the pressure sensor of SDSM for the first 10 days after the deployment. In this figure, sea level has a normal variability in which the high frequency alterations such as waves and the low frequency movements such as the semi-diurnal tidal waves are traceable but only for the first 6 days. The measured depth after November 19 shows abnormal variation with uneven changes and spikes. This is attributed to a sensor’s missfunction and is currently under examination after the retrieval of SDSM.
Fig. 5: The air pressure 3-hourly data measured at Pylos surface buoy for the time period of 11/11 to 31/12 2008.

Fig. 6: The significant wave height 3-hourly data measured at Pylos surface buoy for the time period of 11/11 to 31/12 2008.

Fig. 7: 3-hourly data of water temperature at sea surface, 100m, 250m and 500m depth measured at Pylos surface buoy for the time period of 11/11 to 31/12 2008.
Fig. 8: The raw-data of the depth derived from the SDSM pressure sensor after the recovering of the module. With a sampling rate of 15sec these timeseries present the high frequency variation of the sea surface.

4. Conclusions

Integrated monitoring systems such as the one described before create new potentials in operational oceanography. Data from the deep sea basins can be available in real time; meaning that an overall picture of both air-sea and sea water-sea bottom interactions is feasible. These efforts introduce several constraints that need to overcome such as the deployment and positioning of the units and the interpretation of the produced timeseries. The major challenge for the next few years will be the integration of additional sensors (bio-chemical, geological) into the system and the deployment of additional systems in the Mediterranean Sea. This will be a main step towards a permanent, multi-parametric, open-ocean observing network that will support operational forecasting in the Mediterranean Sea (Nittis et al. 2003).

5. Acknowledgements

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6. References


Abstract: One of the most challenge and difficult atmospheric parameters to measure is rain, especially over the oceans, because of its spatial and temporal variability. Since, the last decades, many satellites have been launched to observe globally the rain variability; however, they have large scale spatial and temporal resolution. Also, there are few surface validation sites over the oceans for calibrating/validating their retrievals. Even though, ocean plays a vital role in the hydrological cycle, till now air-sea feedback mechanisms are not properly resolved by numerical models. Therefore, it is significant to have geophysical (precipitation, wind speed, wind direction, etc.) observations over the oceans that can be used to initialize forecasting models, to improve their numerical parameterisation schemes, and validate satellite retrieval algorithms.

Operational monitoring systems provide in real time this required information. However, rain and wind speed observations using raingauges and anemometers installed on buoys are point measurements and may introduce erroneous values in the measurements. A Passive Aquatic Listener (PAL) integrated for first time with an operational oceanographic monitoring (Poseidon) system, in the southwest Ionian Sea buoy. Acoustical measurements of rainfall and wind speed are reported from 500 meters depth, since November 2008. These spatial averaged measurements are compared with the buoy’s surface anemometer and the Hellenic National Meteorological Service C-band radar rainfall measurements.

Keywords: Underwater acoustics, rainfall estimation, wind speed measurements, operational oceanography.
1. INTRODUCTION

Ambient sound in the ocean is a combination of natural and anthropogenic sounds. Various physical processes including wind, rain, and drizzle are primary sound sources in the frequency range from a few hundred Hertz to 50 kHz. Human generated sounds include ships and sonars, and different marine animals (whales, dolphins, etc.), especially cetaceans, produce sound underwater in this same frequency band. As part of the POSEIDON monitoring and forecasting system, the Hellenic Center for Marine Research (HCMR) has acquired an advanced Passive Aquatic Listener (PAL) sensor that can be used to identify and quantify those sources of sound from deep water acoustic measurements. Understanding the distribution and change of oceanic rainfall patterns is a major component of global/regional water cycle and climate change. In addition to rainfall rate, we need detailed knowledge of precipitation type (in the forms of convective vs. stratiform precipitation) and drop size distribution (DSD). Satellite-based measurements of rainfall provide global coverage of rainfall distribution [1] however, these measurements need to be verified by surface rainfall measurements. The lack of detailed knowledge of DSD is a primary factor that limits the accuracy of radar and satellite rain retrievals [2], [3]. In addition, an understanding and characterization of precipitation microphysics is needed to improve parameterizations in numerical weather prediction (NWP) models (e.g., [4]). Acoustic measurements provide basic information on rainfall science that cannot be provided by satellite observations, including high temporal resolution, rainfall classification, and DSD. Consequently, these measurements if proven robust and accurate can be used to (1) provide continuous measurements of precipitation characteristics in the open sea (e.g., Mediterranean region) and (2) as in situ reference to physically validate satellite observations.

One interesting feature of the acoustical measurement is that the listening area for a hydrophone, its effective “catchment basin,” [5] is proportional to its depth, and yet, the signal should be independent of depth if the sound source is uniformly distributed on the sea surface. Thus, the acoustical measurement of rainfall has an inherent spatial averaging that can be compared to the beam filling assumption of radar or satellite measurements of rainfall. Rainfall makes only one of the acoustic signals that are part of the marine ambient sound budget. Other physical processes that can be measured are wind and sea state (bubbles, salinity, and temperature) conditions. Passive acoustic monitoring for marine mammals, especially whales, is critical to ecological studies of these animals. In section 2, we give a brief technical description of the PAL. In section 3, we describe the study area and data associated with the POPSEIDON monitoring system. In section 4, we present the acoustic data and our future work.

2. PASSIVE AQUATIC LISTENER (PAL)

PALs are autonomous acoustic recorders designed to be attached to ocean moorings. They consist of a broadband, low noise omnidirectional (zenith angle) hydrophone (Hi-Tech-92WB), a signal processing board, a low-power microprocessor (Tattletale-8) with a 100 kHz A/D digitizer, a 2 GB memory card and a 60 Amp-hour battery pack. The sampling strategy can be designed to allow autonomous operations for up to one year. Physically a PAL is a cylindrical instrument 76 cm long by 15 cm in diameter. The hydrophone extends from one end. The nominal sensitivity of these instruments is -160 dB relative to 1 V/µPa and the equivalent oceanic background noise level of the pre-amplifier system is about 28 dB relative.
to 1 $\mu$Pa$^2$Hz$^{-1}$. Bandpass 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). The hydrophone sensitivity also rolls off above its resonance frequency, about 40 kHz. A further sensitivity correction due to the depth of deployment is also present. A data collection sequence consists of a four-second time series collected at 100 kHz.

3. PAL DEPLOYED UNDER THE POSEIDON II FRAMEWORK

The POSEIDON (http://www.poseidon.ncmr.gr) is a monitoring, forecasting and information system for the Hellenic Seas and Coastal areas. It is operated by the Hellenic Center for Marine Research (HCMR) providing real-time information and forecasts for weather, sea-state, hydrological structure and water quality. The POSEIDON system consists of a network of 11 oceanographic buoys equipped with state of the art physical, chemical and biological sensors deployed in the Aegean and Ionian Seas, and an operational forecasting system. The extension of the POSEIDON system (named POSEIDON II) includes, among other aspects, the deployment of PAL sensors on two of the deep sea POSEIDON buoys (shown in Fig. 1). In POSEIDON II the PALs are connected through inductive modem to the surface buoy providing real-time data to the HCMR operational control center.

3.1. Experimental sites

During the 2 – 11 November, 2008 two Passive Aquatic Listeners (PAL I and II) deployed to two different Poseidon Buoys. The first PAL deployed to the North East Aegean Sea at Athos Buoy site, while the second one deployed to South West Ionian Sea at Pylos Buoy site (shown in Fig. 1). Both sites are fully covered with the Hellenic National Meteorological Service (HNMS) weather radars (both rings have 150 km range) that could be used for off line future rainfall validation with the PAL’s rain estimation. The PAL I (shown in Fig. 2) sensor is attached to the mooring line as shown in Fig. 3b at Athos buoy at about 200 meters depth. The PAL I is currently offline with the future potential for online operation. The buoy is equipped with an anemometer that can be used for surface validation of the PAL’s wind speed measurements. The second buoy equipped with a PAL system is at Pylos deployment site. Figure 3 shows the Pylos’ deployment equipped with the PAL II system connected to an inductive modem for the online configuration. The PAL II deployed at 500m depth. The period of the acoustic data used are from November 2008 until to April 2009 and from the Pylos’ site.

3.2. Acoustic data

A data collection sequence takes about 20 sec and consists of eight 10.24-ms time series, each separated by 5 sec. Each of these time series is fast Fourier transformed (FFT) to obtain a 512-point (0–50-kHz) power spectrum. Geophysical generated sounds from rain, drizzle, or wind are generally stationary over a 20-s time interval, whereas banging from ships or moorings or chirps, whistles, or clicks from biological sources are sound signals that are usually non-stationary over that time interval. Thus, a preliminary evaluation of the sound source is to remove non-stationary data samples. The eight spectra are then averaged into a single spectrum and evaluated to determine the acoustic source (rain, wind, or drizzle).
Fig. 1: The POSEIDON buoy network and two Hellenic National Meteorological Service weather radars providing coverage of the two deep sea buoys with PALs. The two red rings represent the 100-km range of the two radars from the two buoys.

A sensitivity adjustment of instrument, depth, and local ocean conditions (overall higher to lower sound levels) is determined by choosing a sound condition where the signal-to-noise ratio is high and assuming a uniform sound source at the surface. At low wind speeds, the recorded signal includes a component from the ambient background and from instrument noise. At high wind speeds, there is a change to the spectral shape of the wind signal due to attenuation of the signal from ambient bubbles in the water. However, at moderate wind speeds (4–8 m/s), the sound signal is well above the background noise and has a uniform spectral slope between 1 and 40 kHz. This signal is adjusted for absorption and depth. Ocean currents will bend the mooring, causing a horizontal displacement of the acoustic sensors. However, during most of the experiment, the mooring line can be assumed to be vertical.

For the validation of whale detections a storing of the entire four second time series is saved if whale detection is occurred. This allows an audio confirmation that the sound source generating a “trigger” is a whale. A typical i.e. whale or dolphin vocalization lasts less than 4 seconds. Consequently, if the eight sub-samples report the same spectra, then the sound source present is assumed to be quasi-stationary: wind, rain, drizzle, continuous ship noise, etc., and not a whale. Alternatively, if one or more of the spectra are different from one another, then a “transient” sound is assumed to be detected. This sound might be whale detection. However, transient sounds associated with shipping, or other biological sources may also meet the transient sound detection criteria. If the detected spectrum is consistent with a whale signal, the PAL records the entire 4.5 second time series for later analysis.
An example of a 4-second time series showing dolphin detection from Pylo’s site is shown in Figure 4. Note the computer memory capacity (2 GB) limits the total number of time series saves to 2200 per deployment. Consequently a rationing code controls when 4-second time series saves are allowed.

The PAL is capable of communicating to the shore via its serial port (shown in Fig. 3b) to an inductive modem. The system via the inductive cable sends every three hours a text phrase of the averaged three hours accumulated rainfall (in mm/hr), mean wind speed (in m/s), number of whales (integer), the percentage (%) of ambient noise and precipitation, battery voltage, the remaining storage space and the last instantaneous measurement of spectrum.
3.3. Surface buoy anemometer data

On both sites, the buoy is equipped with a surface anemometer that reports wind speed and direction every three hours. The wind speed measurements are 10 minutes averaged starting at the last 15 minutes of every three hours window of each day.

3.4. Radar data

As shown in Figure 1 the two POSEIDON buoys that carry PALs are in range (~100 km) of two operational weather radars from the Hellenic Meteorological Service (HMS). The Xrisoupoli radar (40°55’32”N, 24°37’41”E) is a newly deployed high-power C-band (Klystron transmitter) dual-polarization and Doppler radar. It has high sensitivity and narrow (0.85 deg) beam width. It provides measurements of reflectivity at H and V polarization, differential reflectivity, differential phase shift, H/V copolar correlation coefficient, and Doppler information. On the basis of those measurements we can device dual-polarization radar techniques to retrieve rainfall and drop size distribution at high accuracy and resolution [6], [7], [8]. Correction due to rain-path attenuation can also be performed with high accuracy using the differential phase shift information [9]. The second radar (in Adravida; 37°55’21”N, 21°17’13”E) is a newly upgraded high-power S-band (Magnetron) single-polarization Doppler radar. The radar has high sensitivity, but coarse beam width (1.3 deg). Although it is not susceptible to rain-path attenuation due to the long wavelength (10 cm) it provides only H-polarization reflectivity and Doppler information. Lack of polarimetric information does not allow the retrieval of DSD parameters by this radar. However, we can
device algorithms to retrieve precipitation type and rainfall rate [10], [11] on the basis of single-polarization reflectivity measurements. At 100-km range the beam sizes for the Xrisoupoli and Adravida radars is 1.5-km and 2.5-km, respectively. A point to note is that the theoretical contributing area for hydrophone located at 500 m depth is a circular area of 1.5-km radius and the one at 200 meters depth is a circular area of 600 meters [5], which is comparable to the two radar resolutions at 100-km range.

4. PAL DATA ANALYSIS

Ambient sound in the ocean is a combination of natural and man-made sounds. Various physical processes, including wind, rain, and drizzle, are the primary sound sources in the frequency range from a few hundred hertz to 50 kHz. These are sound sources at the sea surface. The microphysics of the sound generation is resonating bubbles created during the splashing of wind waves or raindrop splashes. These bubbles are very near the free surface of the ocean and, consequently, are assumed to behave as vertically oriented acoustic dipole sources. The next step in the acoustic data analysis is to identify the sound source. Different sources include breaking waves from wind, raindrop splashes, ships and biological sources including whales. Each of these sound sources has unique spectral characteristics that allow detection and identification of the sound source. Critical components are the relative sound levels between frequencies and slopes of the spectra.

4.1. Identify the sounds

Different sound sources are identified by their spectral characteristics. Features of sound source spectra that can be used to identify the source include spectral levels at various frequencies, ratios of these levels, spectral slopes and the temporal persistence of the sound source. The data were examined to find times when the sound source could be confidently assumed. Long periods (hours) of steady uniform sound were assumed to be periods of constant wind. Short loud events consistent with typical ship spectra (very loud at low frequency) during non-rainy periods were assumed to be ships. Distinctive rain and drizzle spectra were identified and confirmed with radar. These ‘typical’ sound sources are shown in Figure 5, taken as an example from Anagnostou et al. [5], and were used to build an acoustic classification algorithm that can be used to objectively identify the sound source in the remaining data. The goal is to reliably detect the sound source so that subsequent analysis is not contaminated by sound generated by other source. Figure 6 shows the relationship between 8 and 20 kHz. This comparison of sound levels at two frequencies is particularly illustrative for demonstrating the ability to use ambient sound to identify the sound sources. In fact, multiple measures are actually used. For example, there is an ambiguity for the sound source “wind = 12 m/s” and “ships”. Other features of the sound field are needed to separate these two sources. For this situation, periods of high wind are usually of long durations (hours), while ships pass the mooring in minutes.

The points associated with the "clean" wind speed spectra used to establish the sensitivity correction reveal a distinctive pattern that can be used to quantitatively measure wind speed [12], [13]. The principal features of a typical ship sound spectrum are high sound levels at lower frequency (below 2 kHz), and a consequently very steep spectral slope between 8 – 15 kHz. The next component of the classification routine is to identify precipitation [14].

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Rain has relatively more high frequency content, producing relatively high sound levels at 15-30 kHz and a flatter spectral slope from 8 – 15 kHz. In fact, drizzle has a very distinctive spectral peak from 13-25 kHz associated with bubbles trapped underwater by small (1 mm) raindrops [15].
Heavy rain, containing large raindrops (over 2.2 mm diameter) produces sound at lower frequencies, but still generates a sound spectrum that is easily distinguished from the wind-generated spectrum. Once the sound sources are identified, quantitative analysis is possible. The acoustic wind speed and rainfall rate estimations are then compared with the surface anemometer wind speed and the rainfall rate reported from the NHS weather radars. Anagnostou et al. [5] has report that comparison of high-resolution coastal weather radar shows an increase in effective listening area of the PAL with increasing depth and high correlations.

5. CONCLUSION/FUTURE WORK

Our future goals with this long term deployment are to evaluate, determine uncertainties of, and improve the acoustic rainfall rate and wind speed estimates. The main strategy in evaluating the acoustic rain estimates using the POSEIDON data sets will be by comparing them to the NHS' radar rain estimates, which in turn will be evaluated by the rain gauge measurements. We will try to identify and resolve significant discrepancies between the underwater and the radar rain rate estimates. Acoustic classification of rainfall type is another goal of this project. An acoustic classification algorithm of rain type will be developed with the ongoing POSEIDON data. The influence of wind on the performance of acoustic classification of rainfall type will also identify. In addition, Anagnostou et al. [5] have shown a bias with respect to a published rainfall rate algorithm using data from the tropical Pacific Ocean that has been also used in the two PALs deployed in the POSEIDON system.

Finally, acoustic inversion for rainfall drop size distribution (DSD) is the most ambitious acoustic measurement of rainfall at sea. The published inversion technique [15] used data from a shallow, sheltered pond. The influence of wind on the signal from rain will be estimated and removed from the rainfall signal. The drop size distribution inversion algorithm will then be applied and integrated rainfall DSD parameters will be calculated and compared to co-located radar estimates of the same DSD parameters. Reference data for the assessment of underwater estimates of DSD will be retrievals from Dual-Polarization radar techniques [7], [8]. In this study we will use polarimetric radar data from the Xrisoupoli C-band Dual-Polarization radar. Radar estimates of DSD will be first verified against DSD parameters derived from disdrometer-measured raindrop spectra.

REFERENCES


Using moored passive acoustic recorders to assess seasonal occurrence and movements of southern resident killer whales and other cetaceans in the coastal waters of Washington State

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Abstract: Designating Critical Habitat is mandated for species listed under the U.S. Endangered Species Act. This task has only been partly accomplished for Southern Resident Killer Whales (SRKW) because winter distribution is poorly understood due to a variety of factors limiting visual sightings within their known central California to northern British Columbia range. To capitalize on the unique vocal behavior of resident killer whales, including pod-specific dialects, two types of acoustic recorders were deployed at strategic locations that span the Washington coast. Between 2005 and 2008 recorders were deployed in early winter for an average of 175.5 days at up to four sites. These functioned for an average of 114.8 days and collected a total of 47 SRKW detections. This exceeds the number of visual sightings during the same time period (15). Additionally, Northern resident, transient, and offshore killer whales were recorded as well as Pacific white-sided dolphins, and humpback and sperm whales. SRKW were detected by both types of recorders and in all areas. Detections were made between January and July with the majority of these detections in March, April, and May. This new information will be of key importance to managers in meeting recovery goals.
Ocean Acoustic Noise Budgets for the Environmental Assessment of Offshore Wind Power Generation Sites

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Abstract: A noise budget is a listing of the various sources of acoustic noise and their associated ranking by importance. A number of different types of budgets can be conceived using various acoustic measures such as intensity, energy, or duration of maximum amplitude level. These budgets are typically parameterized by frequency and are usually computed over 1/3 octave bands. As part of the environmental assessment of the proposed offshore wind power generation project under the Rhode Island Special Area Management Plan (SAMP), noise measurements were made using the Passive Acoustic Listener (PAL) systems off the coast of Rhode Island prior to the installation of any wind power facilities. Two PALs were deployed within 2 miles of Block Island in water depths of 20 meters from October 6 to November 11, 2008. The data included noise spectra and source identification every 3 minutes. Short snapshots of unusual sounds were also recorded. From this data, the ocean acoustic noise budget is computed with contributions from shipping, wind/waves, marine mammals, and rain from 500 Hz to 50 kHz. The ship noise data is correlated with ship traffic data from the Automatic Identification System (AIS). (Funding provided by the Rhode Island Office of Energy Resources)
HIGH FREQUENCY DIRECTIONALITY MEASUREMENT OF AMBIENT NOISES FROM BREAKING WAVES IN THE SURF ZONE

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Abstract: The high frequency bands of ambient noise in the surf zone are mainly generated by breaking waves. The noise making mechanisms are well-known bubbles’ resonance during the breaking waves propagate forward to shores. Thus the horizontal line array is put on the seabed to measure the noise of breaking waves, and their directionality characteristics. The experiment location is in a bay, and the sediment of seabed is sand. The directionality calculations in the frequency of 1-5 kHz are used the linear beamforming, which can obtain the spatial variations of noise levels in the evolution of breaking waves. So the results represent the noise levels are highly varied with the wave trains, and the directionality analysis is useful to indicate the noise source spatial distributions. Moreover, the different wave heights and periods make significant noise level changes and source distributions.

Keywords: ambient noise, surf zone, directionality
1. INTRODUCTION

The ambient noises in the surf are mainly generated by breaking waves, which’s mechanism is well-known bubbles resonance while surface waves break. In reference [1] has emphasized that hundreds of papers that have studied this topic from World War II. The surf noise sources are generally made by wave motions shown in Fig. 1, which represents the individual plunging breaking wave in this study. Fig. 1 is also illustrated on the wave motion shapes and acoustic characteristics come from full waves, bubble plumes, bubble clouds, and individual bubbles. The acoustic characteristics from the receiving voltages are low frequency noises dominated by waves interactions and bubble plumes impact, and high frequency noises radiate from bubble clouds and individual bubbles. These can be measured and observed both in the water tank and surf zone. In the surf experiment, it may be difficult to obtain the resonance from an individual bubble, but the acoustic characteristics, such as noise level fluctuations and spatial distribution of noise energy, can be received by using a hydrophone array for more interesting studies. Thus the main study subject in this paper is to measure the noise directionality by a line array and analyze the noises’ spatial distribution by using the beamforming calculation.

Fig. 1: Individual plunging breaking wave representation on (a) motion shapes from completing to breaking, (b) acoustic characteristics in receiving voltages.

2. EXPERIMENT

The experiment is at Sizih Bay, outside of Kaohsiung Harbor, which is the busiest harbor in Taiwan. Fig. 2 was taken on the beach to show the hydrophone line array deployment lien on the sand seabed. The hydrophone line array consists of 16 channels in the spacing of 0.15 m, and the digital data acquisition is used the sampling rate of 50 kHz. Fig. 2 is also illustrated on the hydrophone channel No. 1 is at the shoreward direction and No. 16 is at the seaward direction. Unfortunately, the acoustic data from No. 16 indicate it is broken. Thus only 15 hydrophones can be used for the beamforming calculation here. The line array is deployed at a grazing angle of 45° to waves’ crest lines that can represent the good beam resolution on broadside for the breaking zone.
Fig. 2: Hydrophones array deployment in Sizih Bay, outside of Kaohsiung Harbor.

The experiment measures the surf noise at 10-12 am, while the weather is sunny, the air temperature is about 25-26 °C, and the wind speed from meteorological measurement (Taiwan Central Weather Bureau, CWB) is 4-5 m/s. There are water temperature and depth sensors attached to the hydrophone No. 16, where is the deepest position on the array deployment. The water temperatures are slowly increasing from 22 °C to 22.5 °C at one and half hours during the experiment, which are caused by the sunshine. The sound speed in the sea water is about 1,523 m/s estimated from above data. The wave heights in the open sea estimated by Taiwan CWB are about 0.5 m.

Fig. 3 display the spectrograms at the shoreward hydrophone (channel 1) and the seaward hydrophone (channel 15) that indicate the breaking waves generated high noise energy near the channel 1. The wave guide limitation and bubbles damping may decrease the surf noises to propagate to seaward, which have clear symptoms displayed on the difference of sound pressure levels. Moreover, the surf noise in high frequency bands from resonance of bubble clouds and individual bubbles may exist and move with the wave motion in few seconds, which also can be observed in the spectrogram of channel 15.

Fig. 3: Spectrogram at (a) channel 1 at the shoreward, (b) channel 15 at the seaward.
3. RESULTS

The horizontal directionality of breaking wave noise in the surf zone has modelled by Deane [2], which represents very clear simulation to describe the noise distribution in an individual breaking wave motion. Some studied dissimilarities with this paper are the noise frequency and array deployment distance from the break points. Because it’s very difficult to measure the resonance from individual bubbles in the surf zone, the noise collection from bubble plumes and bubble clouds may be easier to measure and simulate. Thus low frequency noise of 500 Hz in the surf has simulated in reference [2]. The noises in high frequencies of 1-5 kHz are carried out in this study, which are collected by the hydrophone array deployed very close to the break points. So some noise pulses are caused by bubble plumes at a frequency of 1 kHz with a bandwidth of 50 Hz that are shown in Fig. 4. From the pulse amplitudes and delay time in each channel can obviously find this wave broke near shoreward side of the hydrophone array. The linear beamforming method is used for the horizontal noise directionality calculation. Fig. 5 is the computed results for noise directionality in 1 kHz, which also indicates the large noise generated from the shoreward side, i.e. positive angles from 60 to 90 degree.

Fig. 4: Hydrophone array received voltages in the frequency of 1 kHz.

Fig. 5: Noise directionality in the frequency of 1 kHz.
Fig. 6: Noise directionality in the frequencies of 2, 3, 4, and 5 kHz.

The surf noises generated by bubbles are well-known as nature resonance with bubble radius, which have full studies in the reference [3]. The noise directionality in the higher frequencies in 2-5 kHz are also calculated and shown in Fig. 6. Very clear indications represent the smaller bubbles are generated after the wave broke (later than 0.8 sec) and moved with the wave motion. So the noises in the spatial distribution become in the negative angles (back to seaward side), even find in the wide angles. The study on the breaking wave noise in the water tank takes a lot of continuous pictures shown in Fig. 7, which can verify the wave motions with the bubble plumes and bubble clouds [4]. In the last five small pictures can observe many small bubbles exist in the water.

Fig. 7: Visualization of breaking waves with bubble clouds in the water tank. [4]
4. SUMMARY

Some representations of acoustics characteristics from the plunging breaking waves in the surf zone have addressed in the experiment and result description. The wave may break randomly in the surf zone that can be observed in the noise directionality. After the wave breaks, a lot bubble clouds generated noise can be seen in the spectrogram shown in Fig. 3. The noise from bubble clouds move with the breaking waves also can be found in the noise directionality in Fig. 5 and Fig. 6. Moreover, the visualization of breaking waves with bubble clouds in the water tank can prove the bubbles’ generated processes and acoustic characteristics, which also verify the concept in Fig. 1.

5. ACKNOWLEDGEMENTS

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Ambient Noise Profiling with Deep Sound

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Abstract: Deep Sound is an autonomous acoustic recording system, housed in a Vitrovex glass sphere, designed to measure ambient noise from the sea surface to depths of 9 km or greater. The instrument descends under gravity while monitoring its depth. When a pre-programmed depth is reached, a burn-wire releases a weight causing the instrument to ascend to the surface under its own buoyancy. Acoustic recordings, sound speed measurements and pitch and roll motions are made continuously during the descent and ascent. Two hydrophones are mounted at half meter vertical spacing allowing the noise spectrum and vertical coherence (directionality) to be obtained over four decades of frequency (10 Hz - 30 kHz). The data logging system, housed inside the 43.2 cm diameter glass sphere, is able to withstand round trips to the deepest trenches of the ocean. Low power consumption and large data storage capabilities enable the system to descend and ascend to great depths at 0.6 m/s. Deep Sound's design and acoustic characteristics will be described and data from deep ocean deployments will be reported. Deep Sound Mk II, currently under development, is an enhanced design that will also be presented. In addition to the environmental sensors of its predecessor, it has four acoustic channels allowing simultaneous measurements of vertical and horizontal coherence. [Work supported by the Office of Naval Research.]
SNOW FALLING ON WATER, DOES IT REALLY MAKE NOISE?

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Abstract: It is well known that rainfall rates can be estimated by analyzing the spectral character and level of the noise that is generated. There have been a few field reports of increased noise levels associated with the occurrence of snow: these reports suggest an increase in sound levels above 20 kHz. There is also a demonstration that snowflakes can generate both a small impact sound and a separate high-frequency pulse that has behaviour consistent with sound generation by a resonant bubble. One aspect that these earlier studies have not explored is the dependence of this sound generation on the type or intensity of snowfall. We report on observations of noise generated by snow falling into a tank of water quantifying snowfall rate using an Optical Scientific precipitation gauge. Observations include records from 7 distinct snow types as well as rain and freezing rain under a range of precipitation rates. Some types of snow were observed to make a detectable signal while others did not. There was no evidence for the high frequency signal previously seen for snow and the high frequency sound levels showed little correlation with snowfall rates. A small peak at around 12 kHz was seen in spectra of some snow types and a correlation between sound level and snowfall rate was seen at frequencies between 10-15 kHz.

Keywords: Ambient noise, precipitation, snow, sound generation
1. INTRODUCTION

The characteristics of sound generated by rain falling on water have been explored through laboratory and field studies and are now well understood [1]. It is now possible to estimate rainfall rates under a wide variety of conditions by consideration of the ambient noise spectrum. In contrast there has been relatively little published on the noise that snow makes when it strikes water. Recordings of snow falling on a lake are reported by [2] and those data showed increased sound levels at frequencies above 20 kHz. Similar observations are reported in an Alaskan fjord [3]. A detailed analysis of the impact noise of individual snowflakes is provided in [4]. In that study, some snowflakes did make a characteristic sound analogous to the source signature for small raindrops with a pulse associated with an impact, and then after a slight delay, a high frequency pulse with decay consistent with that expected for a resonant bubble. The presence of this resonant peak at frequencies of order 50 kHz provides a likely explanation for the field observations of [2] and [3].

What these earlier studies have not explored is the degree to which snowflake noise might be correlated with snowfall rate. In addition, there are many different types of snowflakes, and it would be surprising if all snowflakes made the same sort of sound. This paper reports on a study of snowflake noise. Examples of nine different precipitation types were recorded including 7 distinct snowflake types.

2. METHOD

For this study, sounds were recorded in a tank measuring 80 x 60 x 60 cm containing fresh water. An International Transducer Corporation, ITC-6050C transducer was positioned facing upward with the transducer element located about 13 cm below the water surface. The signal from the hydrophone was fed through a Reson VP2000 pre-amplifier set to a gain of 10 dB and set to pass frequencies between 0.5 and 50 kHz. This signal was recorded at 100 kS/s with 16 bit resolution using an IOtech Personal Daq 3000 controlled by a laptop computer. All power was supplied by a battery to avoid any signal contamination through the power supply. When observations were being made, the tank was wheeled out of the loading bay in the Chemistry-Physics Building at Memorial University and the hydrophone cable was fed through the loading bay door back to the recording equipment inside. Fig. 1 presents a block diagram showing the data acquisition components.

Data were collected for intervals of 30 seconds and stored as “wav” files. These recorded time series were then loaded into Matlab and scaled to correct for calibration values of the Daq and the transducer. The Matlab “PWELCH” function was then used to convert time series to power spectra.
Fig. 1: Schematic diagram showing data recording components.

Standard meteorological observations were available from a weather station maintained on the roof of the Chemistry-Physics building. Precipitation rate was recorded every minute using an Optical Scientific Inc. Optical Rain Gauge (ORG). The ORG does assign categories to precipitation observations but we relied on direct visual observations of snow (landing on a black cloth) to determine snow types. Snow type was assigned using the simplified categories described by the Commission of Snow and Ice of the International Association of Hydrology [5], these are shown in Fig. 2.

Fig. 2: International snow classification scheme, (taken from [5]). Snow types that made sounds are identified with a Yes, those that did not with a No. Ice pellets as such were not observed but are expected to be similar to freezing rain that was observed. Hail and capped column were not observed.
3. OBSERVATIONS

Observations were made of 7 different snow types as well as rain and freezing rain (essentially ice pellets with a liquid core). Examples of the type of spectra that were observed are shown in Fig. 3: spectra plotted as solid lines were snow types that appeared to make some noise (column, needle, irregular, and graupel), snow types that made no noise are plotted as dotted lines (plate, stellar, spatial dendrite): these spectra are close to or at the noise floor of the recording system. Also shown are a rain spectrum (dash-dot line), and freezing rain spectrum (dashed line).

There was some indication of increased sound levels at frequencies above 20 kHz, but because this is present in all spectra it is most likely caused by some aspect of the instrumentation. What did appear consistently in the spectra of snow crystals that made noise was a small peak near to 12 kHz. In the observations presented here, that peak appears to have a slightly lower frequency than the peak associated with light rain. The spectrum collected during a period of freezing rain stands out having a much higher overall level and with no clear sign of any spectral peak.

![Fig. 3: Typical spectral shapes: freezing rain (dashed line, green), rain (dash-dot line, red), snow that made noise (solid line, blue), snow that made no noise (dotted line, black). The snow types that made noise were column, needle, irregular, and graupel. The snow types that made no noise were plate, stellar crystal, and spatial dendrite.](image)

The data were sorted by snow type and averaged spectral levels were determined in various frequency bands. Low frequencies (below 5 kHz) did not correlate with snowfall rate but interestingly did correlate with wind speed. Frequencies above 20 kHz did show some correlation with snowfall rate but that correlation was weak. Recognising the distinctive peak near 12 kHz frequency, the band from 10 to 15 kHz was plotted against snowfall rate showing a clear sign of correlation (Fig. 4). In Fig. 4, snow types are identified by different symbols: irregular and column, (black upward triangle), graupel (red downward triangle),...
needles (blue circle), all others (black dots). During any precipitation event, multiple spectra were collected and all of these data are presented. However, bold symbols identify mean levels during a given event with error bars indicating the degree of variability in both snowfall rate and sound level during that event. The snowflakes types that did not make noise or could not be clearly identified are all classified as “other” in Fig. 4. And, while many of these observations were characterised by low noise levels, they also appear to be characterised by low snowfall rates.

A straight line was fitted through the event averaged observations (bold symbols in Fig. 4). The resulting line is given by:

\[
SPL = (20 \pm 2) \times \log(ppt) + (28 \pm 2)
\]  

where \( ppt \) is the precipitation rate in mm/hour, and \( SPL \) is mean spectral level between 10 and 15 kHz in dB (re 1 \( \mu \)Pa).

\[
\begin{array}{c}
\text{Irreg./Column} \\
\text{Graupel} \\
\text{Needles} \\
\text{Other}
\end{array}
\]

\[
\begin{array}{c}
\text{SPL (dB re 1 \( \mu \)Pa)} \\
\text{Precipitation Rate (mm/hr)}
\end{array}
\]

Fig. 4: Mean sound level between 10-15 kHz plotted against precipitation rate. Bold symbols indicate average values over all observations during a given snow event: the associated crossed lines indicate standard error in those averages. The straight line indicates a fit through the averaged values. Dots indicate values for all additional observations that could not be classified or were associated with snow types that did not make sounds.

4. CONCLUSIONS

We have reported on a series of observations of noise made by snow falling into fresh water in a tank. We did not observe the characteristic high frequency (above 20 kHz) signal previously reported for snow. It did appear that our data had some high frequency contamination and that signal may have been masked. What was observed was a small spectral peak at around 12 kHz that appears similar to the peak seen in certain rain spectra. Not all snow types made significant noise: the columnar and more “solid” forms (column, needle, irregular, and graupel) made noise, while the lighter structures (plate, stellar crystal,
and spatial dendrite) did not make noise. Fig. 4 shows that snow types that did not make noise were also associated with periods of low precipitation rate: it is possible that if these snow types had been observed during heavier snow conditions, a recognisable signal would have been observed. When sound levels between 10 and 15 kHz were plotted against precipitation rate (Fig. 4), a well defined relationship was observed. These results suggest that snowfall rates could be measured using underwater sound levels. What is needed to further explore this characteristic is a series of field measurements in an area where snowfall rate can be measured and snow type can be assessed.

5. ACKNOWLEDGEMENTS

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Structured Session 23

Multiple-Input Multiple-Output (MIMO) Techniques and Sonar Applications

Organizers: Yan Pailhas & Chris Capus
DETECTOR AND WAVEFORM DESIGN OF MIMO SYSTEM

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Abstract: It has been shown that time reversal (TR), which is developed in the acoustics domain, can also improve the detection performance of a radar system. However, the TR technique is no longer a good choice when the noise level is high since the retransmitted signal contains significant noise components. We investigate a multiple input-multiple output (MIMO) detection process similar to TR detection, during which a waveform designed based on the estimated channel and a parameter indicating the quality of the estimation given a priori is retransmitted, and the detector determines the presence or absence of a target. We develop three detectors, whose theoretical thresholds are derived in closed form. Two schemes are proposed to design the retransmitted waveform with constraints on signal power. We compare the detection performance of different detectors, showing that the detector performing the best has the highest complexity, while the detector with the poorest performance demands the least amount of a priori information. Numerical results also present that both the designed waveforms achieve significant performance gains compared with the signal utilized in the TR process.

Keywords: MIMO, time-reversal, target detection, detector design, waveform design
1. INTRODUCTION

The time-reversal (TR) technique, extended from the concept of phase-conjugation in optics, has attracted increasing interest for a broad range of applications. The unique feature of TR is that it can turn multipath effects, traditionally a drawback for target detection and imaging, into a benefit, which is very similar to the multiple-input multiple-output (MIMO) concept developed in communications. In the TR approach, a signal is first radiated through the medium, then the backscattered signal is recorded, time reversed, energy normalized, and retransmitted [1]. Recently, Moura et. al. explored the MIMO radar target detection problem using TR, showing that TR detection provides significant gains over conventional detection [2,3]. This results from the fact that the transmitter reshapes the waveform to match the channel during the TR process. However, the retransmitted signal in Moura’s algorithm contains noise components, and it is obvious that if the noise level is high, the TR technique is no longer a good choice. Furthermore, [2,3] did not derive analytical expressions for the threshold and probability of detection of the TR detection, which were determined instead by Monte Carlo simulations.

We investigate a MIMO detection process similar to the TR detection in this paper. That is, during the probing phase, an incident wave is first transmitted into the medium and an estimated channel matrix with estimation error is obtained. It is assumed that a parameter indicating the quality of the estimation is given a priori, which can be appropriately chosen depending on the noise level, the channel dynamics, and estimation strategies, etc. [4]. Then, a waveform designed using the estimated channel and the estimation quality parameter under power constraints, instead of the TR signal used in Moura’s scheme, is retransmitted, and finally the detector determines the presence or absence of a target. Note here that similar to the TR detection, it is assumed that the channel remains static during the probing and detecting phases.

2. SYSTEM MODEL

We consider a wideband bistatic MIMO sonar (or radar) system including a pair of arrays A and B as shown in Fig.1, which has \( N_a \) and \( N_b \) sensors, respectively. The channel frequency response is denoted by \( H_{qH}(f_q) \) matrix, \( q = 1, 2, \ldots, Q \), where the \((k,l)\)-th entry of \( H(f_q) \) is the frequency response of the channel between the \( k \)-th sensor of array B and the \( l \)-th sensor of array A at the discrete frequency \( f_q \). We adopt the statistical MIMO model here, that is, the entries of the channel matrix are assumed to be zero-mean circularly symmetric complex Gaussian (ZMCSCG), and they are normalized to have unit variance. The random target response results from the multipath effect, which arises from different propagation mechanisms. For example, the multipaths are due to a rich scattering environment surrounding point-like targets in [3], while in [5], the distributed target itself leads to multipath propagation.

As shown in Fig.1, the target detection process has two steps, and an estimated channel matrix \( \hat{H}(f_q) \) is obtained after the probing phase. In this paper, we consider the situation where minimum mean square error (MMSE) estimation is employed, and denote the estimation error as \( E(f_q) = H(f_q) - \hat{H}(f_q) \), whose entries are ZMCSCG with variance \( \sigma_e^2 \).
Note here that knowing the value of $\sigma^2_e$ requires noise power estimation and knowledge of the waveform length during the probing phase. Then, using the properties of MMSE estimation, the entries of $\hat{H}(f_q)$ can be shown to be ZMCSCG with variance $1 - \sigma^2_e$ [4]. Since the focus of this paper is to design different detectors and retransmitted waveforms and study their effects on the target detection performance of the MIMO system, we assume that the estimated channel matrix and the quality parameter $\sigma^2_e$ are given a priori, and we concentrate on investigating the detection performance of the second step.

3. DETECTOR DESIGN

The target detection problem of the MIMO system can be described as follows:

Under $H_0$: $\tilde{r}_i = \bar{n}_i$; \hspace{1cm} Under $H_1$: $\tilde{r}_i = \bar{Y} \cdot \hat{h}_i + \bar{Y} \cdot \bar{e}_i + \bar{n}_i$ (2)

where $i = 1, 2, ..., N_a$, and the alternate hypothesis $H_1$ and null hypothesis $H_0$ are that the target does or does not exist, respectively. In this section, we develop three approaches to detect the...
target: the conventional detector, the optimal detector, and the generalized likelihood ratio test (GLRT) detector. Only the key equations are presented here due to space limitation, and interested readers may refer to [6] for the detailed derivation.

It is well known that the optimal detector for a known signal in white Gaussian noise is the matched filter, and such a detector is denoted as Detector I, whose performance is examined when estimation error exists. The conventional detector given by [7] can be expressed as

\[
T_i = \text{Re}(\sum_{i=1}^{N} \bar{d}_i^H \bar{r}_i)^{\frac{1}{2}} < \eta_i, \quad \bar{d}_i = \bar{Y} \cdot \hat{h}_i, \quad \eta_i = \sqrt{\frac{\sigma_n^2 \sum_{i=1}^{N} \bar{d}_i^H \bar{d}_i}{2}} \cdot Q^{-1}(Pr_{FA})
\]

where the superscript \( H \) denotes the conjugate transpose of a matrix, \( Pr_{FA} \) is the required probability of false alarm, and \( Q^{-1}(x) \) stands for the inverse Gaussian right-tail function. Clearly, Detector I demands the information of \( \bar{Y} \) and \( \hat{h}_i \) to decide the existence of targets.

Note here that the detector matches to the estimated channel \( \hat{h}_i \) instead of the true channel \( h_i \) as in [7], and this is because only the estimated channel is available at arrays A and B.

Next, we proceed to design Detector II, which is the likelihood ratio test (LRT) detector. The LRT detector is the optimal solution in the Neyman-Pearson sense [7]. After some algebra, the optimal detector can be described as below:

\[
T_{II} = \sum_{i=1}^{N} \left( \left( \bar{r}_i + \bar{B} \bar{g}_i \right)^H \cdot \bar{B} \cdot \left( \bar{r}_i + \bar{B} \bar{g}_i \right) \right)^{\frac{1}{2}} \eta_{II} < \eta_{II}, \quad \eta_{II} = \sqrt{\frac{\text{trace}(\bar{C})}{\sigma_n^2}} \cdot Q^{-1}(Pr_{FA})
\]

\[
\bar{d}_i = \bar{Y} \cdot \hat{h}_i, \quad \bar{C} = \sigma_n^2 \bar{Y} \bar{Y}^H + \sigma_n^2 \bar{T}_{MQ}, \quad \bar{B} = \frac{1}{\sigma_n^2} \bar{T}_{MQ} - \bar{C}^{-1}, \quad \bar{g}_i = \bar{C}^{-1} \bar{d}_i
\]

where the superscript \( ^* \) denotes the pseudoinverse, and \( \bar{T}_{k} \) stands for a \( k \times k \) identity matrix.

It is reasonable to assume that \( T_{II} \) under both hypotheses have Gamma distributions as it has a quadratic form in Gaussian random variables, i.e., \( T_{II} \sim \Gamma(n, \theta) \) under \( H_0 \) and \( T_{II} \sim \Gamma(n, \theta) \) under \( H_1 \), where \( \Gamma(n, \theta) \) denotes the Gamma distribution with the shape parameter \( k \) and scale parameter \( \theta \). Hence, the theoretical threshold of the optimal detector can be given by

\[
\eta_{II} = F_{\Gamma(n, \theta)}^{-1}(1 - Pr_{FA})
\]

\[
\mu_0 = \sum_{i=1}^{N} \left\{ \sigma_n^2 \text{trace}(\bar{B}) + \bar{g}_i^H \bar{B} \bar{g}_i \right\}, \quad \Omega_0 = \sum_{i=1}^{N} \left\{ \sigma_n^2 \text{trace}(\bar{B} \bar{B}^H) + 2 \sigma_n^2 \bar{g}_i^H \bar{g}_i \right\}
\]

where \( F_{\Gamma(n, \theta)}^{-1} \) denotes the inverse cumulative distribution function (CDF) of the Gamma random variable with parameters \( k \) and \( \theta \). From (4) and (5), it is clear that the implementation of Detector II requires knowledge of \( \bar{Y}, \hat{h}_i \), and \( \sigma_n^2 \) at array A.

Detector III is the GLRT detector, which is a practical approach when unknown parameters exist [7]. The GLRT detector replaces the unknowns with their maximum likelihood (ML) estimates, and in our case, it is given by
where $n$ is the rank of $\bar{Y}$, i.e., $n = \text{rank}(\bar{Y}) \leq \min\left(MQ_f, N_bQ_f\right)$, and $F^{-1}_\chi^2$ is the inverse CDF of a central chi-square random variable with $k$ degrees of freedom. Obviously, only the value of $\bar{Y}$ is required to be known for Detector III. We next consider $T_{III}$ under $H_1$. Denote by $\bar{U}\Sigma\bar{Y}^H$ the singular value decomposition (SVD) of $\bar{Y}$, where the $MQ_f \times MQ_f$ matrix $\bar{U}$ and $N_bQ_f \times N_bQ_f$ matrix $\bar{V}$ are unitary matrices, and $\Sigma$ is an $MQ_f \times N_bQ_f$ diagonal matrix with $n$ positive singular values $\varsigma_1, \varsigma_2, \ldots, \varsigma_n$ of $\bar{Y}$ (in decreasing order) on the diagonal. Defining an $n \times 1$ vector $\bar{\beta}$, whose $k$-th entry $\beta_k$ is the square of the corresponding singular value of $\bar{Y}$, i.e., $\beta_k = \varsigma_k^2$, we can rewrite (6) as below:

$$ T_{III} = \sum_{i=1}^{N_b} \sum_{k=1}^{n} \frac{1}{\sigma_n^2} \left( \gamma_{\beta_k} \right) \, \gamma_{\beta_k} \sim \chi^2 \left( \lambda_{\beta_k} \right), \quad \lambda_{\beta_k} = 2\beta_k \left| h_{ik}^* \right|^2 / \left( \sigma_n^2 \beta_k + \sigma_n^2 \right) $$  

(7)

where $|.|$ denotes the modulus of a complex number, $h_{ik}^*$ is the $k$-th element of the vector $\bar{h}_i = \bar{Y}^H \bar{h}_i$, and $\chi^2 \left( \lambda \right)$ stands for a noncentral chi-square random variable with $k$ degrees of freedom and non-centrality parameter $\lambda$. Notice here that $T_{III}$ is a weighted sum of several noncentral chi-square random variables, and it is difficult to derive a closed form for its distribution. In order to calculate the theoretical $\text{Pr}_D$, we approximate $T_{III}$ using a common technique which has been widely adopted in statistics and engineering. This approach approximates a weighted sum of chi-square variables by a single one with different degree of freedom and a scaling factor, which are carefully chosen such that the first two moments remain the same. Therefore, the test statistic of the GLRT detector can be expressed as below:

$$ T_{III} \sim \frac{\alpha}{\sigma_n^2} \chi^2, \quad \alpha = b/a, \quad l = 2a^2/b, \quad \rho_k = \sum_{i=1}^{N_b} \left| h_{ik}^* \right|^2 $$  

(8)

$$ a = \sum_{k=1}^{n} \left\{ N_a \left( \sigma_n^2 \beta_k + \sigma_n^2 \right) + \beta_k \rho_k \right\}, \quad b = \sum_{k=1}^{n} \left\{ N_a \left( \sigma_n^2 \beta_k + \sigma_n^2 \right)^2 + 2 \left( \sigma_n^2 \beta_k + \sigma_n^2 \right) \beta_k \rho_k \right\} $$

4. WAVEFORM DESIGN

In this section, we propose two approaches to design the retransmitted waveform $\bar{Y}$ by maximizing the upper and lower bound of the $\text{Pr}_D$ of the GLRT detector developed in the last section, respectively. The design criteria are under the constraint $\text{trace}(\bar{Y}^H) = MQ_f E_s$ which limits the total transmitted power. As defined in the last section, entries of $\bar{\beta}$ are actually the eigenvalues of the Hermitian matrix $\bar{Y}^H$, and thus, the power constraint can be rewritten as $\sum_{i=1}^{N_b} \beta_k = MQ_f E_s$, and $\beta_k \geq 0$, $k = 1, 2, \ldots, n$. The first waveform is designed to maximize the upper bound, which is obtained by utilizing Markov’s inequality, i.e., $\text{Pr}_D = \Pr\{T \geq \eta\} \leq E[T] / \eta$. 

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Substituting (7) into the above equation and recalling the power constraint, we can express the design criterion for waveform design I as the following constrained maximization problem:

$$\max_{\beta} \sum_{k=1}^{n} \left( N_a \sigma_k^2 + \rho_k \right) \beta_k, \quad \text{s.t.} \quad \sum_{k=1}^{n} \beta_k = MQ E_s, \quad \beta_k \geq 0, \quad k = 1, 2, \ldots, n$$  \hspace{1cm} (9)

Applying Abel’s inequality, we find that the maximization of (9) is achieved by allocating all the available power to the eigenvalue $\beta_k$ which corresponds to the largest $\rho_k$.

Next, we design the second waveform based on the lower bound. Again, adopting Markov’s inequality leads to the bound $Pr_{fa} = 1 - Pr \left[ -\sigma_n^2 T \geq -\sigma_n^2 \eta \right] = 1 - Pr \left[ e^{-\sigma_n^2 T} \geq e^{-\sigma_n^2 \eta} \right] \geq 1 - E \left[ e^{-\sigma_n^2 T} \right] / e^{-\sigma_n^2 \eta}$. After some derivation, we can describe the problem of maximizing the lower bound as below

$$\max_{\beta} \sum_{k=1}^{n} \left( 2 \beta_k \rho_k + 2 N_a \sigma_k^2 \beta_k + \sigma_k^2 \right), \quad \text{s.t.} \quad \sum_{k=1}^{n} \beta_k = MQ E_s, \quad \beta_k \geq 0, \quad k = 1, 2, \ldots, n$$  \hspace{1cm} (10)

This constrained optimization problem can be solved by employing the Karush-Kuhn-Tucker (KKT) conditions [9], and the waveform design II can be given by

$$\beta_k = \left( 1 + 2 \sigma_n^2 \right) / 2 \sigma_n^2 \left( \xi \sqrt{2 \rho_k + 4 \rho_k \sigma_n^2 + 2 N_a \sigma_n^2 / \left( 1 + 2 \sigma_n^2 \right)} - 1 \right)$$  \hspace{1cm} (11)

where $(a)^+ = \max(0, a)$ and $\xi$ is chosen such that the constraint is met, i.e., $\sum_{k=1}^{n} \beta_k = MQ E_s$.

It is clearly seen that this design scheme utilizes the waterfilling strategy [4] to allocate the power, and the larger the $\rho_k$ is, the more power is allocated to its corresponding $\beta_k$.

5. NUMERICAL RESULTS AND DISCUSSION

In this section, we present numerical results showing the target detection performance of a MIMO system with four sensors at array A and two sensors at array B. We choose the number of snapshots $M = 2$ and the number of frequencies $Q_f = 4$ for simulation purposes. The signal-to-noise ratio (SNR) is defined as $SNR = E_s / \sigma_n^2$, and the probability of false alarm is set to be a constant value $Pr_{fa} = 0.01$. Notice here that the algorithms for both the detector and waveform design are based on a known estimated channel, which is actually a realization of the random vector. Therefore, a semi-analytical approach is utilized to obtain the system performance. In other words, we generate 10,000 realizations of the estimated channel matrix, calculate the corresponding $Pr_{D}$ for each realization, and determine the system detection performance by averaging $Pr_{D}$ over all the realizations.

Fig.2 depicts the detection performance of systems employing different detectors at array A for two values of $\sigma_n^2$. The waveform adopted here is the normalized TR signal, and the normalization is used to meet the power constraint. For each $\sigma_n^2$, any difference in performance results from the detector design only since the retransmitted signals are the same for all systems. Observing the curves, we find that Detector II performs the best under any
circumstance, which is consistent with the fact that Detector II is the optimal detector in the Neyman-Pearson sense. In addition, the performance difference between Detector I and Detector II increases as the estimation quality becomes lower, i.e., $\sigma^2_e$ is larger. This can be explained that Detector I is effectively a matched filter to the signal $\hat{Y}h^*_j$, while the optimal detector is a filter matching to the true signal $Yh^*_j$ when the channel matrix is known [7]. Therefore, the difference between the performance of Detector I and the optimal performance should be small when the error in the estimated channel is insignificant, and this happens when $\sigma^2_e$ is small. Furthermore, it is easily seen that Detector III performs the poorest at low SNR but is similar to the optimal detector when the SNR is high. This is because the GLRT detector actually estimates the unknown parameters first and then makes the detection decision based on those estimates. Intuitively, the lower the SNR, the worse the estimation, which leads to worse detection performance. However, although Detector II performs the best, it has the highest complexity and it requires knowledge of $\bar{Y}$, $\hat{h}_j$, and $\sigma^2_e$ at array A. In contrast, as mentioned in Section 3, the implementation of Detector I needs the information of $\bar{Y}$ and $\hat{h}_j$, while for Detector III only $\bar{Y}$ is required to be known.

We next examine the detection performance of the systems retransmitting different waveforms with two values of $\sigma^2_e$ as shown in Fig.3. Here, Detector III is employed for all scenarios, and any difference in performance arises from the designed waveforms only. In Fig.3, the labels TR, WD1, and WD2 correspond to the normalized TR signal, the waveform design I, and the waveform design II proposed in the last section, respectively, and all the waveforms have the same transmitted power constraint. Obviously, the designed waveforms improve the system performance significantly with respect to the TR signal. Specifically, for $Pr_D = 0.8$, the performance gap between waveform design I and the TR signal is about 5dB when $\sigma^2_e = 0.1$ and 4dB when $\sigma^2_e = 0.5$, while the performance gain achieved by waveform design II compared with the TR signal is about 4dB when $\sigma^2_e = 0.1$ and 1.5dB when $\sigma^2_e = 0.5$. Nevertheless, from the waveform design algorithms, we realize that such significant performance improvement is achieved at the price of knowing the quality of channel estimation $\sigma^2_e$ \emph{a priori}. We emphasize here that although both the waveforms are designed based on the GLRT detector, the semi-analytical $Pr_D$ of the systems employing Detector I and
Detector II can also be obtained when the designed waveforms are retransmitted from array B, and similarly, both waveforms achieve considerable performance gains. These numerical results are available in [6], but are not presented here due to the space limitation.

6. CONCLUSIONS

In this paper, we investigated the target detection performance of a bistatic wideband MIMO system, whose detection process is similar to the TR procedure. Based on the estimated channel and a parameter indicating the quality of the estimation obtained during the probing phase, the retransmitted waveform and the detector are designed. Three detectors are developed, whose theoretical thresholds are derived in closed form. Two approaches are proposed to design the retransmitted waveform with signal power constraint, which maximize the upper and lower bound of the probability of detection of the GLRT detector, respectively. Numerical results demonstrate that the optimal detector performs the best but demands the largest amount of a priori information. The performance difference between the conventional and the optimal detector increases as the estimation quality becomes lower. The GLRT detector performs the poorest at low SNR but is similar to the optimal detector at high SNR. Both the designed waveforms achieve significant performance gains compared with the TR signal at the price of knowing the quality of channel estimation a priori.

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ACQUISITION CONCEPTS FOR MIMO SONAR

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Abstract: The objective of sonar processing is to enhance the signal received from the target with respect to noise or clutter in order to enhance detection, classification and localisation. Range-bearing information is commonly obtained through beamforming and matched-filtering of data acquired with a densely sampled detector array. The source sampling, on the other hand, is usually sparse, since data processing is typically performed using single ping data. In this paper we discuss techniques that can be used to increase the source sampling in sonar applications. This is of interest since beamforming with both sources and detectors potentially enhances the resolution of the beamforming and the signal-to-noise ratio in comparison to beamforming with detectors only. This can be achieved either with multiple sources, or by digitally combining data acquired at different times. Examples are continuous transmission, blended acquisition, source coding for multiple sources in a multistatic setting, and the virtual source technique. We illustrate these techniques with seismic data examples and discuss the prospects for their use in sonar applications.

Keywords: sonar, MIMO, continuous sonar, virtual source
1. INTRODUCTION

The term MIMO (multiple-input multiple-output) refers to the use of multiple transmit and receive elements. The concept has attracted great interest in wireless communications, since it achieves higher bit rates and link reliability by using multiple channels simultaneously.

The MIMO concept is also applicable to detection and imaging applications. In most seismic applications, for example, separate experiments are conducted to acquire data corresponding to each source position. In a subsequent step, the data corresponding to the individual source positions, i.e. individual pings, are combined in the processing in order to obtain an image of the subsurface. In order to improve the data acquisition rate, there is an increasing interest in seismic applications for data acquisition using several sources simultaneously [1]. This requires specific data processing approaches such as source coding, data blending [2][3] and the ‘virtual source’ technique [4][5].

These MIMO concepts also have prospects for sonar applications. In many ASW sonar applications, single-ping data are commonly analysed for the detection problem, whereas the processing and analysis of multiple-ping data potentially increases the detection probability. It enables synthetic beamforming with sources. The analysis of data acquired in subsequent pings is hampered by low duty cycles, typically around 10 % for a pulsed sonar. For a source mounted on a platform, this results in large gaps in the source sampling, decreasing the correlation between data acquired in subsequent pings. In addition, valuable detection opportunities might not be exploited due to the low duty cycle. It is therefore desirable to acquire data at a higher duty cycle. Finally, MIMO concepts are also of interest for multistatic operations in which sonars are used on several platforms. In such a scenario, it is relevant whether data transmitted by these sources can be analysed simultaneously.

In this paper we discuss MIMO acquisition concepts such as simultaneous acquisition, the virtual source technique, blended acquisition, and continuous sonar. We illustrate these concepts with data examples and discuss the relevance for sonar applications. It should be explicitly stated that we use the term MIMO in this paper both for applications with multiple sources and for single-source applications where digital processing techniques are used to identify channels that correspond to the source signal transmitted at different times.

2. ACQUISITION CONCEPTS

In this section, we discuss several data acquisition concepts. These are schematically displayed in Figure 1. In conventional sonar applications, a short pulse or ping is emitted and no energy is transmitted during the reception of the backscattered signal, the ping repetition time (prt). For a source mounted on a moving platform, this typically results in large gaps between individual source positions (Figure 1a).

The source sampling can be increased when multiple sources are used, i.e. simultaneous transmission (Figure 1b). In order to maintain the flexibility during the beamforming, pulse coding can be applied in order to be able to retrieve the data corresponding to the individual sources. Examples for source coding are orthogonal wavelets such as an upsweep versus a downsweep, pseudo-random noise codes, binary phase-keyed signals (BPSK), and frequency-
hop coding. As a result of source coding, data corresponding to individual pulses can be retrieved using matched filtering. An important alternative to source coding is beam coding. Instead of retrieving data corresponding to individual sources, matched filtering can be used to retrieve data corresponding to individual beams. This enables a sonar to perform several tasks simultaneously. It is commonly used in phased array radar and is referred to as multi-target capability [6].

The source sampling can also be increased by decreasing the ping-repetition time. Conventional surveys are designed such that the time interval between subsequent pings is sufficiently large that the tail of the previous record does not interfere with the next record. As a result of decreasing the ping repetition time, data corresponding to subsequent pings will interfere: these data are “blended”. The advantage of blended acquisition [1][3], on the other hand, is that the source sampling interval decreases. This concept is introduced to enable spatial filtering with both sources and receivers, such as interpolation and beamforming (Figure 1c). Continuous active sonar is a special case of blended acquisition where the source continuously transmits energy (Figure 1d). The success of continuous sonar relies on the capability of separating the path that propagates directly from the source to the detector from the energy scattered at the target. For this reason, we expect that blended acquisition and continuous sonar are of special interest to bistatic and multistatic scenarios, where sources and receivers are well separated.

![Figure 1: MIMO acquisition concepts](image-url)
A different bistatic/multistatic MIMO acquisition concept is based on the “virtual source” technique [4][5] based on the time-reversal concept [7]-[9]. Virtual sources can be obtained both using noise and active sources.

The virtual source concept was originally introduced in seismic imaging for imaging with receivers located in a well (downhole recordings). The technique uses data generated by several sources located close to the Earth’s surface to create virtual sources close to the region of interest (typically a reservoir). The main benefit of the technique is that it avoids complexities in the imaging procedure introduced by complex propagation between the sources and the downhole receivers. Even better, the imaging with virtual sources improves due to complex propagation since it influences the directivity pattern of the virtual source. The technique is schematically illustrated in Figure 2, and a comparison of an image obtained with virtual sources with an image obtained with conventional ‘surface-to-downhole’ data [4][5] is shown in Figure 3. Both images show the response of two horizontal reflectors, the boundaries of the reservoir. The virtual source image is at least of a similar quality, whereas the imaging of virtual source data is greatly simplified since it does not require detailed knowledge on the complex velocity structure of the overburden.

![Fig. 2. Illustration of the virtual source concept: (left) focusing at an individual receiver, (right) propagation of energy after the formation of the caustic at the receiver.](image)

The concept of virtual sources is also of interest to sonar. As in the seismic example, long-range propagation of energy through a waveguide is complex as well. An important difference, though, is that the imaging problem is stationary in the seismic example, whereas time variations have to be considered in sonar applications. This reduces integration times that can be used to create virtual source data. As a result, the virtual source concept has most prospects in sonar for estimating stationary properties such as seabed properties. Seabed properties are estimated using this concept both with ambient noise recordings [9]-[12] and using data generated by active sources transmitting beam-steered noise sequences [13].

The virtual source technique can also be applied to target detection. It has been demonstrated that the correlation response of passive recordings contains information on the target’s range [14]. We expect that for this application the virtual source technique is of special interest to multistatic acquisition geometries with several transmitters. As in the seismic example, the receivers need to be deployed close to the region of interest, for example towed by an AUV.
Fig. 3. Images obtained with virtual source data (a) and with conventional surface-to-downhole data (b). Note that the virtual source data only requires the availability of a velocity model below 430 m, the depth of the downhole receivers (and thus of the virtual source), whereas the surface-to-downhole data requires an exact velocity model for the full overburden including the near-surface (reprinted from [5]).

3. SUMMARY AND CONCLUSIONS

In sonar, it is common to process data of individual pings separately. The data are generated by a single source that is operated at a low duty cycle. The detection probability may be improved by reducing the duty cycle or by using multiple sources, i.e. by providing more detection opportunities. In this paper, we discussed the following acquisition concepts for this purpose:

- Simultaneous transmission: multiple source are used simultaneously. The data corresponding to individual sources or beams can be retrieved by matched filtering.
- Blended acquisition and continuous active sonar aim to improve the detection probability by providing more detection opportunities. The cost is that energy backscattered at the target interferes with the direct path from the transmitter to the detector. We expect that source coding is effective in mitigating the direct path contribution in a bistatic geometry.
• The virtual source technique is a correlation-based processing technique. It can be used in conjunction with both noise sources and continuous active sources in a multistatic geometry. It has prospects in estimating seabed properties and target detection.

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BROADBAND MIMO SONAR SYSTEM: A THEORETICAL AND EXPERIMENTAL APPROACH

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Abstract: MIMO systems have raised a lot of interest in the recent years. The radar community pointed out the multiple advantages of MIMO systems such as diversity gain for target detection, angle of arrival and Doppler estimation. Coherent processing also allows super-resolution for target localisation. We explore in this paper broadband MIMO sonar systems. In the current literature, the channel matrix has been computed using point scatterer models. The limitation of such models for sonar (and broadband sonar in particular) is explained and a model based on the target form function is proposed. We will show that this deterministic MIMO model can naturally be extended to a statistical model. It has been shown that broadband sonars offer in situ great capability for target classification. Using widely separated transducers, it has been shown that channel matrices are decorrelated from one another, which means that the views of the potential target are independent. MIMO systems improve the process of classification thanks to these multiviews.

We present experiments done in our tank (Width x Length x Depth: 3 x 4 x 2 m) with a broadband MIMO system (2 transmitters, and 4 receivers). The transmitters cover the frequency band of 30 kHz to 150 kHz. We demonstrate experimentally the advantage of Time Reversal for MIMO system by focusing the energy on the target independently of the medium. We propose a pseudo Time Reversal technique which focuses the energy directly back to the receivers increasing the SNR by a factor of N where N is the number of transmitters.

Keywords: Broadband sonar, MIMO system
VIRTUAL POINT SCATTERERS MODEL FOR CYLINDRICAL SHELL

The first formulation for MIMO systems has been done by the radar community. The MIMO system model can usually be expressed by: \( \mathbf{r} = \mathbf{Hs} + \mathbf{n} \), where \( \mathbf{r} \) represents the receivers, \( \mathbf{s} \) the transmitters, \( \mathbf{n} \) the noise, and \( \mathbf{H} \) the channel matrix. The channel matrix includes the wave propagation in the medium from any transmitters to any receivers and the target reflection. At first, targets were represented using the "point target" assumption [1]. Since then, several target models have been proposed such as rectangular-shape target in [2] composed of an infinite number of scatterers. The most popular model for a radar target model is the finite scatterer model [3,4].

In this section we present an accurate multi-static model for a low impedance shell cylinder. In [5], we demonstrated that the sound scattering of a low impedance shell cylinder is analogous to the reflection by two spherical mirrors. Fig. 1 shows the echo formation of an acoustic wave reflected by a plastic cylindrical shell. The location of the two echo centres A1 and A2 (in Fig. 1) can be computed thanks to the well-known formula of reflection by a spherical mirror. The notation of Eq. 1 is explained in Fig. 2. \( \mathbf{A} \) and \( \mathbf{A}' \) represent respectively the source and the source image.

\[
\frac{1}{SA'} + \frac{1}{SA} = \frac{2}{SC}
\]

Assuming an incoming plane wave, the two echo centres, A1 and A2, are exactly in between the centre of the cylinder and respectively the front and the back of the cylinder. In our model A1 and A2 will represent the virtual scatterers. They act like point sources, but contrary to scattering points, they emit the received pulse with a delay. The transmitter Tx transmits a pulse \( s(t) \). The acoustic wave is reflected by the cylinder modelled by the virtual scatterers A1 and A2 to the receiver Rx. Eq.2 expresses the acoustic field \( r(t) \) received at the
receiver Rx. SC represent the radius of the cylinder, c the speed of sound in water, C the centre of the cylinder and \( \tau_{AB} \) the propagation time between A and B.

\[
H(t) = \left( t - \tau - \frac{3SC}{2c} - \tau_{A,Rx} \right) e^{i\phi_1} + \left( t - \tau - \frac{3SC}{2c} - \tau_{A,Rx} \right) e^{i\phi_2} \tag{2}
\]

The two terms \(-3SC/2c\) and \(+3SC/2c\) represent the negative and positive delays of the virtual scatterers. Fig.3 compares the echo spectra of our model with the analytic solution given by Doolittle in [6]. In this example, the cylindrical shell is made of PVC, its diameter is 32cm and its thickness is 3mm. The receiver is placed at 4m from the shell at an angle of 30°. An excellent match is found between the theoretical prediction and our model.

**PROBLEM REFORMULATION FOR BROADBAND MIMO SONAR**

Haimovich et al. in [3] formulates narrowband MIMO radar using a finite point target model with Q scattering points \( \{X_q\} \). The transmitter \( k \) send a pulse \( s_k(t) \), the receiver \( l \) receives from the transmitter \( k \)

\[
z_{lk}(t) = \sum_{q=1}^{Q} h_{lk}^{(q)} s_k \left[ t - \tau_{kX}(X_q) - \tau_{rl}(X_q) \right]
\]

\[
\text{with } h_{lk}^{(q)} = \zeta_q \exp \left(-2j\pi f_c \left[ \tau_{kX}(X_q) + \tau_{rl}(X_q) \right]\right)
\]

The notations can be found in [3]. Assuming the Q scattering points are close, we can write that \( s_k(t - \tau_{kX}(X_q) - \tau_{rl}(X_q)) = s_k(t - \tau_{kX}(X_0) - \tau_{rl}(X_0)) = s_k(t,X_0) \) where \( X_0 \) is the centre of gravity of \( \{X_q\} \). So the previous equation becomes:

\[
z_{lk}(t) = \sum_{q=1}^{Q} \zeta_q \exp \left(-2j\pi f_c \left[ \tau_{kX}(X_q) + \tau_{rl}(X_q) \right]\right) h_{kX}(t,X_0)
\]
The term $\sum \zeta_q \exp(-j2\pi f_c [\tau_{ik}(X_q) + \tau_{il}(X_q)])$ corresponds to a random walk in the complex plane. It explains the phenomena of fading observed in radar. Indeed this value can be statistically lower than the noise level.

We saw in [5,7] that for broadband sonar a formulation in the Fourier domain is more appropriate. Eq.3 becomes:

$$Z_{lk}(\omega) = \frac{E}{M} \sum_{q=1}^{Q} h_{ik}^{(q)}(\omega) S_k(\omega) e^{-j\omega[\tau_{ik}(X_q) + \tau_{il}(X_q)]}$$

(5)

Using the following notations:

$$\tau_{ik}(X_q) = \tau_{ik}(X_0) + \tilde{\tau}_{ik}(X_q)$$

$$\tau_{il}(X_q) = \tau_{il}(X_0) + \tilde{\tau}_{il}(X_q)$$

and

$$H_{lk}(X_0, \omega) = \frac{E}{M} e^{-j(2\pi f_c + \omega)[\tau_{ik}(X_q) + \tau_{il}(X_q)]}$$

we arrive to:

$$Z_{ik}(\omega) = H_{lk}(X_0, \omega) \left( \sum_{q=1}^{Q} h_{ik}^{(q)}(\omega) e^{-j\omega[\tau_{ik}(X_q) + \tau_{il}(X_q)]} \right) S_k(\omega) = H_{ik}(X_0, \omega) F_{\infty}(\omega, \theta_{ik}, \phi_{lk}) S_k(\omega)$$

(6)

$\theta_{ik}$ is the angle of view of the target from the transmitter, and $\phi_{lk}$ is the angle of view of the target from the receiver. Eq.6 can be interpreted as follows: the first term corresponds to the propagation of the wave to and from the target, the second term is the form function of the target, the third term is the transmitted signal.

The main advantage of this formulation is the clear separation between propagation terms and target reflection terms. In our formulation the target form function $F_{\infty}$ is independent of any particular model. The generalization of this equation including multipath and attenuation terms is straightforward:

$$Z_{ik}(\omega) =\sum_{p=1}^{P} A^{(p)}(\omega) H_{ik}^{(p)}(X_0, \omega) F_{\infty}(\omega, \theta_{ik}, \phi_{lk}) S_k(\omega)$$

(7)

where $P$ is the number of multipath and $A^{(p)}(\omega)$ the attenuation through the path $p$.

**EXPERIMENTAL RESULTS AND PSEUDO TIME REVERSAL**

**Experiments**

Sonar MIMO experiments have been done in our test tank (L x W x D: 4m x 3m x 2m) using 2 transmitters (a low frequency transducer: 30kHz-90kHz and a high frequency transducer: 60kHz-150kHz) and 3 wideband receivers. A display of the configuration of the experiment can be found in Fig.4. Fig.4 displays as well the geometrical reconstruction of the
Pseudo Time Reversal

In MIMO systems, the total signal received at each receiver is the sum over the transmitters, i.e. $\Sigma z_{kl}(t)$. The classical assumption made in MIMO is the orthogonality of the transmitted pulses $s_k(t)$. So in the detection problem, the total received signal is projected into each transmitted pulse space, in order to recover the channel matrix elements $h_{kl}$. The optimal detector is then given by the likelihood ratio of the recovered channel matrix [2].

Fig. 5: Spatial sound focus using two transmitters.

We consider here a deepwater propagation type, which means no multi-path. By playing with the delay between the transmitted pulses, the combined sound is focused on certain parts of the space. Fig. 5 displays an example of sound focusing using two transmitters. By knowing the geometry of the transmitters, time delays can be computed to focus on a particular point in space.

The idea of our Pseudo time reversal is to use this combined energy on the target to improve the detection. We want to maximize the total signal from the receiver point of view that means maximizing $\Sigma_k z_{kl} \left(t-\tau_k\right)$ where $\tau_k$ is the delay used to focus the beams. Usually the pulses used are coherent (their cover the same frequency band) so the sum term $\Sigma_k z_{kl} \left(t-\tau_k\right)$ is a coherent summation, which can result in destructive interferences.

Fig. 6 illustrates the interferences due to the coherence in the summation of the same chirp. If the two chirps do not overlap in frequency, the two pulses are incoherent, and no interferences are observed. By using different frequency bands for each transmitter, the sum $\Sigma_k z_{kl} \left(t-\tau_k\right)$ becomes incoherent, and as a result: $\max(\Sigma_k z_{kl} \left(t-\tau_k\right)) = \Sigma_k \max(z_{kl} \left(t\right))$. 

Fig. 4: (left) configuration of the experiment. (centre) geometrical reconstruction of the MIMO echoes without the PVC cylinder. (right) geometrical reconstruction of the MIMO echoes with the PVC cylinder.
Assuming that the K transmitters can emit the same energy in all the frequency bands, the SNR increases by a factor of K.

![Graph](image)

**Fig.6:** Maximum amplitude of the summation of two chirps. The chirps are windowed by a gaussian. The chirp duration is 200µs.

**CONCLUSION**

In this paper we proposed a new formulation for broadband MIMO sonar systems by separating clearly the terms of propagation and the terms of target reflection. This formulation is more flexible for different target models integration. A new model for cylindrical shell has been proposed using virtual scatterers. A new method of pseudo time reversal has been proposed in order to increase the SNR by a factor of K (where K is the number of transmitters) and improve the detection performance of the system. Future works include a demonstration in tank of the Pseudo Time Reversal.

**REFERENCES**


Structured Session 24

Basin-scale Acoustics and Seismic Monitoring

Organizers: Mark Prior & Andrew Forbes
Abstract: Since vertical normal modes in shallow water correspond to ray paths with multiple surface and bottom reflections, a tilt in the seabed must cause horizontal curving of these rays and modes as well as possibly altering the vertical angle. Generally bending is towards deep water, and there is the possibility of more than one horizontal “eigenray” connecting source and receiver, particularly with separations parallel with the shore line. Each mode has its own horizontal shadow zone which is a 3D version of mode cut-off. Theory has been known for at least 30 years, but there is renewed interest since the appearance of recent experimental evidence of horizontal angle shifts. Because it is important to be able to estimate the magnitude of the three-dimensional effects, some rules-of-thumb are presented for calculating horizontal angle shifts, depth/frequency limits, and so on in relation to the seabed’s critical angle.

Keywords: 3D propagation, horizontal refraction, modal shadows
1. INTRODUCTION

Over the years there has been plenty of interest in numerical calculation of propagation in complex environments including the wedge-shaped ocean [1,2,3]. Three-dimensional propagation with variable bathymetry introduces the possibility of horizontal bending, focusing, and interference in isovelocity water purely through the small changes in ray heading after each bottom reflection. In some seminal work using an energy flux approach Weston [4] introduced the useful concept of a ray invariant. This determines a ray’s vertical angle from the local water depth regardless of the earlier history of that ray. In a similar way one can determine changes in the ray’s horizontal heading [5]. These ideas were subsequently used by Harrison [6] to determine the ray curvature and to map out their trajectories in the horizontal plane for a number of geometries including troughs, ridges, the wedge, circular basins and seamounts. By regarding an acoustic normal mode as a family of rays with the same initial elevation angle but a uniform spread of azimuths it was possible to see that in a wedge-shaped ocean, instead of a mode cut-off, each mode has a hyperbolic shadow between the source and the shoreline [7]. In the unshadowed region there is the possibility of receiving the same mode from a single source by two separate routes and this can lead to spatial interference and focusing. In fact this behaviour is an analogue of the familiar refraction in the vertical plane with a stratified medium [8,3]. The varying depth of the ocean acts like a varying sound speed in the horizontal plane.

Designing an experiment to measure these phenomena is difficult because the angle changes are typically small and three-dimensional. Looking for modal shadows and interference effects is probably a more promising approach, and Heaney has recently published some compelling evidence [9].

By using the approach of [6,7] it is possible to set up some rules of thumb which enable one to estimate the importance and magnitude of these effects in any given circumstances. Section 2 describes the foundation of this approach and Section 3 lists the rules of thumb explicitly.

2. FOUNDATIONS

In the incoherent adiabatic mode sum
\[ |\psi|^2 = 2\pi \sum_n \frac{\phi_n^2(0,z_0)\phi_n^2(r,z) e^{-\int_{z_0}^z \log|q_n'| dr}}{K_n r} \]  

as depth changes, the mode at some location \((x,y)\) stretches to fit the water depth. This is expressed, for a WKB mode, by the “phase integral” [10]

\[ n = \int_0^{r(r)} \sqrt{k^2(z(r)) - K_n^2} \, dz / \pi \]  

This is the basis of Weston’s ray invariant [4] for the rays angle in the vertical plane

\[ \int \sin \theta \, dz = \text{const} \]
where the integral is taken over the water depth (or at least the part of the duct for which $\theta$ is real). In principle, both $\theta$ and $c$ are depth dependent in a refracting environment. For isovelocity this reduces to

$$H \sin \theta = \text{const} = H_o \sin \theta_o$$  \hspace{1cm} (4)$$

This says, the shallower the water, the steeper the ray. As long as the ray is less steep than the critical angle (i.e. the mode doesn’t cut off) the process is reversible (i.e. adiabatic). For certain geometries there is also a relationship between the ray’s heading or azimuth $\phi$ relative to the direction of the depth contours and the vertical angle $\theta$, (note that [6,7] used the complement of $\phi$). For Cartesian geometry where the depth contours are straight and parallel, the relation is [5, 6]

$$\cos \theta \cos \phi = \text{const} = \cos \theta_o \cos \phi_o$$  \hspace{1cm} (5)$$

For polar symmetry where depth contours are concentric circles, the relation also involves the distance from the polar centre [6]

$$r \cos \theta \cos \phi = \text{const} = r_o \cos \theta_o \cos \phi_o$$  \hspace{1cm} (6)$$

It was shown in [8] that Eqs. 4-6 can also be understood in terms of horizontal refraction of a vertical mode. Equation (2) shows that the horizontal wavenumber for one particular mode must change according to water depth

$$K_n^2 = k^2 - \frac{(n\pi)^2}{H^2}$$  \hspace{1cm} (7)$$

If the depth contours are parallel then the horizontal wavenumber in the direction of the contours $K_c$ must remain constant. Thus the mode’s heading relative to the contour is given by

$$\cos \phi = \frac{K_c}{K_n} = \frac{\text{const}}{\cos \theta}$$  \hspace{1cm} (8)$$

and Eq. (5) follows. Equation (6) can be derived in a similar way, and with arbitrary bathymetry Eq. (7) defines a local horizontal wavenumber, analogous to sound speed, that defines the mode trajectory through the eikonal equation.

The ray invariants [Eqs. 4,5) or Eqs. (4,6)] can be used to calculate the horizontal trajectories of the zigzag rays or modes, and a number of examples are given in [6]. In a wedge, for example, the rays follow the hyperbola

$$a y^2 = (b x - c)^2 + d^2$$  \hspace{1cm} (9)$$

where the constants are set by the initial values
The equivalent of mode cut-off in 3D can be seen by finding the envelope of a horizontal “fountain” of rays (Fig. 1) all starting with the same elevation angle $\theta_o$ (for that mode) but a uniform spread of azimuths.

![Diagram of fountain of rays](image)

Fig. 1: A “fountain” of rays emanating from a point source in plan view all starting with the same elevation angle but with various headings. Their envelope (grey), also a hyperbola, forms the boundary of the equivalent mode’s shadow. Shoreline is at the top.

This forms a well defined shadow for each mode in the shallow water direction, which is also hyperbolic [7] following

$$y^2 = x^2 \tan^2 \theta_o + y_o^2 \sin^2 \theta_o$$  \hspace{1cm} (14)

Thus, as a receiver approaches the shoreline, or even passes parallel to the shoreline, the modes die out one by one until there are none left as in Fig. 2.

Harrison [7] shows the equivalent shadow behind a conical seamount. Although it is difficult to calculate the envelope analytically it is easy to see it by plotting the fountain of rays (Eq. (25) from [6]) as in Fig 3.

Using the same ray invariants one can estimate many other quantities such as number of bottom reflections, ray cycle distance, total bottom loss (on assumption of some dependence of loss on angle) [11], reverberation [12,13], and noise [14]. In the context of modal shadows there are some much simpler deductions that one can make by assuming that losses are zero until the ray angle reaches the critical angle, after which the ray is terminated. By Eq. (4), imposition of a critical angle is the same as imposition of a critical depth $H_c$, and

$$H_c = H_o \sin \theta_o / \sin \theta_c$$  \hspace{1cm} (15)
Thus the effect of a critical angle on, say Fig. 1, is to truncate all rays attempting to cross the contour at depth $H_c$. So all rays pointing more or less directly upslope get truncated, but, on the assumption that $H_o > H_c$, there is still a strong possibility that some rays will travel slightly upslope before bending towards deep water. These can exist simultaneously with the rays that start off with a downslope component, and so it is possible to find two arrivals from the same mode (and consequently interference) in the intermediate water depths along the coast from the source. From Eq. (5) it can be seen that the asymptotic angle $\phi_o$ of a ray and the mode shadow is the same as the steepest vertical angle at the apex of the hyperbola.
namely the critical angle. In other words $\phi_a = \theta_c$, and this puts a tight restriction on the zone where these effects can be seen. In addition, of course, the seabed needs to be low loss, tilted, and smooth. Nevertheless Heaney [9] has recently found compelling evidence for these phenomena.

3. EXPLICIT RULES OF THUMB

The following rules, based on the above summary of the simple theory, assume a slowly varying smooth seabed.

1. Vertical angle is related to local water depth and initial angle (through the ray invariant or since the mode is assumed adiabatic)

2. Local ray heading (azimuth) is related to vertical angle (given initial heading)

3. Rays with many headings but the same elevation angle (a particular mode) bend to different extents on a slope making a ‘fountain’ shape in plan view. Each of the ray trajectories for a wedge shaped duct is a hyperbola.

4. The envelope of this fountain forms a caustic in the horizontal plane for this mode. On the outer edge is a shadow. For a wedge this shape is also a hyperbola.

5. Each mode has its own shadow and caustic at its boundary which is a three-dimensional version of mode cut-off.

6. The asymptotic angle between this “near-hyperbola” and the depth contours (or shore line) is the same as the steepest ray elevation angle which occurs at the apex of the hyperbola.

7. If the seabed is lossy with a critical angle then by (1) this angle defines a critical depth beyond which neither modes nor rays can pass.

8. The “fountain” of rays (3) is therefore truncated at the depth contour corresponding to this depth.

9. With a lossy seabed the only rays that avoid truncation are those that either set out in the downslope direction or those that go only slightly upslope and just graze the critical depth contour line. Therefore the main areas where a single vertical mode can arrive by two horizontal paths is slightly downslope of the shadow edge. In principle this can be well away from the shadow edge even with a critical angle.

10. Similar considerations in polar coordinates, i.e. where depth contours are circular, lead to shadow zones that wrap around seamounts and blank out a fixed angle sector behind them (equivalent to the hyperbola’s asymptotes).

Corollaries

11. If the low loss boundaries lead one to consider normal modes to be important then they will still be important if the seabed has a component of tilt across the propagation path.
12. Because of (6) and typical seabed reflection properties the amount of horizontal bend must always be pretty small.

13. Because low bottom reflection losses are required for modal propagation the modal shadowing and caustic effects are most noticeable at low frequencies (wavelength comparable with the water depth).

14. It is probably not worth considering these 3D effects in the context of reverberation because from (13) the wavelength needs to be big compared with the water depth. It is therefore extremely large compared with the seabed’s roughness.

4. CONCLUSIONS

Acoustic paths with multiple bottom reflections bend in the horizontal plane towards deep water. This can be regarded as a refraction phenomenon from a modal point of view, and the three-dimensional version of mode cut-off is a horizontal shadow zone for each mode. By using the concept of a ray invariant (Eqs. (3-6)) it is possible to map out these shadow zones for a given bathymetry and to determine where there is the possibility of two arrivals from the same mode. Section 2 briefly laid out the background theory, and Section 3 provided some explicit rules of thumb.

REFERENCES


Acoustic propagation through the fluctuating and inhomogeneous ocean: Basin and global scale issues

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Abstract: The ocean is an “acoustical” medium in which its inhabitants have evolved brilliant acoustic tools for environmental sensing. Human efforts have been rather successful as well, though acoustical fluctuations from the multi-scale ocean sound speed environment have bedeviled many efforts. This talk will give a review of our present understanding of acoustic scattering from small scale processes like internal waves, from basin to planetary scales. Emphasis will be made on effects which influence large scale remote sensing, and future global scale experiments.

Keywords: Global acoustics, Acoustic fluctuations, Internal waves and eddies
1. INTRODUCTION

Interest in basin and global scale acoustics has been re-kindled recently with the suggestion by Dushaw[1] that a modern day repeat of the 1960 Perth-to-Bermuda transmissions[2] could reveal a global climate change induced travel time shift of order 10 sec. The Dushaw analysis also raises important questions concerning ocean acoustic propagation around continental barriers (South Africa and Eastern Brazil), and the cumulative effects of mode coupling and bottom loss. In that context, this paper will present new insights into global propagation based on recent advances in our understanding of long range acoustic scattering by internal waves and eddies. The organization of this paper is as follows. Section 2 discusses mode coupling and bottom interaction for low frequency basin scale propagation, while section 3 examines internal wave and eddy induced scattering around barriers. Section 4 provides a brief summary. It should be understood that this short paper only presents a small proportion of the material covering in the meeting presentation.

2. MODE COUPLING AND BOTTOM INTERACTION

Basin and global scale acoustic propagation presents many challenges to the prediction of acoustic fluctuations caused by small scale ocean sound speed structure. First acoustic variability can be quite strong thus precluding perturbation techniques[3], and at ranges of order hundreds of kilometers, instability of the unperturbed ray due to ray chaos invalidates many theories (such as path integrals [4]) that expand about the deterministic ray path[5]. Recently however it has been shown that coupled mode approaches in both shallow[6] and deep water environments[7,8] can be quite accurate out to very long range for second moments like mean intensity given by

\[
\langle I(r,z) \rangle = \sum_{n=1}^{N} \sum_{p=1}^{N} \frac{\langle a_n^* a_p^*(r) \rangle \phi_n(z) \phi_p(z)}{\sqrt{k_n k_p}} \tag{1}
\]

Here \( \langle a_n a_p^* \rangle \) is the cross mode coherence, and \( k_n \) and \( \phi_n(z) \) are the \( n^{th} \) eigen-wavenumber and vertical function. An evolution equation for the cross mode coherence is[6,8].

\[
\frac{d}{dr} \langle a_n^* a_p^* \rangle = i(l_n - l_p^*) \langle a_n a_p^* \rangle - \sum_{m=1}^{N} \sum_{q=1}^{N} \langle a_q a_m^* \rangle I_{mn,ap} - \langle a_n a_m^* \rangle I_{mn,ap}^* - \langle a_q a_n^* \rangle I_{mp,qm} - \langle a_q a_m^* \rangle I_{mp,qm}^* \tag{2}
\]

where \( l_n = k_n + i\alpha_n \) is the complex eigen-wavenumber and the scattering matrices \( I_{mn,ap} \) for the Garrett-Munk internal wave spectrum[9] are described in reference[8]. The evolution equation was derived using the small angle multiple forward scattering and Markov approximations. In this approach an important resonance condition is revealed such that mode coupling between modes \( n \) and \( m \) is determined only by the internal waves which have horizontal wavenumbers equal to the modal beat wavenumber \( k_{mn} = k_m - k_n \). This equation is easily solved on the computer because the matrices \( I_{mn,ap} \) are independent of range, and have an analytical form consistent with the GM spectrum[8].
2.1 Mode Interactions

This section discusses the expected rates of mode coupling by internal waves and bottom interaction for long range, low frequency acoustic propagation in the 10 to 70 Hz region. As discussed in the previous section, the cross mode coherence evolution equation has constant coefficients and thus the solution will be in terms of various exponentials. The exponential rates then are of keen interest and to an excellent approximation the mode interaction rates can be computed by finding the eigenvalues of coupling matrices discussed by Dozier-Tappert[10,8]. In the absence of attenuation these interaction rates dictate the approach to modal energy equipartition, and thus have an important impact on the acoustical statistical moments, and the approach to saturation. Modifying to account for attenuation the matrix which gives one the decay rates from both coupling and attenuation[6] is

\[ F_{mn} = 2 \text{Re}[I]_{mn}, m \neq n \rightarrow F_{nn} = -2\alpha_n - \sum_{n=1, n \neq m}^{N} F_{n,m} \]  

(3)

where we call the eigenvalues of this matrix \( \lambda_c \). Because the attenuation is weak in the case of interest here (weakly bottom interacting, low frequency deep water propagation) the smallest eigenvalues will be controlled by attenuation, while the larger ones represent coupling rates. In order to get an idea of the distribution of coupling rates we examine the mean eigenvalue, \( \bar{\lambda}_c \), and the minimum \( \lambda_{cm} = \min(\lambda_c) \). Figure 1 shows the equivalent decay ranges \( R_c = 1/\bar{\lambda}_c \) and \( R_{cm} = 1/\lambda_{cm} \) as a function of frequency.

![Figure 1: Estimates of average and minimum decay ranges for a canonical ocean with Garrett-Munk internal wave sound speed perturbations.](image)

In this calculation \( N \) has been chosen so that at each frequency the mode turning points are between the surface and 3700-m (the water depth is 4000-m); This choice of \( N \) gives primarily non-bottom interacting modes. Physically \( R_c \) tells us the average range at which the modes have developed significant coupling, while \( R_{cm} \) gives us the modal energy equipartion range (in the non-attenuating case)[10].

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Figure 1 shows that for low frequency sound transmission through internal waves, 70 Hz sound will have interacting modes rather quickly by a few hundreds of kilometers range. On the other hand at the very low frequencies, interaction will not begin until several megameters. Mode energy equipartition is estimated to occur near 6 Mm for 70 Hz sound while the equipartition range at 10 Hz is tens of Mm. These results are consistent with observations made in the North Pacific Ocean, where energy equipartition for the first 10 modes is reported for 75 Hz sound at 5 Mm range[11]. The implication for basin or global scale transmissions is that frequencies of order 20 Hz would be most desirable.

2.2 Cross Mode Coherence

This section addresses the range decay of cross mode coherence, an effect that has a profound influence on the mean intensity and other second moments. The decay of cross mode coherence is controlled by phase randomization effects caused both by coupling (as previously discussed) and by adiabatic effects. It can be easily shown that the adiabatic cross mode coherence relation is

\[\langle a_n a_m^* \rangle(r) = \langle a_n a_m^* \rangle(0) e^{-\lambda_a(n,m)r}, \rightarrow \lambda_a(n,m) = I_{mm,mm} + I_{mm,nn} - 2I_{mm,nn} \] (4)

where \(\lambda_a(n,m)\) is the adiabatic phase structure function[8], or physically the adiabatic cross mode coherence decay rate. Thus again to quantify the distributions we define the mean decay rate over all the mode combinations (m,n) as \(\overline{\lambda_a}\) and the minimum \(\lambda_{am} = \min(\lambda_a)\). Figure 1 shows the mean and maximum adiabatic decay ranges \(R_2 = 1/\overline{\lambda_a}\), and \(R_{am} = 1/\lambda_{am}\). As previously stated mode coupling also results in relative mode phase randomization. It turns out that this cross mode de-coherence rate is of order \(c\lambda\) (in the two mode case the proportionality factor is 1/2). So, Fig. 1 shows the two relevant mode de-coherence ranges, and the coupling and adiabatic contributions are seen to be relatively close. Importantly these calculations suggest that at very low frequencies cross mode coherence will be maintained out to tens of megameters, while at higher frequencies the coherence will be gone after only a few Mm. Again, the loss of coherence at a few Mm is consistent with 75 Hz North Pacific observations in which no coherence is seen for the first 10 modes at 5Mm range[11].

Finally, the frequency scaling of this problem is of fundamental interest. We find that the average decay ranges in Fig. 1 scale roughly as frequency to the minus 2 power and thus signal stability increases rapidly with decreasing frequency.

The effects of coupling and mode decoherence on mean intensity are discussed next.

2.3 Mean Intensity

The decay ranges discussed in the previous section give one a good idea of the behavior of the mode statistics without considering the specific initial conditions, however an example initial condition is useful to gain some insight concerning the acoustical field details. For a source on the sound channel axis, Fig. 2 shows 15 and 65 Hz mean intensity out to 20 Mm range for a receiver also at the sound channel axis. For the unperturbed calculation the expected complex multi-mode interference pattern is seen, and this pattern is superimposed on a slow decay caused by the water column and bottom attenuation. In the adiabatic cases
the phase randomization of the modes, causes less constructive and destructive interference, and so the mean intensity pattern is smoother than the unperturbed case. In the fully coupled calculation the modes are exchanging energy, and the cross mode coherence is decaying by both coupling and adiabatic effects. A particularly striking feature of the 65 Hz full theory case is the absence of any interference effects past a few Mm; this effect is due to the rapid mode decorrelation. In addition in both the 15 and 65 Hz cases the fully coupled calculation is always below the adiabatic one; this feature occurs because coupling re-distributes the acoustical energy away from low modes, to higher modes. These results suggest that future basin or global scale transmission may be best carried out at frequencies of order tens of Hz, where mode coupling and randomization are not so strong.

Figure 2: Estimates of 15 and 65 Hz range evolution of unperturbed, adiabatic, and coupled mode mean intensity at 1000-m depth for a canonical ocean with Garrett Munk internal waves. The source depth is 1000-m. For the 15/65 Hz calculation a total of N=12/52 modes were used.

Now the discussion here has centered on effects at a single frequency. When broadband signals are considered we know that the resulting wavefront has been observed to be quite stable out to multi-megameter ranges\[12\]. This result implies that cross frequency coherence can in-fact be quite high. Theory has been developed to estimate the cross frequency coherence but this topic is beyond the scope of the present paper.

3. CONTINENTAL SHADOWING AND SCATTERING

The analysis of Dushaw\[1\] suggests that there is no direct acoustic path from Perth to Bermuda, and that some scattering process must be at play to account for the Bermuda ensonification. Important new results from ocean internal wave scattering studies are thus relevant here. Application of ray and parabolic equation methods to sound transmission through internal waves has revealed that acoustic energy primarily scatters along the pattern of the unperturbed wavefront as opposed to across it [13,14].
Figure 3: Direct numerical simulation of cross range ray scattering through Garrett-Munk internal waves. The three panels show scattered ray distributions (points) at nominal ranges of 340, 680, and 1000 km range. A total of 3000 rays are simulated using a zero initial angle. The spread in cross range is roughly circular (solid line).

Figure 3 shows a numerical simulation of ray propagation through internal-wave-induced sound-speed perturbations in the horizontal plane. Each ray has a zero initial angle but rapidly spreads out into the transverse direction. Recent work provides an analytic description of this scattering[14] in terms of the spectrum of internal waves and ray path geometry. An order of magnitude calculation shows that the rms transverse deviation of ray paths is

$$Y_{\text{rms}} = R^{3/2} \left( \frac{\langle \mu^2 \rangle}{L_H} \right)^{1/2}$$

where $\langle \mu^2 \rangle$ is the variance of fractional sound speed, $L_H$ is the internal wave horizontal correlation length, and $R$ is range. Importantly the transverse scattering of acoustic energy is seen to grow rapidly with range, that is as range to the three-halves power. We can examine the effect of this cross range scatter to the problem of Perth to Bermuda propagation. In particular we would like to know the horizontal spread of rays off the coast of South Africa. Taking $R=10$ Mm (roughly the distance from Perth to S. Africa), $\langle \mu^2 \rangle^{1/2}=0.001$, and $L_H=10$ km, we find $Y_{\text{rms}} \sim 300$ km (an effective horizontal angle deviation of 1.7 deg.). At the final range these estimates increase by roughly a factor of 3 giving $Y_{\text{rms}} \sim 900$ km, and an rms horizontal angle of roughly 5 deg. For ocean eddies we can take $\langle \mu^2 \rangle^{1/2}=0.003$, and $L_H=50$ km.

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km, we find $Y_{rms} \sim 400$ km at R=10 Mm; a significant effect. This spreading can be compared to diffractive effects, quantified by the Fresnel zone. South Africa is roughly half way between Perth and Bermuda, so the horizontal Fresnel Zone near South Africa is roughly $R_f = \left( \frac{\lambda R_{pb}}{2} \right)^{1/2}$, where $\lambda$ is the acoustic wavelength, $R_{pb}$ is the range between Perth and Bermuda (about 20Mm). For $\lambda=50$ m, we get $R_f \sim 15$ km, much smaller that the transverse scattering. Thus transverse scattering by internal waves may account for passage of acoustic energy around African continent.

4. SUMMARY

A modern repeat of the Perth to Bermuda transmissions to measure anthropogenic ocean warming would require an electronic source with adequate bandwidth, low center frequency, and high power. The HLF-6A source used in the Alternate Source Test (AST)[15] is a reasonable candidate, with a center frequency of 28 Hz, and a bandwidth (3dB) of 10 Hz. This source has a transmission level of 195 dB re 1 $\mu$PA at 1 m (260 Watts), which may be marginally adequate for these very long ranges. Vertical arrays for mode forming as well as horizontal towed arrays for bearing angle estimation would be very important for resolving the acoustic path taken by the sound and to quantifying signal fluctuations. In particular, towed and mode resolving moored vertical arrays deployed in the vicinity of the South African ocean would be quite useful for resolving the acoustic path issue and shadowing by the continent.

REFERENCES


THE EFFECT OF HORIZONTAL REFRACTION ON BACK-AZIMUTH ESTIMATION FROM THE CTBT HYDROACOUSTIC STATIONS IN THE INDIAN OCEAN

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Abstract: Both large-scale spatial variations of oceanographic characteristics and changes in the bottom topography can induce horizontal refraction. This study uses numerical modelling to investigate the horizontal refraction of low-frequency underwater sound propagating in the Indian and Southern Oceans, and the resulting effects on estimates of back-azimuths from the Comprehensive Test-ban Treaty (CTBT) hydroacoustic stations. It is shown that the horizontal wavenumbers of different modes in the deep water regions of the Southern Ocean within and beyond the Antarctic Convergence Zone (ACZ) have stronger horizontal gradients than those in the Indian Ocean. The deviation of bearing to the signal source from the two CTBT hydroacoustic stations off Cape Leeuwin (Western Australia) and off the Chagos Archipelago (BIOT) resulting from horizontal refraction does not exceed 0.20° in the most parts of the Indian Ocean. By contrast, strong horizontal wavenumber gradients in the high-latitude regions of the Southern Ocean and, especially, across the ACZ, cause the acoustic propagation path to deviate by as much as one degree from the geodetic line. This deviation depends strongly on the azimuth and range from the two CTBT hydroacoustic stations to signal sources located in the Southern Ocean beyond the ACZ.

Keywords: Horizontal refraction, Back-azimuth estimation, CTBT hydroacoustic station
1. INTRODUCTION

Location of hydroacoustic events using back-azimuth of signal arrivals at the CTBT underwater listening stations is currently carried out ignoring possible effects of horizontal refraction due to large-scale spatial variations of the ocean environment [1]. In order to examine all possible errors of locating remote sources of underwater noise from the CTBT stations, it is necessary to investigate the effect of horizontal refraction on the estimates of bearing to noise sources from the CTBT receive triads. Mesoscale oceanographic features, such as eddies and internal waves can cause noticeable refraction at high and moderate frequencies [2-3]. At low frequencies, large-scale spatial variations of the sea depth and oceanographic characteristics can accentuate horizontal refraction for long-range acoustic propagation [4-6]. In this study, we followed the computational procedure proposed for the analysis of the Perth-Bermuda propagation experiment results [5]. It involves a combination of an adiabatic mode theory in the vertical dimension and a ray theory in the horizontal dimension and, therefore, takes into account horizontal refraction of individual modes due to both transverse sound speed gradients and bottom interaction over the continental slopes and sea mounts. The ray model was constructed on the surface of the Earth represented by an ellipsoid of rotation and expressed in terms of the latitude $\phi$, longitude $\lambda$ and azimuth angle $\alpha$ measured clockwise from the north. The ray equations on the Earth ellipsoid are [5]:

\[
\begin{align*}
\dot{\phi} &= \cos \alpha / \mu(\phi) \\
\dot{\lambda} &= \sin \alpha / \nu(\phi) \cos \phi \\
\dot{\alpha} &= \frac{\sin \alpha}{\nu(\phi)} \tan \phi - \left( \frac{\sin \alpha}{\mu(\phi)} \frac{\partial}{\partial \phi} + \frac{\cos \alpha}{\nu(\phi) \cos \phi} \frac{\partial}{\partial \lambda} \right) \log k_n
\end{align*}
\]  

(1a)  
(1b)  
(1c)

where $k_n$ is the horizontal wavenumber of mode $n$. Overdot signifies the derivative with respect to arc length $s$. The variables $\mu$ and $\nu$ are:

\[
\begin{align*}
\mu(\phi) &= r_{eq} (1 - \varepsilon^2)(1 - \varepsilon^2 \sin^2 \phi)^{3/2} \\
\nu(\phi) &= r_{eq} (1 - \varepsilon^2 \sin^2 \phi)^{1/2}
\end{align*}
\]  

(2)

and $r_{eq}$ and $\varepsilon$ are the equatorial radius and eccentricity of the Earth respectively. The last term in Eq. (1c) accounts for distortion of the ray paths due to the transverse gradients of the horizontal wavenumber $k_n$ based on the Snell's law. If this term is neglected, the solutions of Eq. (2) are geodesics on the ellipsoid [7].

The modal horizontal wavenumbers were calculated using the KRAKEN program [8] on a horizontal grid with 0.1-degree grid size. The sound speed profiles were derived from the climatology salinity and temperature data gridded with 1-degree resolution in the World Ocean Atlas 2005 (http://www.nodc.noaa.gov/OC5/WOA05/pr_woa05.html). The bathymetry data were taken from the ETOPO2 Global 2-Minute Gridded Elevation Data (http://www.ngdc.noaa.gov/mgg/fliers/01mgg04.html). Both sound speed profiles and bathymetry were interpolated to 0.1-degree resolution to calculate modal wavenumbers. The system of ordinary differential equations in Eq. (1) can be solved using a 4-th or 5-th order Runge-Kutta method [9]. In the integration process, the maximum integration increment in
distance was limited by the grid size and the modal wavenumbers were interpolated within the current grid cell in order to reduce errors of numerical integration.

2 SPATIAL VARIATIONS OF MODAL WAVENUMBERS

![Figure 1](image)

*Figure 1. The map of mode 1 horizontal wavenumber at 20 Hz in the Indian and Southern Ocean region. Climatology data are taken for the winter season. The horizontal resolution is 0.1° for both latitude and longitude.*

Fig. 1 shows the variation of mode 1 horizontal wavenumber at 20 Hz over the region of the Indian and Southern Oceans in the winter season with 0.1 degree resolution. The spatial variation of horizontal wavenumbers is similar in general for different modes and at different frequencies. The Antarctic Convergence Zone (ACZ) is a transition area which divides the whole ocean region into the northern part with relatively low horizontal wavenumber values and the southern part with higher horizontal wavenumbers. In Southern Ocean region beyond the ACZ, the wavenumber increases with latitude. In the temperate ocean region above the ACZ, the modal wavenumber is relatively higher in the region south and southwest of Australia compared to the other regions. In the tropical ocean region, the modal wavenumber slightly decreases in the direction towards the north.

3 POTENTIAL EFFECT OF HORIZONTAL REFRACTION ON BEARING ESTIMATION

As follows from Eq. (1) the rate of azimuth deviation away from the local geodesic, governed by the last term of Eq. (1c), is proportional to logarithm of the transverse gradient of an equivalent refraction index $N_a = k_a / k_{a0}$, where subscript zero refers to an arbitrary reference value. In the deep water regions, the normal modes propagated over large distances are trapped either in the SOFAR channel in the temperate ocean or in the near surface channel in the high-latitude areas of the Southern Ocean and, therefore, the modal wavenumbers are dependent primarily on the sound speed profile rather than bathymetry. To examine the influence of wavenumber gradients, we selected two geodesic transects revealing the strongest wavenumber gradients in the two distinctive ocean environments: the deep water
regions of the Indian Ocean and the Southern Ocean. The aim is to investigate possible ray distortion, which is assessed with respect to the rate of azimuth deviation from the geodetic line represented by the last term of Eq. (1c), for the rays which cross the transect in the perpendicular direction, and to analyze the potential effect of horizontal refraction on bearing estimation in these two regions. The first transect crosses the Indian Ocean from 45°S 90°E to 15°N 60°E and the second one crosses the Southern Ocean from 45°S 90°E to 68°S 75°E.

Figure 2. The azimuth deviation rate of rays from the geodetic line due to the wavenumber gradient along the transect in the Indian Ocean (from 45°S 90°E to 15°N 60°E). Left panel shows the azimuth deviation rate for modes 1-3 at 20 Hz and the right panel shows the deviation rate for mode 1 at different frequencies.

The rate of azimuth deviation from the geodetic line for ray trajectories of different modes at different frequencies along the transect in the Indian Ocean is shown in Fig 2. If considering only the effect of changes in the sound speed, the strongest azimuth deviation rate takes place in the beginning of the transect, where the sharpness of the sound speed minimum around the SOFAR channel axis decreases rapidly along the transect. For the rest of the region, the deviation rate is nearly zero. It is also seen in Fig. 2 that the ray deviation rate along this transect is almost independent of mode number and frequency for the modes trapped in the water column. For modes interacting with the seabed, e.g. mode 3 at 20 Hz and mode 1 at 5 Hz, the ray deviation rate varies significantly over the region where the interaction occurs.

As shown in Fig. 3, the azimuth deviation rate along the transect in the Southern Ocean slightly decreases with mode number and increases with frequency. Such dependence takes place because higher order modes at lower frequencies penetrate deeper in the water column and hence are less sensitive to rapid change in the sound speed in the upper water layer across the ACZ. It can be noticed that the ray deviation rate across the ACZ and over the Antarctic continental slope is much higher than that along the transect in the Indian ocean, with a factor of around four for mode 1 at 20 Hz.
4 BEARING ERRORS DUE TO HORIZONTAL REFRACTION

Figure 3. The azimuth deviation rate of rays from the geodetic line due to the horizontal wavenumber gradient along the transect in the southern Ocean (from 45° S 90° E to 68° S 75° E). Left panel shows the deviation rate of the first three modes at 20 Hz and the right panel shows the same for mode 1 at different frequencies.

Figure 4. The map of bearing deviation from the true azimuth observed from the HA01 (left panel) and H08S (right panel) stations for noise sources located in the Indian and Southern Oceans. The errors are due to horizontal refraction calculated for the spatial variations of mode 1 wavenumber at 20 Hz shown in Fig. 1.

Figure 4 shows the effect of horizontal refraction on bearing deviation from the true azimuth to sources of underwater noise observed from the HA01 and H08S stations, shown as a map of source location in the Indian and Southern Oceans. The deviation is calculated for the wavenumber of mode 1 at 20 Hz in winter shown in Fig.1. For the entire region, the bearing error due to refraction reveals strong azimuth and range dependence. The bearing deviation for HA01 does not exceed 0.2° for most parts of Indian Ocean region north of ACZ, except the shallow water regions over the continental slope and underwater mounts. When the source of noise is located beyond the ACZ in the Southern Ocean, relatively strong wavenumber gradient across the ACZ causes noticeable deviation of bearing to the true source location. The bearing errors due to refraction are less than 0.2° for the source location observed from HA01 at the azimuth of about 193° along which the propagation path is almost perpendicular to the ACZ frontal zone and therefore is negligibly affected by horizontal
refraction. The absolute value of bearing errors due to refraction generally increases with azimuth moving away from this direction in both sides and reaches the maximum value of nearly $1^\circ$ in the westernmost and easternmost parts of the southern Ocean observed from HA01. For the H08S station, the propagation path to noise sources least distorted by horizontal refraction is along the azimuth of around $185^\circ$. The variation of bearing errors to both sides from this azimuth is generally similar to that for HA01. The bearing errors for the northeast part of the Indian Ocean are considerable due to nearly perpendicular angles between the propagation path to H08S and gradients of the modal wavenumbers in this area.

4 CONCLUSIONS

In this paper, we investigated the effect of horizontal refraction on estimation of bearing to low-frequency sources of underwater noise observed in the Indian and Southern Oceans from the two CTBT hydroacoustic stations. The gradient of modal wavenumbers along two transects in the deep water regions of the Indian and Southern Ocean was used to examine the effect of horizontal refraction on azimuth deviation of the propagation path for different modes and frequencies. The study revealed that the effect of horizontal refraction can cause nearly $\pm1^\circ$ errors of the back-azimuth estimation for sources of noise located in the Indian and Southern Oceans from both HA01 and HA08S CTBT hydroacoustic stations. The effect of horizontal refraction is highly dependent on the azimuth and range to the noise source.

REFERENCES

improving the accuracy of in-water travel-time predictions for seismic event location using two-dimensional underwater acoustic propagation modelling

mark k prior

abstract: the international monitoring system (ims) of the comprehensive nuclear-test-ban treaty organization includes seismic, hydroacoustic and infrasound sensors that record signals travelling through the solid earth, ocean and atmosphere. signals may be generated by events including earthquakes, volcanic eruptions and explosions. the times at which signals from the same event arrive at geographically separated sensors can be used to locate the event, provided that travel time from the sensor to all possible event positions is known. the time taken for sound to travel through the ocean from its generating event to the receiver can be approximated as being simply proportional to distance, using an average value for the sound-speed of seawater. however, this approach ignores variations of sound speed with position and time and this can result in location errors for events whose position is determined by the arrival times of waterborne signals. these errors can be reduced if databases containing seawater sound-speed as a function of time, latitude, longitude and depth are used along with a numerical propagation model. this database/model combination allows travel time to be predicted from all candidate locations to receivers of interest. this paper investigates the benefits that can be achieved by the use of a database/model approach, relative to the approach of a single, average sound speed. improvements are quantified via time residuals – differences in measured and predicted arrival times – for signals travelling by waterborne paths from well-located events to ims hydroacoustic stations. improvements in accuracy are investigated for travel-time predictions with temporal resolutions of one, three, four, six and twelve months. it is found that residuals can be reduced by up to a factor of two and that this reduction is achieved even if only one travel time table is used to cover all months.

keywords: seismic monitoring, travel time, underwater
1. INTRODUCTION

The preparatory commission for the comprehensive nuclear-test-ban treaty organisation (CTBTO) operates the International Monitoring System (IMS), a global network of sensors which includes hydroacoustic stations designed to detect signals propagating through the ocean [1]. The hydroacoustic network uses two types of station – hydrophone and T-stations. Hydrophone stations use hydrophone triads deployed in the ocean deep-sound-channel while T-stations use seismometers located on land, near to the coast. Hydrophone triads are deployed in a two-kilometre-side, triangular configuration in the horizontal plane. This allows the arrival azimuth of signals to be determined from measurements of the time differences between signal arrivals at the three hydrophones. T-stations cannot determine arrival azimuth and are restricted to measurement of arrival time.

Time series data measured on hydroacoustic stations are transmitted by satellite link and received at the Commission’s International Data Centre (IDC) in Vienna where they are processed automatically. The first stage of processing identifies discrete arrivals in terms of their arrival time and – for hydrophone stations – azimuth. This information is later used in a global association (GA) program which identifies the events that gave rise to the signals. Examples of typical events are earthquakes, volcanic eruptions and anthropogenic explosions. During GA, hydroacoustic, seismic and infrasound arrivals are considered together.

To convert measurements of signal arrival time and azimuth into estimates of event time and location, GA requires information about the travel time between IMS stations and all possible event locations. For water-borne paths, the simplest approach would be to use a single, average, seawater sound speed to relate travel time directly to the event-station distance. This simple approach neglects the influence of spatial and temporal variations in sound speed which can have important effects. Any inaccuracy in the estimation of travel time will result in a difference between the observed and predicted arrival time of signals from an event. This time difference is known in seismological terminology as a “time residual”.

In an attempt to reduce time residuals, it is possible to predict travel time using a combination of a numerical acoustic propagation model and databases of seawater sound-speed, ocean depth and seabed composition. Look-up tables of travel time as a function of putative-source range and azimuth from the station can be incorporated in GA calculations to improve signal-event association and reduce time residuals. The seasonal variation of seawater temperature gives rise to changes in sound speed over the course of a year and different travel-time tables might have to be used for different times of the year.

This paper reports the results of a study aimed at quantifying the benefits of including travel-time look-up tables, relative to a baseline case of using a single sound-speed value. Benefits are quantified in terms of the reductions achieved in time residuals. A further sub-study is also reported that was intended to investigate the variation of time-residual reduction with different temporal resolutions in travel-time calculations.

The acoustic modelling approach adopted is described in the next section. Section three gives the results of the study in terms of time-residual reductions. These results are considered as a single dataset and are also broken down as a function of the water mass in which signals originated. Section three also includes an assessment of the number of travel-time tables needed per year to achieve the best observed benefits. Conclusions are then drawn.
2. METHOD

The acoustic modelling software used in the study reported here was developed for CTBTO by Science Applications International Corporation (SAIC), Arlington, Virginia. It used the following databases:

- The Sandwell and Smith digital bathymetry database, [2]
- The GDEM-V (Generic Digital Environment Model – Variable resolution) seawater sound-speed database [3] and
- The NOAA/NGDC sediment thickness database [4].

The KRAKEN normal-mode acoustic propagation model was given information from these databases and run to simulate the propagation of sound in the band 6 Hz to 12 Hz. While the IMS hydrophone stations receive signals in a band up to 125Hz, calculation of propagation over the full band was prohibitively expensive in computational terms. The band used was chosen to match one filter band in which CTBTO automatic processing identifies hydroacoustic arrivals. KRAKEN output as a function of frequency was converted to time-domain results by a post-processing module using standard Fourier techniques. Although source-station propagation was the quantity of interest, it was computationally convenient to specify the IMS stations as pseudo-source locations and to calculate propagation to a series of pseudo-receiver locations regularly spaced in range and azimuth, as measured from the IMS station location. Acoustic reciprocity was invoked to relate model results to travel times from candidate event locations to IMS receivers. The software linking databases and model was written in the MATLAB programming language and used a graphical user interface to allow the user to set up the calculations. Results were produced at a range resolution of one point every 55 km, out to a maximum range of 21,000 km and for 720 azimuths spaced by 0.5 degrees.

Travel-time information from the tables produced by the modelling was used to predict the time at which signals from a series of historic events would arrive at IMS stations. The events were selected from a CTBTO database which contains information about all the events which have been detected by the IMS in the decade for which it has been operational. Ideally, signals would be studied from in-water explosions for which detonation time and location were known accurately and independently. However, such events are rare and a large enough dataset could not be produced. Instead, the study concentrated on signals that travelled by waterborne paths, after being generated by earthquakes; so-called T-phases. Processing procedures at IDC [5] allow T-phases to be associated with events but their arrival times and azimuths are not allowed to influence GA’s calculation of event location – a calculation that essentially selects the location which best explains observed arrival times and azimuths at seismic stations. Thus, T-phase time residuals represent a measure of the difference between theoretical and observed arrival times of signals from events whose locations were derived independently of the travel-time information whose accuracy is the subject of this study.

Signals were extracted from the database from events selected using the following criteria:

- Event time within 2007
- Event depth less than 30 km
- Event location determined by GA using arrivals from at least 8 IMS stations
- Semi-major axis of ellipse describing event location uncertainty less than 30 km
• Event resulted in at least one arrival being detected at a hydroacoustic station.

These criteria were derived from previous studies intended to identify events that had locations known to a good degree of accuracy and which resulted in water-borne arrivals. Event time was restricted to lie within 2007 to limit the number of signals studied to a manageable amount – 5148.

For all signals studied, the distance from the receiving station to the event location was calculated and a travel time was predicted using an assumption of a single sound speed (SSS) of 1485 ms$^{-1}$. A second travel time was then calculated by interpolation of the information in the travel-time tables (TTT) produced by the acoustic model. Residuals were then calculated for the two travel times as being the difference between the observed signal arrival time and the time predicted from the event time and the travel time. An investigation of the relative sizes of these two types of residual is given in the following section.

3. RESULTS

Figure 1 shows histograms of the residuals calculated using a single sound speed and using the travel-time tables. Both histograms show a bias in residuals towards positive values, indicating that signals have a tendency to arrive after the time predicted using both the SSS and TTT methods. This is a common feature of seismic residuals and stems from the problem of observing emergent arrivals against a background of noise. The very large positive residuals, on the other hand, probably arise due to incorrectly associated arrivals or poor event locations.

The histograms illustrate an improvement associated with the TTT approach in that fewer large residuals are observed and the histogram is more concentrated in the smaller values. The values in the figure legend indicate that the TTT approach resulted in a 35% decrease in rms time residual, relative to the SSS approach.

Figure 2 shows a map of the locations of the events that generated the signals studied. Events in different ocean areas (north/south-Atlantic/Pacific, Indian and Southern Ocean) are shown by different markers and the percentage reduction in rms time residual is displayed for each area. Marker size is proportional to the time residual for signals from each event and the circles plotted in Central Asia show the residual/marker-size relation. IMS hydrophone and T-station locations are also plotted on the map with ‘H’ and ‘T’ respectively. The figure shows that the benefit of the TTT approach is strongly dependent on the location of the signal-generating event. Indian Ocean events show only a marginal 9% reduction while time residuals from events in the Southern Ocean are reduced to less than half their SSS value. This effect is a consequence of spatial changes in average seawater sound speed. The value of 1485 ms$^{-1}$ represents a good average value for the Indian Ocean but not for the colder, slower Southern Ocean.

The results shown on the map were produced using 12, monthly travel-time tables. The consequences of using 1, 2, 3, 4 and 6 tables equally spread through the year were investigated. It was found that the same reductions in time residuals could be achieved using a single table for each station. This indicates that, although sound speed undoubtedly changes with season, the variation in travel time due to this change is small compared to the other factors affecting residuals.
4. CONCLUSIONS

Time residuals associated with T-phase arrivals at IMS hydroacoustic stations, calculated using travel-time tables generated by acoustic propagation models linked to environmental databases, are significantly smaller than those produced using an assumption of a single average seawater sound-speed. Averaged globally, this reduction is approximately 35% in rms residuals but the reduction achieved shows significant variation with geographic location and can be as high as 51%. Reductions in time residuals were shown to be achievable using one table of travel-time as a function of range and azimuth for each IMS station. Extra reductions achieved by using different tables for different times of the year were not observed to be significant. This is probably because of other influences on T-phase time residuals that are not directly related to the accuracy of seawater sound-speed information. The result is therefore probably not as applicable to residuals associated with signals generated from in-water explosions where these other influences are less important.

The views expressed in this paper are those of the author and do not necessarily reflect those of CTBTO preparatory commission.

5. ACKNOWLEDGEMENTS

The results produced here were made possible by software developed by SAIC and by support and advice provided by many colleagues at CTBTO. Selection criteria for events quoted in section 2 were developed by Frank Graeber.

REFERENCES

Fig. 1: Histograms of time residuals calculated using a constant sound-speed of 1485 m s\(^{-1}\) and using travel-time tables. Numbers give reduction in rms residual for each area, re 1485 m s\(^{-1}\) case. Circle size is proportional to event residual in seconds.

Fig. 2: Map showing locations of events used in study. Marker type indicates ocean area. Numbers quoted give % reduction in rms time-residual for each ocean area. Marker size is proportional to time residual from event in TTT case.
Directionality of Acoustic T-phase transients from shallow submarine earthquakes

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Abstract: Acoustic t-phases radiated from undersea earthquakes were recorded in an experiment carried out using a towed horizontal line array in the South Fiji Basin. The directionality of the t-phases was determined to high spatial resolution using a cross spectral beamformer to process the array hydrophone data. The t-phase signal was highly directional, and the directional pattern throughout the event was different for earthquakes at different sites. For earthquakes in the fore-arc trench at the northern boundary of the basin, strong t-phase components arrived from directions farther south of the epicentre, in a region where a number of seamounts rose within the sound channel. For other events originating along the subduction zone arc, the t-phases radiated from ridge slope sites to the north. A simple model based on ray path travel times for elastic waves in the earth and acoustic waves in the water suggests that the components of the t-phase signal were coupled into the water by downslope propagation at the ridges and seamounts.
Structured Session 25

Bioacoustics

Organizers: Andrzej Orlowski & Egil Ona
Abstract: We develop a method for estimating the mean and standard deviation of the instantaneous velocity of large aggregations of underwater biological scatterers observed by continuous wide area, long range Ocean Acoustic Waveguide Remote Sensing (OAWRS). Our approach is based on analytic expressions for the expected Doppler shift and spread of the acoustic field scattered from fish groups in the continental shelf. These expressions are derived from a model for scattering from a moving target submerged in a stratified ocean waveguide. We demonstrate that the moments of the received field ambiguity function are proportional to the statistics of the fish group’s instantaneous velocity. Illustrative examples are presented for simulated shoaling and migrating fish groups, and we examine the application of the method to data from past OAWRS field experiments. The ability to determine the instantaneous velocity statistics of large aggregations of biological scatterers may offer new possibilities for classifying species and quantifying migration patterns over ecosystem-wide scales.
Using split beam phase angles to determine if the acoustic sampling volume is totally or partially covered by a school

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Abstract: Calibrated and digitized data from multi frequency measurements can be used to discriminate between different biological species in the ocean. This technique is based on comparing the frequency response at the digital sampling level using some sort of discriminant analysis. In order to obtain a correct frequency response, the digital samples should be derived from the same volume, preferably containing targets all across the cross section of the beam. In many cases this is not the case. Frequently, the samples contains data from partial coverage of school and water, and for small schools only a fraction of the pixels may have valid data. Including these samples into the discriminant analysis will decrease its accuracy. This paper investigate the potential use of measured split beam phase angles to determine if the sampled volume is totally covered by schools or not. A filter for extracting the valid data for measuring the frequency response is suggested.
Modelling of sandeel (Ammodytes marinus) target strength

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Abstract: Acoustic target strength of sandeel (Ammodytes marinus) was computed using a finite element approach. Target strength was found as a function of fish tilt-angle and acoustic frequency, and was computed from three-dimensional reconstructions of the fish form. Sandeels lack swimbladder and are small perch-like fish that bury in the bottom substrate. The morphometric data were derived from digitized cross-sections of sandeel as well as high-resolution magnetic resonance images. The finite element method is a versatile tool to estimate fish target strength. It can be used over a great frequency range and with no constrictions on fish tilt angle. It can also in theory handle scattering from all parts of the fish like its swimbladder, flesh, and bone. The disadvantage is the considerable computational cost, especially at high frequencies, resulting in a practical high-frequency limit. Results from the finite element computations were also compared with results from other scattering models and in situ measurements.
COMPARISON OF ACOUSTIC ESTIMATES OF FISH STOCK WITH FISH CENSUS DURING RESERVOIR DRAINING

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\textbf{Abstract:} Hydroacoustic methods are being increasingly used for both fisheries and ecological studies. While usefulness of these methods in deep waters is unquestionable, their accuracy in shallow waters is often doubted. Draining of the shallow Malta reservoir (Poland) provided an unique opportunity to check the accuracy of acoustical estimates of fish stock. Before the draining detailed investigations of the fish population were performed using standard gillnetting and horizontally directed echosounder (split beam Simrad EY500, 120 kHz). During reservoir draining all fish were collected and weighted. Roach (Rutilus rutilus) and perch (Perca fluviatilis) were dominating species both in numbers and weight. All together 11 fish species were caught of total weight equal to 34 348.9 kg that is 536.7 kg ha\textsuperscript{-1}. Fish biomass estimated acoustically under assumption that fish are distributed with random aspect relative to the beam (i.e. using deconvolution) and using Frouzova et al. [2005] regression for TS/length relationship accounted to 548 kg ha\textsuperscript{-1}, which is surprisingly close to the density estimated from the total catch. It has been shown that the TS/length relationship has major effect on fish biomass estimation, while different methods (i.e. based on SED or tracked fish) give very similar results.

\textbf{Keywords:} hydroacoustics, fish biomass, horizontal beaming, reservoir draining, Water Framework Directive
1. INTRODUCTION

Fish stock assessment in inland waters is necessary for both: fisheries management as required by Sustainable Fisheries Act of 2007, and ecological environmental assessments as a result of EU Water Framework Directive (WFD stresses importance of biological indicators and requires monitoring of fish abundance among other parameters). A wide range of sampling techniques have been developed for the assessment of fish populations in lakes and reservoirs [1] including trawling, gill nets, fyke nets, electrofishing, etc. but none of them is suitable for all fish species and all types of habitats [2]. Over the past few decades, hydroacoustics has become increasingly important to the assessment of fish populations in standing waters. All the techniques, both fisheries methods and acoustics can only assess the segment of the population lending itself to sampling, and they have their strengths and weaknesses as sampling techniques. Since traditional fishery methods are both expensive and labor intensive, whole lake evaluations with these methods are infrequent. By contrast, hydroacoustics allows surveying of large areas within a short time, but it is not suitable for species identification. When fisheries methods and hydroacoustics are used together, they provide invaluable information about fish populations in lakes. There is a number of publication related to comparison of different fish sampling gears with hydroacoustics [3, 4, 5], however their results are not unanimous. In general there is a good correlation between abundance and size distribution received by all the gears, but when one comes to absolute values acoustics in some cases overestimate the fish population, in others underestimate as compare to other methods. In the absence of the ground truth data it is difficult to conclude which estimates, received with hydroacoustical or fisheries methods are more reliable, since all of them are subject to many sources of error. To our knowledge there is only one report referring to direct verification of the hydroacoustic methods with total fish stock while reservoir draining [6]. The results reported were very promising for acoustics as a quantitative tool. Estimate based on the echo integration (by transects) amounted to 5.1 tons, while total fish collected during draining was 5.8 tons. When instead of in situ TS, the fish size and weight distributions from gill nets were applied for calculations, the result was even closer, 6 ton versus 5.8. However, this comparison was done for the deep reservoir where acoustical methods perform well. It is commonly accepted that in shallow waters the transformation of acoustic data into an index of fish biomass is not straightforward. The physical limitations of the environment require horizontally aimed transducers to maximize sampling volume. Boat motion, sediment and water surface interface reverberations, and wind-driven sub-surface bubbles may adversely affect acoustic data quality. In addition fish avoidance may contribute to biases in estimates [7, 8, 9]. It is a common practice in lake surveys to apply in situ measurements of TS as a scaling factor to obtain fish densities from the integrated volume backscattering strength [10]. However, horizontal acoustic measurements of TS are problematic because there is no way to determine the orientation of the fish relative to the axis of the acoustic beam. The relationships between TS and length for the side aspect and random orientation have been developed under laboratory conditions [11, 12, 13], but only few researches have compared in situ measurements of fish length from horizontal beaming with measurements collected by fisheries methods [14, 15]. Development of acoustic technology with very narrow beam transducers and negligible side lobes, together with new sophisticated software allowing for noise removal [16] make nowadays possible to survey horizontally very shallow waters with depths between 1.5-5 m, so more data can be gathered.
The aim of the present paper was to estimate fish biomass in very shallow reservoir (mean depth < 3 m) using combined: nets and mobile acoustic surveys and to compare it with the result of direct fish count while draining the reservoir.

2. MATERIALS AND METHODS

The experimental work was conducted in September and October 2008 in Malta Reservoir (total area 64 ha, volume 2x10^6 m^3, mean depth 2.8 m, maximum depth 5.5 m), located in the city of Poznań (western Poland). This is a very shallow hypereutrophic reservoir which serves recreational, mainly sport activities. During summer chlorophyll $a$ concentrations fluctuate between 7.5 – 58 µgd⁻¹m⁻³, mean Ptotal around 0.2 mgdm⁻³, and Ntotal 0.6 mgdm⁻³.

Hydroacoustic measurements were conducted from the 5 m long boat “Echo” along parallel transects at the constant speed of 8 km.h⁻¹ (Fig. 1). The total length of transects was about 8.4 km, which gives the coverage coefficient, defined as the ratio between the total transects length and the square root from the area under study around 10 according to Aglen [17]. This is high enough to expect coefficient of variation for biomass estimate to be well below 10% [18].

![Fig. 1 The contour of the Malta reservoir with bathymetry and survey transects](image)

Hydroacoustic surveys were conducted both during the day, and at night during complete darkness. The Simrad EY500, split beam echo sounder with frequency 120 kHz and elliptical transducer (opening angles at -3dB were 4 and 10 degrees) were used. The transducer was aimed horizontally (2 degrees from the surface, beaming perpendicularly to the direction of the boat) and was fixed on the side of the boat on a special frame at the depth of 0.5 m. The pulse duration was set to medium (0.3 ms), the ping repetition rate to 5 Hz, and the TS and Sv thresholds to –50 dB and –56 dB respectively. Simrad post-processing software EP500 and Sonar 5-Pro software version 5.9.6 [16] were used for data analysis. Target strength frequency distributions were received by the automatic track analysis of Sonar 5. Tracking was based on single echo detections defined by 0.8-1.3 relative pulse width, a one-way beam compensation of 3 dB, and a maximum phase deviation of 0.3. To build a track the following
criteria were set: at least 3 echoes for the same target, separated by a maximum one missing ping within a 0.3 m gating range. From each track, the average TS from successive echoes was calculated in linear domain. The received histograms were de-convoluted to account for random aspect of fish distribution and than used for scaling the integrator values.

Data for length-frequency and species composition were collected in September and October 2008 from the research catches using Nordic gillnets [19]. During four fishing occasions 24 gillnets were deployed in total. Nets were set between 19:30 to 21:30 at the depth 1.5-2.5 m perpendicularly to the shore and lifted in the following morning between 6:30 to 7:30. The catch was sorted to species, weighted and total length of each individual fish was recorded to the millimetre below. A beach seine with a wing length 150 m, mesh sizes 25 mm, and mesh sizes 12 mm in a cod-end was used for commercial fishing during draining of the Malta Reservoir. Fifteen samples of total weight 624 kg were collected from the beach seine catches for species and size composition, and average weight of fish species. A natural logarithmic transformation was applied to length and weight measurements and linear regression was applied to estimate parameters of the weight-length relationship described by the allometric equation $W = aL^b$, where $W$ is total wet weight (g) and $L$ is total length (cm). Parameter values were estimated with the Arc software for linear regression analysis [20].

3. RESULTS AND DISCUSSION

Converting acoustic energy into fish biomass involves several steps: 1- estimate mean TS from the observed data (based on single echo detections (SED), or tracked fish, criteria for which affect final results); 2 - estimate acoustic fish density (using Sv/TS scaling method with different source for TS or by echo/fish counting); 3 - convert mean TS to mean fish length using equations appropriate for given species and size range (for the horizontal beam there are very few of these equations); 4 - convert mean length to mean weight using equations appropriate for given species and size range; and finally 5 - calculate fish biomass by multiplying mean density by mean weight. Consequently, if any of these estimates are inaccurate, error will propagate and result in wrong biomass estimates.

When beaming horizontally no information on fish orientation relative the acoustic beam is available, practically all horizontal-aspects are possible. Several equations [11, 12, 13, 15] have been derived to convert horizontal-aspect measurements of acoustic energy into fish length. The equations were derived from different fish assemblages, and therefore may not be the best suitable for our applications, with different species and size ranges [10] but better data were not available. As can be seen from the Fig. 2 the number of registered fish along the individual transects was pretty stable for a given month and part of a day, with coefficient of variation not exciding 5 %. However, differences between the months (September and October) and between the day and the night were substantial, indicating that only night data from September can be used for comparison of fish stocks. The same is confirmed by the maps of fish distribution (Fig. 3) based on day and night surveys, which clearly show that during the day some of the fish stayed in places not available for sonar detection.
Fig. 2. Number of tracked fish along each of the transects in September and October during the day and night (each transect was run in two directions with transducer looking once to the right and once to the left from the boat).

Fig. 3. Maps of fish distribution in Malta reservoir during the day (above) and night (below).
The large difference between the September and October acoustic estimates suggests that the change in fish behavior had place making fish less detectable to sonar in October. Most probably it was related to diurnal horizontal migrations of roach, which was the dominating species in the reservoir [21]. The fish either stopped their migrations to open water in October and stayed close to the shore being undetectable, or they were distributed so close to the bottom that they could not be distinguished from it. High gillnet catches of roach and perch in the inshore zone indicate that substantial fish densities may not be detected by sonar. Some reduction of fish abundance could also be due to the mortality of 0+ fish, but it would not affect the biomass so much. So in our opinion only September data can be used for comparisons with the total catch.

Acoustic estimates of total fish stock were based on echo-integration only. The echo counting could not be applied due to low share of single echoes (between 20 and 30%). The results were compared for 3 different sources of mean TS: from SED, tracked fish and the catch basket, and different regressions for TS/length relationship (Table 1).

<table>
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</thead>
<tbody>
<tr>
<td></td>
<td>Side aspect</td>
<td>Side aspect</td>
<td>Side aspect</td>
<td>deconvolution</td>
</tr>
<tr>
<td>SED</td>
<td>87</td>
<td>104</td>
<td>227</td>
<td>548</td>
</tr>
<tr>
<td>Tracked fish</td>
<td>81</td>
<td>104</td>
<td>211</td>
<td>492</td>
</tr>
<tr>
<td>Catch basket</td>
<td>46</td>
<td>102</td>
<td>128</td>
<td>337</td>
</tr>
</tbody>
</table>

The highest acoustic estimate in September was 548 kg ha\(^{-1}\) for the regression of Frouzova et al. [13] for freshwater European species with de-convolution. For the other regressions accounting only for the side aspect the estimate was lower and fluctuated roughly between 80 and 200 kg ha\(^{-1}\), which confirms that probably fish aspect distribution was random. If we take from acoustics only the estimated fish density i.e. the number of fish per unit volume and multiply this by the average fish weight from the catches, we arrive at 550 kg ha\(^{-1}\), thus the value very close to acoustic biomass estimate and the fish census after draining. The most important factor affecting acoustic estimates of fish biomass seems to be appropriate relation between the fish length and TS for the given population that is the species, size range and aspect distribution. The same was conclusion of Boswell et al. [15] who applied structural equation models (SEMs) to evaluate how the choice of TS-fish length equation affected estimation of fish biomass, and how error occurred and propagated during this process. Comparison of fish length distributions from \textit{in situ} measured TS and TS calculated from the catch (Fig. 4) shows substantial differences, indicating that the regression used was not the ideal one for determining fish size distribution.

Based on fish catches we estimated fish population structure, size frequency distributions (Fig. 4) and weight/ length relationships for the dominating species (Table 2). Roach and perch were the most abundant species both in numbers and in weight (they made over 80% of the population). The total catch with the beach seine (only fish larger than 10 cm were taken into account) was 34348.9 kg or 536.7 kg ha\(^{-1}\). In the old River Cybina bed (approximately 2.5 km long, 5 m wide and 1.6 m deep) density of fish was estimated after draining of the reservoir with the help of electrofishing. An average catch weight in two electrofishing occasions carried on a 100 m reaches was 100 kg. It makes a total biomass of fish left in the old river
bed equal approximately 2500 kg, which increases the total fish density to 575.8 kg ha\(^{-1}\). This value is very close to one estimated acoustically, proving that even in shallow lakes acoustics can be a very useful tool for estimating fish abundance.

Table 2. Parameters of the weight-length (W-L) relationship of selected fish species from the Malta Reservoir.

<table>
<thead>
<tr>
<th>Species</th>
<th>a</th>
<th>b</th>
<th>R(^2)</th>
<th>length range (cm)</th>
<th>weight range (g)</th>
<th>n</th>
</tr>
</thead>
<tbody>
<tr>
<td>Roach Rutilus rutilus</td>
<td>0.00574</td>
<td>3.2258</td>
<td>0.9521</td>
<td>5.0 – 30.5</td>
<td>1.0 – 382.0</td>
<td>526</td>
</tr>
<tr>
<td>Bream Abramis brama</td>
<td>0.00485</td>
<td>3.2487</td>
<td>0.9958</td>
<td>5.4 – 51.6</td>
<td>1.2 – 1832.0</td>
<td>109</td>
</tr>
<tr>
<td>Perch Perca fluviatilis</td>
<td>0.00517</td>
<td>3.2992</td>
<td>0.9925</td>
<td>7.1 – 40.0</td>
<td>4.0 – 981.1</td>
<td>601</td>
</tr>
<tr>
<td>Pikeperch Sander lucioperca</td>
<td>0.00308</td>
<td>3.3057</td>
<td>0.9896</td>
<td>13.3 – 63.0</td>
<td>19.5 – 2879.0</td>
<td>117</td>
</tr>
</tbody>
</table>

Fig. 4. Fish length frequency distribution from acoustics (left) and recalculated to dB according to regression for all European species horizontal (right, Frouzova et al. 2005) in the Malta Reservoir

4. ACKNOWLEDGEMENTS

This work was funded from the project NN304 052234 from the Polish Ministry of Science and Higher Education granted to M. Godlewska.

REFERENCES


IDENTIFICATION AND CHARACTERIZATION OF FISH SCHOOLS USING THE SIMRAD MS70 3D MULTIBEAM SONAR

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Abstract: The Simrad MS70 scientific multibeam fishery sonar provides an extension from two to three spatial dimensions plus one temporal dimension, and allows for a more accurate characterization of fish schools. Characteristic variables like the centre of mass and geometric dimensions of schools may be calculated based on segmentation values defining which voxels (volume elements) belong to the school. The paper considers an existing segmentation algorithm based on a flood-fill method. By inspecting the ratio between total acoustic backscatter and total volume of a school, the flood fill algorithm is found to perform unsatisfactorily. Suggestions for an automatic method based on characterization of the statistical properties of the noise in the data are presented.

Keywords: Multibeam sonar, school characterization
1. INTRODUCTION

The introduction of time series of three dimensional images provided by multibeam sonars enhances the potential for statistical analysis of acoustic observations of fish schools. As a first step in the analysis, a school needs to be segmented by identifying which voxels (volume elements) belong to the school. In this paper an existing segmentation algorithm based on a flood-fill method is described, and its performance tested on a school observed by the Simrad MS70 multibeam sonar. Considering the total volume and total backscatter from the school, the segmentation algorithm is shown to perform unsatisfactorily. Suggestions for an improved segmentation algorithm are given.

2. MATERIALS AND METHODS

2.1. The MS70 sonar

The MS70 sonar is mounted with port oriented beams in the port instrument keel of RV “G. O. Sars”, and covers 60° horizontally and 45° vertically, which is often enough to cover a school in one ping. The sonar was operated in continuous-wave (CW) mode, with pulse duration equal to 2 ms. The MS70 sonar emits sound in 20 horizontal fans in the frequency range [75,112] kHz, but use sub-bands for each fan. The frequency for each adjacent fan transmitted at sequentially lower frequencies until the 20th fan, aiming at 0° towards the sea-surface, transmitted at 75 kHz. Four fans transmitted simultaneously (i.e. 112 kHz at 45°, 113.9 kHz at 47.5°, 115.7 kHz at 50°, and 117.6 kHz at 52.5° transmitted concurrently, and followed by the next four, etc.). The sonar received sound in 25 x 20 beams, where the -3 dB beam widths were between 3° and 4°, varying vertically with the frequency. The first side lobe was –35 dB relative to the main lobe vertically, and –25 dB horizontally.

![Fig. 1: The beam array of the MS70 sonar. Figure used with permission from Hans Petter Knudsen, Institute of Marine Research.](image)

Each beam is partitioned radially into voxels of thickness0.38 meters. Beyond 1100 voxels in the radial direction, the signal to noise ratio is usually dominated by vessel and background noise, limiting the range of the sonar to approximately 400 meters. The number of voxels at each time step thus exceeds 500 000, providing high resolution especially in the radial direction.
The MS70 is quantitative sonar, which means that it can be calibrated. The calibration method [2] was still being developed at the time of data collection.

2.2. Materials

A school of herring was observed over 12 pings (approximately 60 seconds) during a survey outside Røst, to the west of Bodø in Northern Norway, at the 25th of January 2006. The research vessel did not circle the school, but passed it in a straight motion. The size of the school was found to be approximately 150 m × 100 m × 40 m. During the observation the ping-rate was in the range [4,6] seconds.

Along with the data, segmentation values were provided, classifying voxels as either belonging to the school or not. These segmentation values were calculated using a flood-fill algorithm provided by the sonar visualizing application Sonar Explorer [1].

In the flood-fill algorithm in Sonar Explorer, the user selects a point in the middle of the school by visual inspection, defining a reference value $r$. To reduce variation, a smoothed version of the data is used. Propagating outwards from that point by a flood-fill method, the algorithm calculates identification values representing the degree of membership of the voxels to the selected school. These identification values are calculated from the fall-off function $f$ given by the expression

$$f(x) = 1 - \left( \frac{|x - r|}{t} \right)^2.$$

(1)

Here $x$ is the value of the voxel being processed and $t$ is a tolerance value defined by the user. The fall-off function $f$ is clamped to $[0,1]$. All voxels with absolute deviation from the reference value $r$ exceeding the tolerance value $t$ will thus have $f = 0$. When the school is surrounded by voxels having fall-off value $f = 0$, the identification of the school is complete.

2.3. Statistical analysis

For the school of herring observed by the MS70 sonar, total backscatter $B_T$, total volume $V_T$ and the ratio $B_T / V_T$ were calculated, based on the segmentation values. The total backscatter $B_T$ was calculated as the sum of the products of volume backscatter $s_v$ (linear values) and volume $V$ of the segmented voxels. The centre of mass $\xi$ at each ping was calculated as an average of midpoints of the voxels of the school, weighted by the products $s_v \cdot V$ of the voxels. From the trace of $\xi$ it is possible to identify the motion of the school. If the school has not been accurately segmented, the quality of these variables may be compromised.
3. RESULTS

Based on the estimates of the variables $B_T$, $V_T$ and $B_T/V_T$ for the school of herring observed by the MS70 sonar, there is evidence that the segmentation algorithm did not perform accurately in this case. Figure 3 shows that both the total volume and the total backscatter increase with time, while the ratio of total backscatter and total volume seems to be constant. This implies that more voxels containing fish are added to the segmentation, either because fish are joining the school or because the school is not accurately segmented. As the probability of having two schools merging at the particular time of the observation is considered to be small, it is assumed that the inconsistency with regard to the total backscatter and total volume is due to inaccurate segmentation.

Two possible causes for the unsatisfactory performance of the flood-fill algorithm are improper choice of the tolerance level $t$, and that the reference value $r$ by chance could have been selected to be not entirely representative for the school. This calls for an automatic segmentation algorithm, in which parameters are selected by rules formulated based on statistical theory.

With an uncertainty in the segmentation, variables like $\xi$ must be interpreted with care. Figure 3 displays the trace of the estimated centres of mass of the observed school. For the first two pings the estimated centres of mass deviates from an expected linear motion of $\xi$. However, due to the possible errors in the segmentation of the school, it would be unwise to draw strong conclusions concerning the trace of $\xi$. 

![Graphs showing total volume, total backscatter, and ratio between total backscatter and total volume of the school.](image)
Fig. 3: Centres of mass of a school observed by the MS70 sonar, where the first ping is in turquoise and the 12th ping (farthest from the observer) is in green. Values are in meters in a reference coordinate system centred at the first vessel position.

4. CONCLUSIONS

The results from the school observed by the MS70 sonar, presented in this paper show the need for a segmentation method that is accurate, so that the analysis of properties of schools can become reliable. The flood-fill algorithm used to obtain the results is dependent on visual inspection and choice of tolerance level by the user. The algorithm does not contain an analysis of the noise present in the data. An algorithm should be developed that is automatic and that takes all types of noise in the data into consideration (vessel noise, transducer noise, background noise and so on). Based on statistical theory rules for the selection of parameters in the algorithm should be defined. Work on this is in progress.

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REFERENCES


EXTRACTING BOTH FISH AND WATER VELOCITIES FROM DOPPLER PROFILER DATA: EXAMPLES FROM COD IN SMITH SOUND NEWFOUNDLAND

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Abstract: With standard methods of inverting Doppler sonar data, fish velocities can only be determined if all of the profiler beams sample the velocity from a large fish school at the same time. Consequently, you can either measure water velocities (when few fish are present), or fish velocities (when many fish are present). Notably, you cannot determine water velocities and fish velocities at the same time. We apply a new method of analyzing Doppler profiler data that allows extraction of both water and fish velocities from data with comparatively high concentrations of fish. The method was used to observe over-wintering Atlantic cod that aggregate in Smith Sound Newfoundland. Currents in this enclosed coastal area are slow (about 10 cm s\textsuperscript{-1}) and the fish appear to move passively with the water much of the time. However, there are times when the fish have velocities different than those of the water and averaged velocity profiles show clear differences in fish and water velocities.

Keywords: Fish Acoustics, Doppler sonar, Fish Behaviour
1. INTRODUCTION

The presence of fish can lead to errors in data collected with Doppler sonar systems. The problem arises because Doppler profiling assumes that volume backscatter is Doppler shifted in proportion to the water velocity. If the backscatter has come from actively swimming fish, it can lead to biases ([1] and [2]). In some cases this “contamination” can be used to observe characteristics of fish behaviour (see for example [3] and [4]). Extraction of fish velocities has only been possible if all of the acoustic beams of the Doppler profiler are sampling velocities of the same fish school simultaneously. Given that the beams can be separated by a distance of 100 m, and that data must be averaged over a time interval measured in minutes, this method is not generally applicable. In this paper, we report on an alternative method of processing the Doppler sonar data that does not require the simultaneous presence of fish (or water) in the multiple Doppler beams. As a result, it can be used to extract water and fish velocity at the same time as long as a sufficient number of samples can be accumulated. We apply the method to observations of Atlantic cod (Gadus morhua) over-wintering in Smith Sound, Newfoundland.

2. METHOD

Normally with Doppler profiling systems, velocity components are measured by multiple acoustic beams. It is then assumed that all of the multiple beams are sampling components of the same velocity so that the measurements can be combined to estimate that velocity. For example, for an instrument with acoustic beams oriented in horizontal directions 0°, 90°, 180°, and 270° and downward at 20° to vertical as shown in Fig. 1, orthogonal velocity components can be found as:

\[
\begin{align*}
V_x &= \frac{v_2 - v_1}{2\sin \theta} \\
V_y &= \frac{v_4 - v_3}{2\sin \theta} \\
V_z &= \frac{v_1 + v_2 + v_3 + v_4}{4\cos \theta}
\end{align*}
\]

(1)

where, \(v_1, v_2, v_3\) and \(v_4\) are the measured beam velocities (positive toward the transducer), and \(V_x, V_y\) and \(V_z\) are the desired velocity components. If one of the beam velocities is corrupted, Equations (1) is incomplete and cannot be solved because the information in the four beams is assumed to be linked. The use of four beams does afford a level of redundancy so that the problem can still be solved, however, if a second beam is corrupted or only three beams are used, this is not the case. In the case of fish detections, the backscatter from the fish provides a legitimate velocity estimate and it need not be discarded but the approach to solving for velocities given by Equation 1 cannot be used.

The loss of data can be avoided if the measured velocities (\(v_1, v_2, v_3\) and \(v_4\)) are considered to be independent samples of velocity [5]. In that case, each velocity estimate is considered as:
Fig. 1: Doppler profiler coordinate system for a four beam, downward looking system. Vectors labelled \( v_1, v_2, v_3, \) and \( v_4 \) identify the orientation of the four acoustic beams and measure velocity components \( v_1, v_2, v_3, \) and \( v_4 \) used in Eq. 1.

\[
v_j = \tilde{V}_j \hat{k}_j = V_x k_{xj} + V_y k_{yj} + V_z k_{zj}
\]  

(2)

where \( \hat{k}_j \) is a unit vector defining the direction of the acoustic beam making the \( j \)th sample: for example, in the absence of any rotations, beam 1 would sample \( \hat{k} = (-\sin 30, 0, \cos 30) \). Heading, pitch and roll are incorporated by adjusting the value of \( \hat{k}_j \). In Equation 2, there is no reference to beam number, that information is contained in the value of \( \hat{k}_j \) so that all observations are considered equal and independent. The velocity is found by forming an error function as

\[
\sum \varepsilon_j^2 = \sum (V_x k_{xj} + V_y k_{yj} + V_z k_{zj} - v_j)^2
\]  

(3)

where \( \varepsilon_j^2 \) is the “error” in the \( j \)th measurement and \( k_{xj}, k_{yj},\) and \( k_{zj} \) are the x, y, and z components of \( \hat{k}_j \). Optimal values for \( V_x, V_y, \) and \( V_z \) are found in the conventional least-squares sense by minimising \( \sum \varepsilon_j^2 \). Fish and water velocities can be extracted using Equation (3) if individual observations (given by \( v_j \)) can be distinguished as coming from fish or water based on the backscatter strength.

3. OBSERVATIONS

Two, RD Instruments, 300 kHz Acoustic Doppler Current Profilers (ADCP’s) were deployed in water of 205 m depth in Smith Sound Newfoundland from December 2004 until May, 2005. This area was attractive for testing the new processing scheme because it is used by overwintering Atlantic cod (\textit{Gadus morhua}) [6] and therefore assured a high probability of finding significant concentrations of fish. Both instruments were moored at 150 m depth with one directed upward and the other downward: we will only consider the down looking
instrument here. In order to optimise the detection of individual fish, the instruments were configured with relatively small depth bins (1.2 m), and profiles were acquired as rapidly as the memory and battery constraints would allow: 3 minutes per profile. The instrument was configured to collect data in “beam” coordinates so that velocities could be extracted after data recovery using Equation (3).

Significant concentrations of fish were seen during 80 days of the deployment. An example of 6 days of data is presented in Fig. 2: this sample interval was selected because it shows a good range of conditions upon which algorithm performance can be evaluated. Backscatter coefficients calculated as described by [7] are shown in Fig. 2a for one of the acoustic beams: fish are present at around 180 m depth with the concentration decreasing toward the end of the record. For the purpose of extracting fish velocities, backscatter was considered to be caused by fish when it exceeded -55 dB. Velocity data was extracted from the 175-180 m depth interval indicated by the dashed lines in Fig. 2a. Water velocities were extracted every hour but fish velocity estimates were enhanced by averaging data over 3 hour intervals. Data is only plotted for intervals where velocity standard deviations were less than 5 cm s$^{-1}$, water velocities always meet this criterion. East velocity components are shown in Fig. 2b and north components in 2c, solid lines are fish velocity and dotted lines are water velocity. Fig. 2 shows that fish and water velocities agree much of the time but there are times when there are clear differences (consider the interval between day 60 and 61).

Fig. 2: An example of six days of data: a) acoustic backscatter ($S_v$ in dB re 1 m$^{-1}$). b) is East component of fish (solid line), and water (dotted line) velocity averaged between 175 and 180 m depth, and c) is East component of fish (solid line), and water (dotted line) velocity averaged between 175 and 180 m depth. The dashed lines in a) identifies the depth interval sampled for velocity in b) and c).

The difference between fish movements and water movement can be seen in an average profile over the six day interval being considered, (Fig. 3). In Fig. 3, velocity reported by the
ADCP is indicated by a continuous line, stick vectors with dotted lines represent water velocity extracted with Equation (3), and solid stick vectors are fish velocities. Both estimates of water velocity agree between 155 and 175 m depth where no fish are present. Beyond 175 m depth, fish and water velocities diverge: in this interval the ADCP computed velocities are biased by the presence of the fish.

![Graph showing velocity profiles](image)

*Fig. 3: Profiles of velocity averaged over the 6 day period starting on day 59, 2005: ADCP determined velocity (continuous line), fish velocity (solid stick vector), and water velocity (dotted stick vector). Standard errors are indicated by crosses at the end of the stick vectors.*

4. CONCLUSIONS

We have presented a method that allows extraction of both fish and water velocities from the same Doppler profiler data (Equation (3)). The method can be applied when intermediate concentrations of fish occur so that the profiler sees backscatter from both fish and water. At extremely high concentrations of fish, water velocities cannot be extracted and conversely, at very low concentrations, estimates of fish velocity become inaccurate. This method has been applied to observations of over-wintering Atlantic cod in Smith Sound Newfoundland. In an example of six days of data, for most of the time the cod move passively with the water. However, there are times when significant velocity differences between fish and water occur and this behaviour is consistent through the entire 80 days when large concentrations of cod are observed in the data set. Time average profiles (in the present example based on 6 days of data) show substantial differences between water velocity and fish velocity. In such a case, were water velocities to be measured by the conventional Doppler profiler approach, substantial biases in the velocity estimate would result.
REFERENCES


STANDARDIZATION OF HYDROACOUSTIC METHODS - EFFECT OF PULSE DURATION

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Abstract: Assessing the ecological status of water bodies is currently a major challenge for all European countries. There is an urgent need in developing new monitoring tools that will be simple, cost-effective, fast and accurate. Acoustics is one of the most powerful tools for studying aquatic environments, however to enable the comparability between different water bodies and countries it requires standardization. The aim of the present paper was to investigate effect of pulse duration, one of the acquisition parameters, on estimated acoustically fish abundance and size distribution. Measurements were performed using SIMRAD EK60 split beam echo-sounder, 70 kHz, with two identical transducers pinging alternatively through the multiplexer. The results have shown that while acoustical parameter, $S_v$, seems not to depend on pulse length, the estimated fish density and the fish size distributions do depend, especially in a surface layer with dense fish populations. Conclusion is that if acoustics is to be used for the classification of the ecological state of an ecosystem, the standard acquisition parameters must be determined.

Keywords: hydroacoustics, Water Framework Directive, fish biomass, pulse length
1. INTRODUCTION

Hydroacoustic studies are increasingly being used in all types of aquatic ecosystems in order to acquire the detailed information about the aquatic living resources, and particularly about the fish [1]. Recent technical improvements in hydroacoustic methods, both in hardware and software, make them an efficient but quite sophisticated tool, which is used more in academic research than in practice. It seems that hydroacoustic instrumentation has now matured to be used also routinely in a number of applications, especially in fisheries, however, it is necessary that the internationally accepted standards for system specifications and different task procedures are set out to ensure comparability of results between the different water bodies and between different countries. This is not an easy task, as not only the systems developed by different manufacturers differ greatly in specifications of both hardware and software, but also the sensitivity of results to variation in parameter settings is poorly investigated. Some work on comparison of different systems and frequencies has been already done: Rudstam [2] compared the performance of single- and split-beam devices, while others have compared dual- and split-beam devices [3, 4]. Comparison of the Simrad and Biosonics systems has been reported by [5]. Comparison of different echo-sounder frequencies have also been undertaken [6, 7]. However up to now no systematic work on comparison of acquisition and analysis parameter settings within a single system have been performed.

In situ target strength (TS) is theoretically the optimal measure to scale echo-integration values to fish density [8], so the precise knowledge of TS distributions is of primary importance for the proper fish biomass estimation. It is widely recognized that the frequency of sound is decisive for the smallest size to be resolved with an echo-sounder, but little attention has been paid so far to the variability due to the different pulse length used in the studies. The question is: do all the pulse lengths equally well describe the fish size distribution? The aim of this paper was to answer this question and to investigate which consequences, if any, has the pulse length on the fish density estimates. Fish abundance is among the biological quality elements used by the European Water Framework Directive for the classification of the ecological state of the ecosystem, and thus it is crucial to provide an absolute estimate of it as accurate as possible.

2. MATERIALS AND METHODS

The measurements were performed from 8 to 10 September 2008 in Lake Hancza, Poland (area 330 ha, max depth 108 m). The lake is oligo-mesotrophic and it contains very diversified fish population, with roach (Rutilus rutilus), perch (Perca fluviatilis), vendace (Coregonus albula L.) and white fish (Coregonus lavaretus) being the dominating species. The SIMRAD EK60, split beam, 70 kHz echo-sounder was used with two identical circular transducers with nominal beam angle of 11 degree, pinging alternatively through a multiplexer, the ping rate was 5 pings per second each. Both transducers were aimed vertically down and they were mounted on a special frame one after the other as close as possible (distance between the two transducers was less than 2 cm) so that one can assume that the same fish population was registered by both transducers. Temperature and oxygen profiles were taken at the deepest point of the lake from the surface to 25 m, every one meter before the survey and during calibrations. Special attention has been paid to properly
calibrate the two transducers to be sure that the calibration parameters are not the additional source of variability. The measurements were conducted from a small boat sailing at a speed of 8 kmh⁻¹ and geographical positions were recorded by the GPS connected to a sounder. Surveys were starting 1 hour after sunset, at complete darkness, when all fish were scattered. Every half an hour a different pair of pulse length combinations was chosen — all together four pulse lengths (0.128, 0.256, 0.512 and 0.1024 ms indicated as short, medium1, medium2, long, and thus 16 combinations were investigated. Data were stored in a computer and later processed by the Sonar 5 Pro analysis software [9]. The TS threshold was set to -65 dB in the upper layer and -60 dB in the lower layer (based on in situ TS distributions), and Sv threshold to -55 dB. The criteria used to distinguish individual targets i.e. single echo detections (SED) were set to default values, minimum and maximum returned pulse width 0.7 to 1.3 times the transmitted pulse duration, maximum gain compensation 3 dB (one way) and maximum phase deviation of 0.3 degrees. To estimate fish density integration method was used with interval circa 250 m.

3. RESULTS

The volume backscattering strength Sv (MacLennan et al., 2002) is a basic acoustic parameter which characterizes the amount of energy reflected by the unit volume of a water body under investigation. Fig. 1 presents comparison of the volume backscattering strength Sv for all the pulse lengths (sixteen combinations) for two layers: 1.8-11.8 m, and 11.8 – bottom. There is a high linear correlation between the two sounders for the two layers, showing no difference between the Sv measured using the different pulse lengths.

![Fig.1: Sv values in dB received with transducer 1 and transducer 2 operating at different pulse lengths. Filled rectangles are above thermocline, empty - below the thermocline, the solid line is the regression line](image-url)

**Fig.1:** Sv values in dB received with transducer 1 and transducer 2 operating at different pulse lengths. Filled rectangles are above thermocline, empty — below the thermocline, the solid line is the regression line.
The other important acoustic parameter is the target strength of fish (TS), which characterizes amount of energy reflected by single individual. Although the total energy reflected by fishes did not differ between the two transducers operating with different pulse lengths (Fig. 1), the number of single echoes differed greatly, especially in surface layer, where short pulse detected the largest number of single echoes and the long pulse the smallest (Fig. 2). In a deep layer the difference was not so strong, but also clear.

![Fig.2: The single echoes TS distributions for different pulse lengths above the thermocline, and below the thermocline](image)

Volume backscattering strength, $S_v$ and the mean TS of single echoes were used to calculate fish density according to the equation:

$$\rho \ [\text{ind} m^{-3}] = 10^{0.1(S_v-TS)}$$

i.e. using $S_v/TS$ scaling method. The calculation was performed for each transect (Fig. 3). Estimated fish densities below the thermocline were two orders of magnitude lower than in a surface layer, above the thermocline. In general the densities estimated with different pulse lengths were similar, but in the surface layer the was observed a large difference at transects 5 and 15, and for the deeper layer at transect 10. The detailed inspection of the results leads to conclusion that high differences appear at high fish densities and when difference between pulse lengths is large.
4. DISCUSSION

High correlation of $Sv$ values with the regression line passing through zero and the slope equal to unity as received for all pulse length combinations in Lake Hancza means that the total energy reflected by fish and received by both transducers was independent of the pulse length. However, the number of single fish echoes received by transducers with different pulse lengths did differ, just because of difference in the resolution, which is pulse length dependent ($C\tau/2$, where $C$ is sound velocity and $\tau$ is pulse length).

For transducers operating with the different pulse length large differences in the number of detected echoes were observed, especially in the surface layer, 1.8 -11.8 m. This led to significant differences in the TS distributions and in consequence in fish density at some of the transects.

Fig. 2: Comparison of fish densities along the transects estimated with different pulse lengths
The results have shown that volume backscattering strength \( S_v \) is practically independent of the pulse length and could be used as an indicator, characterizing fish abundance. The major advantage of using the acoustical parameters is providing more comparable monitoring data between regions and countries. However they have no direct biological meaning and it might be difficult to convince stakeholders to use them as indicators of the ecological state of the ecosystem. The biological parameters derived from acoustic data, such as number of fish per hectare or fish size distribution are, in general, pulse length dependent. So if derived from acoustics biological parameters are to be used as ecological quality indicators for the WFD, the standard procedure should include the length of the sound pulse to be used in monitoring.

5. ACKNOWLEDGEMENTS

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REGRESSIONS FOR CONVERSION BETWEEN TARGET STRENGTH AND FISH LENGTH IN HORIZONTAL ACOUSTIC SURVEYS

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Abstract: Conversions, in both directions, between fish target strength (TS) and body length (L) are extremely important for the accuracy of acoustic fisheries surveys. At present there is no universal method available for correcting for the angle at which the beam meets the fish during mobile horizontal surveys. It is strongly recommended that the common assumption of random fish aspect distribution is checked by a series of fixed-location observations. For random aspect situations, the deconvolution of the aspects of fish in the beam can be achieved by converting all TS to side aspect where the relationship to length is well known. For converting catch data to TS, the regression of TS of average horizontal backscattering strength (TSAS) on length can be applied successfully. The average regressions for common European species are being suggested for routine acoustic surveys, in combination with direct pelagic fishing for the assessment of ecological status of lakes and reservoirs for the EU Water framework directive 2000/60/EC. This approach reduces the resolution of local fish size variations but simplifies the processing and provides much better estimates of fish abundance.

Keywords: Fish Stock estimate, length, abundance, biomass, backscattering, echo-integration
1. INTRODUCTION
Large inland waters of Europe usually have a significant epipelagic fish community, represented mainly by the family Cyprinidae, inhabiting the upper 5 m of the water column [1, 2]. Unbiased acoustic sampling of these fish is notoriously difficult due to high noise background, sound field deformities, vertical microstratification of the living organisms and the different scattering properties of underwater objects when scanned horizontally compared to a vertical aspect [3]. Recent advances in reduction of the noise effect [4, 5], and modifying the surveys according to the stratification conditions, can help with many fish-independent complications, but the high variation of fish target strength from different angles is almost always encountered (see [6] for the latest review). Strong acoustic directivity of the fish’s body has important consequences for the sizing of fish (defining of fish length or weight from acoustic target strength – TS) and for setting the threshold defining what we consider to be the fish of interest.

2. MATERIAL AND METHODS
The acoustic records of known fish, scanned horizontally from all possible angles, were determined on the fish rotating carousel on the axis of a Simrad ES 120_4 split beam transducer (8.3 x 4 degrees, EY 500 split beam echosounder), and converted to the backscattering cross section (σ_{bs}) using the data of [7]:

\[ \sigma_{bs} = 10^{\frac{TS}{10}} \]  

(1)

The average value for all 360° of the horizontal plane was calculated as indicated in Fig.1. This operation was carried out for 56 fish of six common European freshwater species (brown trout, Salmo trutta, common carp, Cyprinus carpio, roach Rutilus rutilus, bream Abramis brama, bleak, Alburnus alburnus and perch, Perca fluviatilis) with a standard length range of 7-71 cm. For the average \( \sigma_{bs} \), the “TS of average sigma” (TSAS) was calculated as follows:

\[ TSAS = 10 \log (\text{average } \sigma_{bs} \text{ for all } 360^\circ) \]  

(2)

The fish stock of the Orlík Reservoir, Czech Republic, was surveyed in 2008 using a Simrad EK60 split beam echosounder (120 kHz, horizontal survey: ES 120_4 split beam transducer, pulse duration 0.128 ms, frequency bandwidth 10.92 KHz, in situ receiver calibrated gain 25.78 dB, pulse interval rate according to the width of the reservoir 0.1-0.4 sec). Only night results are reported. On the same nights, multimesh pelagic gillnets [8] were set in the reservoir. Their original 12 mesh sizes – 5, 6.25, 8, 10, 12.5, 15.5, 19.5, 24, 29, 35, 43, 55 mm knot-to-knot - were extended with 70, 90, 110 and 135 mm meshes to capture all sizes of large fish. The gillnets were set for approximately 12 hours, according to the European Standard [8]. The fish captured in the epipelagic gillnets (depth 0-4.5 m depending on the actual depth of the water) were measured to the nearest 5 mm and weighed. Young-of-the-year (YOY) fish were removed from the data set (see discussion) and the rest created the “Catch basket” in the Sonar 5 processing software [4]. Further processing followed the scheme given in Fig.3.

3. RESULTS AND DISCUSSION
The nonlinear character of \( \sigma_{bs} \) and large signal strength variations between the side and head/tail aspects caused several-fold differences between TSAS and \( \sigma_{bs} \) of the average TS.
(Fig. 1). This meant that the average backscatter energy of the whole 360° circle of fish aspects was significantly higher than corresponded to the average TS. TSAS showed significant linear regressions with the slope similar to other aspects (Fig. 2). TSAS was roughly 5 dB stronger than the average all-aspects regression. The parameters of the TSAS regression for different fish lengths are given in Table 1.

<table>
<thead>
<tr>
<th>Fish length (mm)</th>
<th>a</th>
<th>b</th>
<th>R²</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standard</td>
<td>23.153</td>
<td>-93.407</td>
<td>0.8026</td>
</tr>
<tr>
<td>Fork</td>
<td>23.579</td>
<td>-95.322</td>
<td>0.8104</td>
</tr>
<tr>
<td>Total</td>
<td>23.728</td>
<td>-96.471</td>
<td>0.8401</td>
</tr>
</tbody>
</table>

**TAB 1**: Regressions of TS of average $\sigma_{bs}$ (TSAS), in dB, on fish standard, fork and total length in the horizontal plane, according to the equation $TSAS = a * \log L + b$.

![Fig. 1. The dependence of $\sigma_{bs}$ on body angle (side = 90 and 270°) with the position of TSAS and $\sigma_{bs}$ of average TS indicated by horizontal lines.](image)

The use of $\sigma_{bs}$ for the scaling of echo-integration is a standard procedure in marine vertical surveys where the majority of fish are in large shoals and there is no possibility of estimating the TS of single targets [3]). In the short ranges of horizontal surveys and with relatively regular fish spacing, the detection of single targets and robust TS measurements is not a problem. The major problem starts with the direction of the target in relation to the beam (the difference of TS between the weakest and the strongest echo in a horizontal plane can be in the order of 25 dB [7]). It is possible to solve this problem using the fish aspect, derived from the fish vector, in fixed-location observations [9]. In mobile horizontal surveys some assumptions of fish aspect distribution must be made (Fig. 3). Use of all-aspect regressions for TS to Length conversions gives rather approximate information [6]. In lakes, a random distribution of fish aspects is assumed and handled by the ‘deconvolution’ procedure [10]. The disadvantages of this approach, compared to deriving the $\sigma_{bs}$ from the direct catch, is the need for extensive fish tracking and increasing uncertainty towards the sizing of small fish (‘tail end of deconvolution’). As a result of this, the abundance estimates of horizontal surveys using in-situ targets for echo-integration scaling have very little factual meaning [2]. The assumption of random aspect distribution is violated in narrow parts of lakes [9] and in rivers [10]. It is therefore highly recommended that the common assumption of random fish aspect distribution should be checked by a series of fixed-location observations.
The alternative processing path, using a ‘catch basket’ to create an average $\sigma_{bs}$ using the regressions in Table 1, does not require tracking and provides a well-defined fish population. The TS distribution derived from the ‘catch basket’ has a much reduced variability compared to the in-situ distributions because it does not translate length into a single aspect (TSAS) and does not consider extremes such as side or tail/head aspects (Fig. 4). Fish sizes of interest can thus be selected directly in the ‘catch basket’. The regressions in Table 1 can be used for setting and interpretation of TS thresholds (Fig. 4). In our survey, direct catch data suggested that a length of 8 cm is a good boundary between YOY and older fish (YOY threshold). The TSAS curve crossed the YOY threshold at approximately TS=-49 dB. This would be a noise threshold for the TS readings compensating for the beam position (off-axis position). For uncompensated data, two ways compensation for the beam pattern factor should be allowed. The threshold applied can be directly transferred from the TS (40 log R) domain into sv (20 log R) by using the option of integrating all backscattered energy of the echogram areas with a detected signal within TS domain [4]. This means that areas with a good signal above the threshold (irrespective of single target criteria) is defined in the 40 log R echogram and echo integration of the 20 log R data is done exactly in these areas of the echogram.
Fig. 4. An example of in-situ TS frequency distribution, in the open water of the Orlík Reservoir, compared with the TS frequency distribution of a ‘catch basket’ (length of fish >YOY translated into TS). Standard   TS/length relationships for the strongest (side) and the weakest (tail/head) aspects and TSAS (TS of average $\sigma_{bs}$) are given. Any horizontally recorded fish of a certain length can have any TS between the tail and side aspect. TSAS marks the midpoint aspect with respect to the reflected biomass $\sigma_{bs}$. YOY threshold of 8 cm shows the maximum size of YOY fish. If the noise threshold is higher than the side aspect of 8 cm fish, no YOY fish could be recorded. If the noise threshold corresponds to TSAS for 8 cm fish (-49 dB), then it cuts through the population of recorded targets at a point where the largest YOY fish has more than 50% biomass below the threshold and the smallest fish >YOY has less than 50% of its biomass below the threshold, and vice versa. This ‘overlap’ is due to the strongest aspects of YOY and the weakest aspects of >YOY. If the TS-frequency distribution is flat near the threshold (as in this case), the two ‘overlapping’ errors compensate for each other. The figure also gives the relative expression of integrated biomass assuming the increasing threshold in dB on x-axis [12]. The sv integral with threshold = -70 dB was set as 100%.

Small targets with uncertain association to small fish, least reflective aspects and invertebrates, formed the left hand peak with the highest abundance targets (Fig. 4). Using the TSAS regression to set the threshold left most of these small targets away from the population. The Eckmann threshold echogram [12] shows that by using the threshold to exclude the uncertain small targets we loose less than 10% of all echo energy over the -70 dB threshold. The small targets, despite their enormous densities, represent only a small proportion of the biomass and the loss of this information is tolerable when overall fish biomass and density are what is required (as for the EU Water Framework Directive 2000/60/EC surveys). Using ‘catch baskets’ and TSAS regressions for setting the average $\sigma_{bs}$ and threshold thus efficiently bypasses a laborious tracking analysis and solves the problem of unrealistically high abundances of small targets (Fig. 4).

Obviously, the size composition of the population has a spatial resolution equivalent to the density of direct capture (pelagic gillnets) sampling sites. In our experience, the use of three epipelagic Nordic gillnet series in one site is a good cost-benefit compromise. Fig.4 also suggests that it is not feasible to estimate the density of YOY fish with horizontal beaming. The weakest TS (tail/head aspects) reach values less than -70 dB and recording of such small targets is very rarely achievable during horizontal surveys. Thus it is very rarely possible to
record YOY fish quantitatively and it is much safer to make open water YOY estimates using night fry trawling.

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REFERENCES

DIDSON COUNTING – MANUAL OR AUTOMATIC?

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Abstract: Digital Identification Sonar (DIDSON) using 1.1 to 1.8 MHz was employed to investigate seasonal and diel fish abundance in turbulent deep pools in the Danube River in Austria. Classification of four different fish densities (low, medium, high with single fish and high with fish-shoals) was applied resulting in significant differences in processing time, abundance estimates and fish size measurements achieved from automatic counting compared to the manual counts. The proportional time demand for image counting between automatic techniques and manual counting was 1:40. Automatic counting delivered reasonable abundance estimates at low and medium fish densities and at high densities when single fish were well separated from each other. In dense aggregations (shoals) where fish were packed abreast, automatic techniques tended to overestimate the abundance, whereas manual counting underestimated the abundance due to increased difficulty of visually separating single fish from each other. Automatic sizing overestimated fish length at low to medium densities, and at high densities the length was slightly underestimated compared to manual sizing, possibly due to the beam margin effect. The frequency of outliers (= total length >130 cm) was below 10% of total automatic counts. Manual sizing tended to slightly overestimate the fish length at further ranges.

Keywords: DIDSON; acoustic imaging; manual-counting; automatic-counting;
1 INTRODUCTION

Reliable automated fish counting and sizing by acoustic methodology is a frequent demand for marine and freshwater acousticians. Recent developments in sonar imaging like Dual-frequency IDentification SONar (DIDSON) provide near-video-quality images of fish [1] and remarkably improve the applicability of acoustic methods in acoustic boundary zones like in rivers. Nonetheless, we still lag behind regarding accurate and efficient quantitative and qualitative analyses of acoustic data. Hence, manual counting and sizing – although very time consuming – is still regarded as the reference for automatic estimates. Several analysis approaches have been published based on various acoustic applications including deep to shallow marine environments [2] and freshwater habitats like lakes [3] and rivers [4] [5]. To address this issue, the fish-counting and fish-sizing capability of Dual-frequency Identification SONar (DIDSON) processing software was tested on increasing fish densities using stationary DIDSON recordings in a deep pool habitat of the Danube River.

2 MATERIAL AND METHODS

2.1 Study site and sampling set-up

The data for this study were obtained from a stationary DIDSON recording conducted in a deep pool habitat (Latitude: 48.1483 299° N, Longitude: 016.916 4371° E) of the Austrian Danube River east of Vienna in February 2008. At low water level (water-gauge Hainburg = 1543 m³ s⁻¹) the pool had a maximum depth of 10.1 m and a low circular current velocity of U = 0.01 – 0.10 m s⁻¹. Both water temperature and turbidity were low at 4.2°C and 4.88 ntu, respectively. The bottom consists of gravel. Different fish species utilize this pool throughout the year, in winter predominately as a refuge habitat. The DIDSON was mounted on a mechanical pan and tilt unit, which was fixed on the anchored research boat. Sampling settings for the DIDSON included a 40° tilt downwards from the horizontal plane, low frequency mode (1.1 MHz), a 0.85 m start range, a 20.01 m window length, a 10.56 m focus, a 36 dB receiver gain and a frame rate of 8 frames s⁻¹. At an operating frequency of 1.1 MHz the 20-m window length has a down-range pixel size of 4 cm and a cross-range resolution of 20 cm.

Fig. 1: DIDSON images representing four subsets of data investigated by manual and automatic counting with low fish density (a), medium fish density (b), high fish density of single fish (c) and high fish density of fish-shoal recorded in a deep pool of the Danube River.
2.2 Test data

In order to test the DIDSON manual and automatic processing methods at increasing fish densities, subsets (10 minutes each) of four different fish density recordings, including low fish density, medium fish density, high fish density with single fish and high fish density with fish-shoaling, were selected. Regarding abundance, a low fish density corresponded to an average of 2 individuals min\(^{-1}\), a medium fish density to an average of 17 individuals min\(^{-1}\) and high densities to an average of 35 individuals min\(^{-1}\). Additionally, we distinguished between high fish densities where fish were well recognizable as individual fish and high fish densities where fish were packed in dense fish-shoals making clear visual separation between individuals difficult or impossible (Fig. 1a, b, c, d).

2.3 DIDSON manual and automatic processing

DIDSON processing software V5.20 offers several ways to count and size fish. Manual counting and sizing was comparatively tested side by side with three different settings of automatic echogram counting and sizing. This manual approach was conducted in the DIDSON viewer during playback mode by marking and sizing each single fish. The playback mode was frozen when a frame showed a clear shape of the investigated fish in the central beam. The fish was zoomed in and counted by marking it with a single left mouse click. The fish length was calculated by clicking on snout and tail, and the data were stored for further analysis in a separate file. Echogram counting and sizing was conducted in the DIDSON echogram mode. The range was restricted to 18.2 m in order to avoid implementing static bottom echoes or rocking bottom into the processing. Echogram counting and sizing was conducted in three different ways separately for each dataset. First we applied the default settings for echogram auto-counting, which uses the central beam for processing (EC); second, we enlarged the effective beamwidth for echogram detection by increasing the number of processed beams to 16 (EC 16 beams); third, we activated the clustergram feature for processing (EC cluster). Clusters are a processed collection of pixels that touch each other. For the processing parameters we applied 12 for the minimum track size. This represents the minimum number of samples per cluster for counting. The minimum threshold for image processing was set to 6 db, the minimum cluster area to 200 cm\(^2\) and the maximum number of fish per frame to 32. The clustergrams are formed (1) when background subtraction was performed, which means that the static components of the image are subtracted out, leaving only the moving pixels, (2) by the remaining pixels that move in a regular fashion from frame to frame rather than jump randomly around (generally caused by noise), (3) when all pixels are set to zero that are less than a selected threshold and (4) when the image is convolved with a rectangular kernel b beams wide by s samples in range. For further details, refer to the DIDSON Operation Manual V5.20 edited by Sound Metrics Corporation and the DIDSON Fish Assessment tutorials at www.soundmetrics.com.

3 RESULTS

Figure 2a shows the abundance estimates (mean ± standard deviation; n = 10) of the four different fish density data-sets derived from four different counting methods. At low fish density, manual counting (MC) yielded an abundance estimate of 1.8 ± 1.5 individuals min\(^{-1}\), echogram counting using the central beam (EC) yielded an estimate of 1.0 ± 0.9 individuals min\(^{-1}\), echogram counting with enlarged beamwidth using 16 beams (EC 16 beams) yielded...
1.8 ± 1.5 individuals min⁻¹, and echogram counting with activated clustergram mode (EC cluster) yielded 1.6 ± 1.8 individuals min⁻¹. There was no significant difference between the manual counts and the automatic counts at low fish density (t-test, p > 0.05). At medium fish density, MC yielded an abundance estimate of 16.6 ± 14.0 individuals min⁻¹, EC 6.8 ± 6.4 individuals min⁻¹, EC 16 beams 12.3 ± 9.4 individuals min⁻¹ and EC cluster 7.2 ± 5.1 individuals min⁻¹. There was no significant difference in the abundance estimates between MC and EC 16 beams (t-test, p > 0.05), but there were significant differences between MC and EC as well as between MC and EC cluster (t-test, p < 0.05). At high fish density with single fish, MC delivered an abundance estimate of 35.7 ± 28.7 individuals min⁻¹, EC 25.0 ± 22.0 individuals min⁻¹, EC 16 beams 37.6 ± 18.7 individuals min⁻¹ and EC cluster 16.2 ± 6.2 individuals min⁻¹. There was no significant difference between MC and EC 16 beams (t-test, p > 0.05), but there were significant differences between MC and EC as well as between MC and EC cluster (t-test, p < 0.05). At high fish density with fish-shoaling, MC yielded 33.0 ± 20.5 individuals min⁻¹, EC 39.3 ± 29.3 individuals min⁻¹, EC 16 beams 53.7 ± 28.7 individuals min⁻¹ and EC cluster 17.8 ± 8.5 individuals min⁻¹. There was no significant difference between MC and EC (t-test, p > 0.05), but there were significant differences between MC and EC 16 beams as well as between MC and EC cluster (t-test, p < 0.05). Figure 2b shows the total length (TL) estimates (boxplots including 5th/95th percentile) of the four different fish density data-sets derived from four different counting methods. At low fish density, manual counting (MC) delivered a mean TL ± standard deviation (SD) of 43.7 ± 15.6 cm, echogram counting using the central beam (EC) a mean value of 49.6 ± 16.1 cm, echogram counting with enlarged beamwidth using 16 beams (EC 16 beams) a mean of 48.5 ± 18.9 cm and echogram counting with activated clustergram mode (EC cluster) a mean of 59.3 ± 22.3 cm. At low fish density there was no significant difference in the TL estimates between MC and EC as well as between MC and EC 16 beams (t-test, p > 0.05), but there was a significant difference between MC and EC cluster (t-test, p < 0.05). At medium fish density, MC delivered a mean TL ± SD of 45.2 ± 15.1 cm, EC 76.1 ± 22.5 cm, EC 16 beams 59.0 ± 26.4 cm and EC cluster 64.4 ± 23.2 cm. At medium fish density there was highly significant difference in the TL estimates between manual counting and automatic counting methods (t-test, p < 0.01). At high fish density with single fish, MC delivered a mean TL ± SD of 55.6 ± 14.4 cm, EC 73.0 ± 29.4 cm, EC 16 beams 50.2 ± 26.0 cm and EC cluster 56.7 ± 24.1 cm. At high fish density with single fish there was no significant difference in the TL estimates between MC and EC cluster (t-test, p > 0.05), but there were highly significant differences between MC and EC as well as between MC and EC 16 beams (t-test, p < 0.01). At high fish density with fish-shoaling, MC delivered a mean TL ± SD of 50.8 ± 12.3 cm, EC 69.8 ± 30.7 cm, EC 16 beams 53.8 ± 33.1 cm and EC cluster 50.4 ± 26.4 cm. At high fish density with fish-shoaling there was no significant difference in the TL estimates between MC and EC 16 beams as well as between MC and EC cluster (t-test, p > 0.05), but there was a highly significant difference between MC and EC (t-test, p < 0.01).

In the total length estimates the proportion of outliers (= fish with a total length > 130 cm) was 0% in manual counting, 5.4% in echogram counting using clustergrams, 7.3% in echogram counting using 16 beams and 18.4% in echogram counting using one central beam.
4 DISCUSSION

Our study revealed that in manual as well as in automatic DIDSON fish counting and sizing at a low frequency of 1.1 MHz, the accuracy of the estimates strongly depends on fish density. Manual counting and sizing delivered reasonable results during low, medium and high fish densities, when single fish were well separated from each other, but it was very time consuming. The ratio of time demand between manual and automatic counting was 40 : 1. In very dense aggregations (fish-shoals), manual counting and sizing is biased due to the inability to visually distinguish individual fish, probably yielding lower abundance estimates. Manual sizing provided reasonable fish length estimates. Nonetheless, at low frequency mode, when fish shape (especially at further range and in dense aggregations) becomes blurred, accurate manual sizing became difficult and presumably tended to oversize fish; this approach apparently delivers more precise fish length in high frequency mode due to better range resolution [6].

Automatic counting worked fast, and especially echogram counting with enlarged beamwidth by 16 beams delivered good abundance estimates in low, medium and high fish densities with well separated individual fish that swam continuously in one direction and that were of sufficient size regarding range resolution. In densely packed fish aggregations, where individuals can hardly be separated, it seems advantageous to restrict automatic counting to the central beam in order not to overestimate abundance due enlarged beamwidth, as observed in automated counting by Maxwell and Gove [7]. In automatic sizing, the variability and proportion of outliers was partly acceptable. Auto-sizing also works fast but
needs cross-checking by visual approval in the DIDSON viewer, which is time consuming. Overestimations in fish sizes, sometimes dramatic, could have been caused due to connecting errors as proposed by Handegard & Williams [2]. Beam margin effects might have underestimated the fish sizes during echogram counting with enlarged beamwidth of 16 beams at high fish densities. Likewise, the appropriate length estimates during application of clustergrams at high fish densities seemed to be an artefact due to splitting of closely packed fish traces into several smaller portions, thus lowering the average fish length of the total dataset. Handegard & Williams [2] termed similar observations as splitting error.

Recommendations to improve manual and automatic fish counting and sizing in DIDSON recordings include the development of (a) a standard procedure for manual fish sizing, (b) new tools in image analysis to enhance the marks of marked fish for more than one frame, (c) correction factors for the length deviance in the outer part of the beam and (d) echo integration in DIDSON recordings, comparable to an average fish cluster area, in order to estimate the biomass of fish-shoals. In this respect, the newly developed Cubic Cross filter detector in Sonar5-Pro seems to be promising and remains to be tested on the entire data-set [8].

5 ACKNOWLEDGEMENTS

I want to thank the Austrian Water Authority (Schifffahrtssaufsicht Hainburg), the via-donau, Mike Sawkins at Macartney A/S in Denmark and Bernhard Berger for their support during fieldwork. The project was funded by the Austrian Federal Ministry of Infrastructure as part of the ‘Flussbauliches Gesamtkonzept - Naturversuch Bad Deutsch Altenburg’.

REFERENCES

Structured Session 27

Underwater Noise Measurement and Mitigation

Organizers: Stephen Robinson & Paul Lepper
MEASUREMENTS OF ANTHROPOGENIC AND NATURAL AMBIENT NOISE IN THE CONTEXT OF EFFECTS ON MARINE ANIMALS

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Abstract: There is considerable interest in the effects of noise of human activities on marine animals and this includes the anthropogenic contribution to the ambient noise. In particular, the contribution from distant shipping, known as traffic noise, dominates the low frequency part of the ambient noise spectrum (below about 200 Hz) in many parts of the world, particularly around North America and Europe where most measurements have been made. To assess the effect of this anthropogenic component requires a characterisation of the range of ambient noise from both natural and anthropogenic sources. Such characterisation also provides a context for assessing the impact of anthropogenic noise in general. Measurements of ambient noise have been made in waters around Australia using bottom moored systems set to sample the ambient noise at regular intervals. Low shipping densities in some areas near Australia allow the natural components of ambient noise to be measured at frequencies dominated by traffic noise in many other regions. The results show that natural low frequency ambient noise at times reaches levels comparable to or higher than high levels of traffic noise. The main components of this high level ambient noise are wind dependent noise from breaking waves and the noise of biological choruses that result when large numbers of animals are calling. Marine animals have, therefore, been subject to noise levels from natural sources that are as high as those of traffic noise in areas of high shipping densities, though not continuously as is the case for traffic noise.

Keywords: Ambient noise, marine animals, anthropogenic noise, shipping noise.
1. INTRODUCTION

The interest in the effects of noise of human activities on marine animals extends from individual high level sources of sound, such as seismic airguns used in surveys of the sea floor and sonar used for many purposes from naval defence to fish finding, to lower level, more pervasive and widespread sources such as distant shipping. In assessing the impact of any anthropogenic sound, it must be placed in the context of the background or ambient noise of the ocean, since this forms the basic limitation on the use of sound by marine animals. Natural ambient noise varies typically by 20 dB or so over relatively short time scales and variations of 30 dB occur. Since marine animals have evolved in this environment it is reasonable to assume that they cope adequately with noise levels over the range of natural ambient noise. Assessments of biological significance of the anthropogenic noise exposure will require a comparison with the range of ambient noise. For example, masking effects of anthropogenic noise may be compared with masking by ambient noise. The typical variation in ambient noise levels of about 20 dB would cause a variation of around factor of ten in detection ranges (assuming square law propagation loss). Actual variation in detection range will be even higher, because it is also affected by varying propagation loss. Masking by anthropogenic noise needs to be assessed within this context.

Measurements of anthropogenic noise, require the anthropogenic noise to be separated from the background noise, and assessment of potential impacts on marine animals requires a comparison of the range of anthropogenic noise with the range of ambient noise. Ambient noise has many components that are natural, but there is now a significant component of that is anthropogenic: traffic noise, the noise of distant shipping. Traffic noise was defined by Wenz [1] as the cumulative effect of the contributions from all ships across an ocean basin, excluding the noise from ships that are close enough to be detected as individual ships. Traffic noise dominates the low frequency part of the ambient noise spectrum (below about 200 Hz) in many parts of the world, particularly around North America and Europe and there is interest in determining whether this is having a significant impact on marine animals. There is also concern that traffic noise is increasing, and there is some, though limited data supporting this view [2].

In its review of ocean noise and marine mammals, the National Research Council of the USA identified the importance of the ambient background noise in assessing the impact of anthropogenic noise, and recommended long term monitoring and modelling both ambient noise and noise from identifiable anthropogenic sources, and the effect on marine mammals [3].

This paper examines some of the issues involved in assessing the impacts of anthropogenic noise on marine animals, particularly those related to the effects of ambient noise, and the context of the typical acoustic environment of marine animals. It draws on measurements of ambient noise around Australia where anthropogenic sources contribute less than in many areas where ambient noise has been studied.
2. CHARACTERISTICS OF AMBIENT NOISE

Ambient noise is the background noise from many different sources, and usually, definitions exclude the contributions from individually identifiable transient sources such as a whale calling or the sound of a passing ship. Although it is a complex combination of sounds, ambient noise can usually be expressed in terms of relatively small number of components, each produced by the contributions of a particular type of source. The most effective way of characterising, predicting or forecasting the noise is to characterise the behaviour of the individual components, each showing temporal and spatial variation related to behaviour of the sources. This is necessary to understand the acoustic environment of marine animals and to provide the context for effects of exposure to anthropogenic noise.

The main components of ambient noise were established in early studies of ambient noise [1, 4]: (a) sea surface noise: the noise of wind and wave action at the surface, usually referred to as wind-dependent noise following Wenz [1], and rain noise; (b) biological noise, the noise of fish, whales and invertebrates; and (c) traffic noise, the noise of distant shipping. These may be considered to the prevailing components, i.e. the ones that are usually present, though several other components may be evident from time to time, and there may be times or locations where one or other of these three main components are insignificant. The importance of the biological component of the ambient noise has often been neglected and it is often missing from noise prediction curves. Biological noise is usually considered in terms of individual transient sounds but it provides a major component of ambient noise when many individuals of a species are calling and the calls merge into a continuous noise known as a chorus [5 – 8]. Choruses often reach levels of around 20 dB above lower levels of background noise and may be evident for several kilometres from a chorus, with even higher levels within the area of calling animals [6, 8].

The noise from breaking waves is usually referred to as wind-dependent noise because the noise levels correlate better with wind speed than with any measure of the surface waves [9]. A very good correlation of noise level with wind speed may be observed when other components of noise are low and the wind speed is sensed in the vicinity of the noise recording system [10].

Traffic noise involves so many distant sources that the noise itself is not readily evident as due to distant shipping, apart from the general spectral slope that is similar to the typical slope of ship noise [1]. Consequently, separating traffic noise from other components of ambient noise is not straightforward.

Characterising the ambient noise at any location or for any area requires the separation of the noise into its components, and the determination of the behaviour of these components. One approach is to measure each component when the contribution of the others is minimal, e.g. traffic noise when wind speeds are low and biological noise is not evident, wind-dependent noise in areas of low traffic noise and low biological noise. Biological noise is usually more easily recognised than the other components because the individual sounds are characteristic.

Sea noise prediction methods are usually expressed in terms of components and the most common form appears to be based on those of Wenz [1] which were obtained from measurements around North America. These do not show wind-dependent noise below 100 Hz because of the difficulty in separating this from traffic noise in the areas of measurement where traffic noise is high.
3. MEASUREMENTS AND ANALYSIS

Measurements of ambient noise have been made in a number of areas around Australia using bottom moored systems set to sample the noise at regular intervals. Usually an anemometer is moored on a buoy in the vicinity to provide estimates of the wind speed. Considerable care was taken in the design of these systems to avoid contamination by noise from the recording system and its moorings or local pressure fluctuations from turbulence due to interaction of the water flow with the measuring system. Traffic noise varies widely around Australia [11] and some measuring locations were chosen to be in areas of low traffic noise to allow low frequency wind-dependent noise to be measured. Biological noise is usually easily recognised by the characteristics of the individual transient sounds. The individual sounds may merge during choruses that result when large number of animals are calling, but the individual sounds can be distinguished early or late in the chorus when fewer animals are calling.

Ambient noise measurements require an absence of identifiable individual sources such as animal calls or the sound of a passing ship. The dependence of noise on wind speed was determined by fitting regression lines to noise level $N$ as a function of logarithm of wind speed $u$ to obtain a relationship of the form of Equation 1, where $A$ is a constant. This determines a relationship of noise mean square pressure proportional to wind speed to the power $n$. In areas where there other components of noise are very low it has been possible to obtain results where the noise is almost entirely wind-dependent over a wide range of wind speeds, and these show values of $n$ approaching 3.

$$N = A + 10n \log_{10} u$$  \hspace{1cm} (1)

Traffic noise levels were determined in the absence of biological noise, as the residual noise at low wind speeds that shows no wind-dependence. In fact, this component contains all residual noise from distant sources as is probably the case in other measurements of traffic noise, though studies have shown that the noise levels tend to vary generally in accordance with variations in shipping densities and propagation conditions [11].

4. AMBIENT NOISE AND MARINE ANIMALS

Figure 1 shows a summary of ambient noise from measurements around Australia. Traffic noise in a particular area varies ± 5 dB around the averages shown and varies over a wide range in different areas. The wind-dependent noise curves of Figure 1, measured in areas of low traffic noise, show that noise levels rise with decreasing frequency below 100 – 200 Hz, while still showing evidence of the broad peak at around 500 Hz evident in the conventional prediction methods such as the curves of Wenz [1]. These results are consistent with the few results in early and recent work in areas around North America where traffic noise was less significant than usual [e.g. 12]. Wenz also noted the wind dependence at low wind speeds and some of the data presented show similar spectral shapes to those of Figure 1. At 30 knots the low frequency wind-dependent noise levels in Figure 1 are comparable to upper levels of
“usual traffic noise” given by Wenz. Some areas, however, may show higher levels of traffic noise, as observed, for example, in recent measurements off California [2].

Fig. 1: Summary of ambient noise for the Australia region, showing the wide range of traffic noise levels, some examples of biological choruses and wind-dependent noise extending to low frequencies.

Biological choruses rise to levels that exceed both wind-dependent noise and traffic noise in Fig. 1 over most of the frequency range 50 Hz to 3 kHz. The example shown of a fish chorus is comparable in level above 50 Hz to the highest levels of traffic noise observed elsewhere. Overall, there is substantial variation in natural ambient noise levels, providing a dynamic acoustic environment that marine animals have evolved with. It is reasonable to assume that they cope adequately with this in their use of sound. In this context, traffic noise is another variable component that generally lies within the range of the variations in level of natural components of ambient noise. Marine animals have, therefore, been subject to noise levels from natural sources that are as high as those of traffic noise in areas of high shipping densities, though not continuously as is the case for traffic noise. In many parts of the world, shipping densities are much lower than in the high traffic areas around North America and Europe, and marine animals would be exposed to much lower levels of traffic noise, as is the case for many areas in regions near Australia.

REFERENCES


SOUND POWER LEVELS OF DEVICES IN AIR AND WATER

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Abstract: The power output in watts (W), and acoustic intensity in W/m² provide alternatives to measurements made as acoustic pressures. For a source, the total output power provides a single value (or single spectrum) covering all directions of emission.

EU directive 2000/14/EC specifies the measurement and declaration of dB LWA for a wide range of outdoor machinery, radiating sound into air. This has increased the perception of airborne noise for buyers of competitive products, using a single figure specification. This helps drive noise levels down. These benefits could be extended to the underwater sector.

Underwater sources include ships and ROVs (remotely operated vehicles). Measurements of work class ROVs with powerful hydraulic systems have shown that pump noise is often dominant. Very large (1500HP) subsea pumps are being used for oil and gas extraction. To further investigate the scope for sound power measurements a small underwater pump has been tested in both air and water.

The underwater tests have used a reverberant tank system which can give sound power levels quickly with a single hydrophone. The benefits and limitations of this technique are discussed. Comparisons with the airborne measurements are continuing.

The conversion of a sound power spectrum to a single band integral requires a choice of bandwidth. This may not be practicable in the general case, but some sources can be modelled to achieve this useful summary of the data.

Keywords: Sound power levels, underwater ship noise, machinery noise.
INTRODUCTION

Measurement of sound both in air and in water has been dominated by the measurement of the acoustic pressure. This is principally due to the availability of accurate and reliable pressure sensors, the microphones or hydrophones. But for many purposes it is the sound power which is the more significant parameter.

In some medical applications it is the heating of tissue which is considered significant, and the key parameter is then the incident time-averaged intensity at a point in the tissue in watts/m². Temporary threshold shifts (TTS) in hearing sensitivity are now believed to correlate with sound energy levels (SEL). The detection of sound will also be in part determined by the background noise energy. For systems optimised for both air and water, such as pinniped hearing, energy considerations are of interest.

Here consideration is given to the emission of sound power by sources capable of operation in both air and water. These ideas are founded on some experience with ROV noise. These underwater vehicles can be tested in both air and water, with significant correlations. The noise emissions from their hydraulic pumps often dominates in both conditions and testing of representative pumps could thus provide some enlightenment.

Tests in both environments can use the highly reverberant fields of tanks and chambers, procedures for the latter being set out in standards such as ISO 3741. The $L_{WA}$ sound power rating, originally based on such tests is now required to be specified on a wide range of machinery sold in the EU. As specified, ‘sound power level $L_{WA}$’ means the A-weighted sound power level in dB in relation to $1 \mu W$ as defined in EN ISO3744:1995 and EN ISO 3746:1995. [1] The competition thus induced is likely to give a reduction in noise output, eventually reducing background underwater environmental noise.

So perhaps there is a place for an underwater equivalent ($L_{WL}$)

THE DETERMINATION OF POWER

This is most accurately derived from measured sound pressure level data sets, using the known properties of the medium to make the conversion.

$$I = \frac{W}{A} = \frac{P^2}{\rho c}$$

$P$, the sound pressure level, is measured in Pascals, the S.I unit, or related units ($\mu Pa$).
$I$, the intensity, is the power, W, flowing through a unit area $A$ in watts/m².
$\rho c$ is the acoustic impedance, the product of density $\rho$ and sound speed $c$.

This relationship is valid for a plane wave, where the energy flows in one direction, but becomes less accurate if there is significant reverberation, when echoes provide energy flow in many directions. For this reason, most acoustic test sites will endeavour to
minimise reverberation, especially if the aim is to characterise a specific source. However,
directional sources will require a large data set.

**USING FREE FIELD PRESSURE DATA TO CALCULATE POWER**

A free field, with no reverberation, is best. In practice, measurements made of a source
located by a single reflecting plane can be used, and indeed these are favoured for accurate
measurements of air borne noise where a hard ground plane provides both the reflector
and the support for the machinery being tested. For ship noise test standards such as those
currently under development by ANSI working group WG47, the sea surface also provides
a good reflector when calm.

The measured sound pressure level $P$ at a range $r$ can be presented as a source level, a
property of the source rather than the environment. In the far field, this considers the
energy to emanate from a point, the “acoustic centre”. It can then be used in suitable
environmental propagation models to predict the sound pressure in more complex,
reverberant and absorbent, conditions such as shallow waters.

Avoiding decibels for clarity, a useful source characteristic, in a free far field condition, is
the source output $S$.

$$ S = P \cdot r \quad \text{Pa} \cdot \text{m} $$

This product of pressure and range remains constant as the energy spreads out over an
increasingly large sphere. Conversion to decibels gives the source level $SL$

$$ SL = 20 \log S = 20 \log P + 20 \log r = PL + 20 \log r $$

This “core” sonar equation links the source level to pressure level in a free far field.

Source output $S$ is a directional parameter, a vector, which gives $W_\theta$, the energy flow in
the specified direction $\Theta$ when converted by use of the specific impedance $\rho \cdot c$.

$$ W_\theta = \frac{S^2}{\rho \cdot c} \quad \text{W/sr} $$

The units are watts per steradian. Solid angles measured in steradians are 3D analogues of
angles measured in radians, given by the area $A$ (m²), through which the sound power
passes, divided by $r^2$. For an omnidirectional source, with a single value of $W_\theta$, the total
power is obtained by multiplying by the $4\pi$ steradians which cover all directions from the
point source.

**THE VARIATIONS WITH FREQUENCY**

A major simplification available to acousticians working in air is the assumption that it is
only human beings which need to be protected. They have therefore used the “A”
weighting, as representative of human hearing sensitivity. Of course the circumstances underwater are very different, with a need to consider the huge disparity of hearing sensitivity over many species.

It is, on reflection, rather odd that the $L_{WA}$ rating, which is otherwise a specification of the source, independent of the environment, includes the characteristics of a particular receiver. But this allows the data to be compressed to a single decibel level which is necessary for a simple specification to facilitate comparisons. There are various schemes proposed which give a variety of weightings for underwater species, either by groups (M weighting)[2] or by individual species (dBht) [3]. But if these were included in an underwater $L_{WU}$ it would dramatically increase the data set size. It is simpler to provide a single spectrum over all measured frequencies.

A source wattage with no receiver weighting provides a more logical division of the predictive stages in an EIA (environmental impact assessment). It also means that the dB reference level can be truly watts (or picowatts as in $L_{WA}$). However, this then requires a measurement by hydrophones calibrated over all frequencies. In reality bounds would be needed to form a band integral with flat top weighting. Decisions on the limits may be driven by a variety of issues, including a lack of receiver sensitivity, or minimal source output power.

For many applications a power spectrum can be used, plotting the power per unit frequency band against the centre frequencies. These can be 1 Hz constant frequency bands (W/Hz), or others such as 1/3rd octave bands, which are more representative of typical critical masking bands for many creatures, including humans.

THE USE OF A SOURCE MODEL

A source model can avoid the need for arbitrary band limits if a particular class of source exhibits adequate similarity of spectral distributions. This is the aim of the “red/white” model appropriate for shipping noise.

![Fig 1 A “red & white” noise model for a ship’s underwater noise emission](image-url)
The high frequencies are dominated by cavitation noise which falls rapidly, but predictably, with frequency giving a $1/f^2$ reduction of outputs in successive 1Hz bands. Urick [4] showed how this model gives a finite integral over all frequencies without the need for arbitrary limits. This model is fully specified by the peak frequency and level at which the red HF noise model abuts a white LF noise model. The model of Fig 1 used in recent work discussed by Theobald et al [5], is based on data measured at AUTEC on a large bulk carrier “Overseas Harriette”[6], and an interpretation by Hazelwood [7]. It has ignored much detail, but provides simple data suitable for environmental modelling work, where the detailed variations in the source cannot be covered.

However, field evidence from workclass ROVs, dominated by their hydraulic pump noise, shows a spectrum extending to higher frequencies, for which this model is inapplicable. This pump noise has been found to be highly stable and predictable, especially when the hydraulic power is not being used, a frequent condition. Such pumps can be quite compact and suitable for testing onshore.

THE ANALYSIS OF REVERBERANT DATA MEASUREMENTS

A simpler alternative to a pressure polar data set is to measure the sound power in a highly reverberant environment such as a tank. The earlier airborne work using a reverberant chamber provided the basis for ISO 3741 data measurement. However, there are limitations to the frequencies suitable for these techniques.

Broadband noise energy from a compact source such as a pump can be divided into the initial direct field and the reverberant field due to the multiple reflections off the walls. For test purposes, a white noise voltage source feeds a suitable transducer.

The distribution of wave directions in the reverberant field means that the wave pressures are incoherent, and that their energies are additive. The mean square pressures associated with the direct $P_d^2$ and reverberant $P_r^2$ pressure fields can thus be added to give the total $P_t^2$ (Kinsler et al [8], pp 314-326). For steady state circumstances the power of the source $W$ equals the losses at the walls (ignoring absorption in the water). The energy density $E$ in a uniform reverberant field is given by

$$ E = \frac{P_r^2}{\rho c^2} $$

and the rate of loss for a tank with an absorbent wall area of $A$ (m²) can be shown to be

$$ \frac{E c A}{4} = \frac{P_t^2}{4 \rho c} = W $$

The total field $P_t^2 = P_d^2 + P_r^2$ can now be related to source power $W$

$$ P_t^2 = W \rho c \left( \frac{D_f}{4 \pi R^2} + \frac{4}{A} \right) = S_p^2 \left( \frac{1}{R^2} + \frac{16 \pi}{A D_f} \right) $$

or more simply

$$ P_t^2 = S_p^2 \left( \frac{1}{R^2} + \frac{1}{R_c^2} \right) \quad \text{if} \quad R_c^2 = \frac{A}{16 \pi} \quad \text{and} \quad D_f = 1$$
This relationship can be used to analyse the results of measurements of the distribution of total pressure $P$ with range from the source $R$. The equivalent range $R_e$ measures the tank absorption, the range at which the two terms are equal if the source is omnidirectional ($D_f = 1$). To measure $P_r$, an omnidirectional hydrophone will be required, with the same sensitivity to the direct and reverberant pressure fields.

Fig 2 shows the plot taken from this work, with the tank characteristic analysis based on plotting $P^2$ (Pa²) against the inverse square range (1/m²), to give a straight line with a gradient which measures source power. The reverberant field is found to be insensitive to position or alignment and its value used to calibrate the tank absorption.

The relationships were developed by Hazelwood and Robinson [9], and the band analysis automated by Hayman. The airborne equivalent reported by Sabine in 1922 (see Kinsler & Frey [8]) uses the reverberation time, but this is shorter and harder to measure in water.

Fig 2 is for the NPL 5.5 m diameter wooden tank, which was found to be sufficiently reverberant for the technique to be successful. However, frequencies below 2.5kHz were found to have insufficient modal density, whilst the Sabine loss level rose above 16 kHz. However this reverberant correction was 16dB +/- 2dB over 9 adjacent 1/3rd octave bands (covering 2.23 kHz – 17.8 kHz)

### REVERBERANT TANK CALIBRATION

<table>
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<tr>
<th>Range R m</th>
<th>1/R²</th>
<th>Spec noise dB/V</th>
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- **Intercept**: 14.101376 Pa²
- **Gradient**: 7.673306 Pa²/m²
- **Noise SL**: 128.8 dB/uPa.m

### REVERBERANT TANK PLOT

![Graph showing the relationship between range and pressure squared](image)

**y = 7.6733x + 14.101**

**R² = 0.9961**

**Fig 2** Extraction of direct field data from measurements made in a reverberant tank

The key test is the uniformity of the reverberant field over an adequate area of the tank (away from the source and the edges). Reverberant field data as shown varies by less than 1dB at higher frequencies where the tank modal density is high.

An alternative to a tank calibration, if a compact source of known power if available, is to compare the source with the standard in the reverberant field.

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SOME RESULTS FOR INDUSTRIAL EQUIPMENT

Fig. 3 below shows the power voltage spectrum of a small hydraulic pump suitable for a small ROV, providing a 140 bar oil pressure, used to power its manipulator tool. Here this is being constrained by its safety valve with no useful work being done. A notable feature is the extreme tonality and harmonics of what might be described as “oil hammer” noise. This will contribute to errors in the reverberant averaging process. The fundamental at 2.4 kHz is thought to represent the pressure pulse rate from the pump, with a sharp onset contributing to the “comb” spectrum. The pump was measured in a reverberant field within the wooden NPL tank of Fig 2, that is, with the pump more than 2 m from the hydrophone. The results are not sensitive to geometry or alignment so can be carried out quickly.

![40ms time series #68, Hann window FFT](image)

Fig 3 FFT analysis of a reverberant pressure field generated by a pump

Although the total power is a simple concept, the errors in assessing the tonal contributions will require careful consideration. The integration of measurements over broader octave bands such as 5-10 kHz help, with values around -53 dB//V/Hz being typical of the data shown. The voltage spectrum is measured for a Reson 4033 hydrophone. With a hydrophone sensitivity of -203 dB//V/μPa for this band, this gives a reverberant pressure field of 150 dB//μPa in band. The reverberant correction of 16 dB//m then gives a source level of 166 dB//μPa·m. This is about 0.3 watt noise output in this octave band, compared with the electrical input of 400 watt.
CONCLUSIONS

The benefits of a power $L_{WA}$ rating for the presentation of data to a wider audience has been demonstrated by its use to characterise airborne equipment, and consideration should be given to its use underwater.

The reverberant tank technique has advantages of speed but is limited in accuracy and frequency range, but it has been used to good effect in specific applications. Further work in the laboratory and in larger test sites is needed to further explore these issues.

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CUMULATIVE NOISE EXPOSURE ASSESSMENT FOR MARINE MAMMALS USING SOUND EXPOSURE LEVEL AS A METRIC

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\textbf{Abstract:} Sound exposure level (SEL) has been suggested as an important parameter when considering the impact of anthropogenic noise on marine mammals by the US Marine Mammal Criteria Group of the National Marine Fisheries Service (NMFS). This metric allows the cumulative exposure of an animal to a sound field for an extended period to be assessed against a predefined threshold for injury criteria. In many cases, this can become the dominant mechanism by which a marine mammal may suffer injury.

In this paper, the SEL metric has been used to estimate the cumulative exposure of animals using two examples of sound sources: (i) a large tanker vessel; and (ii) sub-sea marine impact piling. The method has been used in a predictive manner using models for noise sources, and also using measured data of the noise radiated by the source. The method can be used to investigate the effect on the cumulative exposure of factors such as source level, transmission loss model and animal behaviour.

\textbf{Keywords:} Sound exposure level, cumulative exposure, injury criteria

1. INTRODUCTION

The impact of anthropogenic noise on marine life is of growing importance with the increasing number of anthropogenic activities in the oceans \cite{1}, particularly the increasing number of installations of off-shore wind-farms \cite{2}. Whist the impact on fish species is often an economic concern, the impact on marine mammals is a concern because of the potential to cause injury or disrupt the natural habitat of the species. Disruption of disturbance can be
governed by a number of factors, including the annoyance of the sound, auditory masking and the activity in which the animal is partaking. The annoyance of the sound can be comprised of a number of contributions including the received level (sound pressure level), the temporal and frequency characteristics of the source, the acoustic environment and the duration of the radiating activity. However, injury is simply defined as the on-set of auditory permanent threshold shift (PTS) [3], which is governed by either a peak level or an exposure level; sound pressure level (SPL) and sound exposure level (SEL) respectively. The fundamental difference between these two parameters is that SPL can be an instantaneous value and SEL is the total noise energy to which the mammal is exposed during a given duration – 1 second is typical or the pulse duration for impulsive source. Due to the very high level required to give rise to instantaneous PTS [3], the likelihood of PTS occurring due to a prolonged exposure is far more likely. For the case where a sound source is of a prolonged nature, for vessel activity, marine piling etc., cumulative exposure can also be a useful parameter. This considers a summation of the SEL’s to which the animal is exposed.

Cumulative exposure can be considered important for two scenarios; i) an animal close to a sound source might be startled by a high SPL, which is not high enough to cause instantaneous PTS, and proceeds to swim away at a given speed. The time taken for the animal to evade the exposed area is sufficient for the cumulative exposure to exceed that required to cause the on-set of PTS; ii) an animal traversing or entering an area of acoustic activity at a given distance and swim speed might be exposed to an SPL which is not sufficient enough to cause any avoidance behaviour by the animal. The time taken for the animal to traverse the area of acoustic activity might be sufficient for the cumulated exposure to exceed that required for the on-set of PTS.

This paper considers two hypothetical case studies for the use of SEL and cumulative exposure for a marine mammal traversing an area of acoustic activity (a large tanker vessel and sub-sea piling event). The use of a swim-by model enables the closest distance of approach possible without exceeding the SEL criteria for injury to be calculated, or inversely enables the cumulative exposure to be calculated for an animal traversing at a set distance.

2. SOUND EXPOSURE LEVEL AND CUMULATIVE EXPOSURE

Although SPL is routinely used for both the hearing threshold and receive level, other parameters are often used in the literature for impulsive sources such as marine piling [4]. The reason for this is that the SPL parameter is based on the root mean square of pressure and is therefore not strictly suitable for impulsive signals. Other researchers in the literature have suggested the use of peak pressure (dB peak) to provide a more accurate description of the generated acoustic pressure [5]. For a symmetric waveform this is half the value of the peak-to-peak amplitude. However, the waveforms encountered in piling noise measurements can sometimes exhibit significant asymmetry, and so the peak-to-peak values have been used [6][7].

Given these problems with using SPL for impulsive signals, calculating the energy in the pulse and expressing it as an SEL is more appropriate for sources such as impact piling events. The SEL for such an event is calculated by integrating the square of the pressure waveform over the duration of the pulse. The duration of the pulse is defined as the region of the waveform containing the central 90% of the energy of the pulse. The calculation is given by:
The value is then expressed in dB re 1 μPa²·s and is calculated from:

\[
SEL = 10 \log \left( \frac{E_{90}}{E_0} \right),
\]

where \( E_0 \) is the reference value of 1 μPa²·s.

The SEL for each impulsive noise event can be aggregated by summation to calculate the total SEL (or cumulative SEL) for the entire exposure duration. This is the main use of the metric, and in the case of a sequence of pulses from marine piling, the total SEL would be calculated for the entire sequence. This cumulative exposure is used to describe the noise dose which considers not only the peak pressure exposed to, but also the duration of the exposure. For injuries such as TTS and PTS, it is this cumulative exposure which is the important parameter to consider.

3. FLY/SWIM BY MODEL

The approach of Southall et al [3] recognises that even if the initial received levels are not great enough to cause injury, harmful effects can result from lower level sounds which last for a longer duration. In this paper, cumulative exposure levels are calculated for animals passing close to the selected noise sources. The source levels and characteristics for which the cumulative exposures are calculated for vessel propeller noise, based on a red/white noise model derived from source levels published by Arveson and Vendettis [8], and a sub-sea impact piling source, based on source levels measured by McHugh et al [9] which are discussed further in section 3. This allows the calculated cumulative exposures to be compared to the thresholds obtained from the literature, for example from the criteria published by Southall et al [3]. To do this, a trajectory is chosen for each animal whereby the animal swims past the source in a straight line at constant speed, heading and depth (it is assumed that no aversion is exhibited), where the trajectory is defined by the position of closest approach.

To calculate the cumulative SEL, the SEL is calculated for discrete intervals of one second assuming a realistic swim speed for the animal, and these are integrated over the entire “journey” (length of the trajectory) or time of exposure. Piling events can be assumed to have a well-defined start and stop time (they are not conducted continuously) which defines the duration. However, the vessel noise source is generally continuous and so a trajectory length has to be chosen over which the integration can take place. This was chosen to be 30 km. It should be noted that the large vessel remains stationary during the exposure, chosen to represent the behaviour of a dynamical positioned vessel maintaining station.

The receive levels at the receptor used in this paper are based on predictions calculated from assumed source levels and modelled transmission loss, which are discussed further in section 3 and 4 respectively.

4. REPRESENTATIVE SOURCE LEVELS
4.1. Vessel propeller noise

For purposes of this paper, a stationary source is considered for cumulative exposure calculation which represents propeller noise for a dynamical positioned vessel whilst maintaining station. Given the lack of availability of such data, the Arveson and Vendettis [8] data for a large vessel at high speed (shaft speed 140 rpm, approximately 16 knots) with substantial cavitation is considered. At these speeds, the higher frequencies are dominated by the cavitation but at lower frequencies the noise arises from different mechanisms. The Arveson and Vendettis [8] low frequency data is typified by peaks showing source levels of up to 182 dB re μPa·m at 38 Hz.

Although the higher frequency cavitation noise and lower frequency tonal noise are generated by different mechanisms, they make a similar contribution in the transition frequencies around 100 Hz. This complicates the identification of the source but allows summary third octave band data to be used to represent the total energy, as in the simple “red/white” model used by Hazelwood and Connelly [10] to estimate the noise power contribution over all frequencies [11]. This model is represented in Fig. 1 and is used for the broadband propeller noise cumulative exposure calculations in section 6.

![Fig. 1: Third octave band source level based on a red/white ship noise model.](image)

4.2. Marine impact piling

Marine impact piling noise is significantly more intense than ship noise, but is typically a short duration source with each operation typically lasting no longer than 40 minutes. For extremely close ranges, the peak levels generated by marine impact piling may actually exceed a level sufficient enough to cause instantaneous injury. However, for most scenarios, the potential for injury will depend on the integrated SEL over the full piling duration. A number of measurements have been made on marine impact piling for offshore windfarm construction [12][13][14][15][16] and the majority of the energy is typically reported [13] in the range 100 Hz to 1 kHz, with a pulse duration of the order of 0.15 seconds. Reported source levels for shallow water piling are very high, with peak source levels of in excess of 230 dB re μPa·m and SEL source levels in excess of 212 dB re 1 μPa²·s @ 1 m (dB re
μPa²·s·m²). Data published by McHugh et al [9] considers a sub-sea piling event (which will be considered for the purposes of this paper) at a depth of 95 m, reporting a source level of 210 dB re μPa·m [9]. Although a source level in terms of energy or SEL is not reported, it has been assumed that the piling pulse energy content and pulse duration has a similar relationship to the peak value as that obtained in shallow water. By this method, the energy (or SEL) source level was estimated as 195 dB re 1 μPa²·s·m² (or approximately 20 Joules).

5. TRANSMISSION LOSS

Single frequency transmission loss was calculated using the parabolic equation method, implemented using the RAM code. This code allows for range-dependent bathymetry to be implemented in the code to calculate transmission loss as a function of range and depth. This approach was used to estimate receive level at range for a single frequency at which the peak source level of a typical sub-sea piling event was believed to have occurred in the spectra – this was taken to be 210 dB re μPa·m at 200 Hz. For this work, it is the peak energy source level (195 dB re 1 μPa²·s·m²) which is of importance so this was also propagated to provide the received SEL.

This propagation model was performed for a sub-surface piling event (bottom source) in deep water >1 km. A number of single frequency source levels were also propagated to represent different harmonic components of propeller noise at the surface. Fig. 2 shows the transmission loss for a 100 Hz surface source as a function of range and depth (the black line on the colour map image indicates the seabed profile for the chosen transect), with a number of selected depth profiles obtained using the RAM code displayed for comparison with geometrical spreading.

![Fig. 2: Transmission loss for a 100 Hz source at a depth of 10 m.](image)

It can be seen from Fig. 2 that beyond a few km’s range, the transmission loss in the water column calculated using the RAM code does begin to settle between the transmission loss for simple spherical and cylindrical spreading. It does in fact approximate to a hybrid 14 log(r), at range for a 100 Hz propagation. This hybrid geometrical spreading was used to estimate the broadband transmission loss for third octave band frequencies which was subsequently used to calculate the broadband cumulative SEL, discussed in the following section, for propeller noise based on the source levels from the red/white ship noise model shown in Fig. 1.
6. CUMULATIVE SEL CALCULATED FOR LOW FREQUENCY CETACEANS

Using the methodology described in section 3, the swim-by model has been used to calculate the cumulative exposure assuming a number of conditions. These conditions are that the mammal, in this case, is a low frequency (LF) cetacean, swims at a constant speed and maintains a constant heading and depth whilst traversing the area of acoustic activity. For the sub-sea piling activity case study, a representative example was selected based on an energy source level of 195 dB re 1 \(\mu Pa^2\cdot s\cdot m^2\) at 200 Hz, being performed in around 2 km’s of water with substantial bathymetry changes in the area and a standard sound speed profile for deep water. Fig. 3 shows this particular example for a LF cetacean swimming directly above the impact piling source at a depth of 500 m with a swim speed of \(5 \text{ ms}^{-1}\). For this case where the piling activity takes around 60 mins, with the cetacean directly above the source at 30 mins, the total cumulative SEL is only around 192 dB re 1 \(\mu Pa^2\cdot s\). This falls below the criteria for injury suggested by the Marine Mammal Noise Exposure Criteria [3] of 198 dB re 1 \(\mu Pa^2\cdot s\) for single/multiple pulse sources.

![Fig. 3: Calculated SEL received level (upper plot) and cumulative SEL (lower plot) at 200 Hz for a LF cetacean at 500 m depth directly above the seabed source.](image)

Using the broadband approach discussed in section 5 for a ship/propeller noise source (based on the red/white model), the cumulative SEL per third-octave band to which the receptor is exposed whilst traversing the area of acoustic activity (assumed to be 30 km across) can be calculated. An example of this is shown in Fig. 4 for a LF cetacean traversing the area with a constant swim speed of \(5 \text{ ms}^{-1}\) at a depth of 10 m and assuming a constant heading. Fig. 4 shows the cumulative SEL results for a number of distances of closest approach between 1 m and 17 km. The maximum total SEL occurs for the 40 Hz band (corresponding with the peak of the red/white noise model) with a value of 184 dB re 1 \(\mu Pa^2\cdot s\) at 1 m. Even at 1 m, this is substantially less than the criteria for injury.
suggested by the Marine Mammal Noise Exposure Criteria [3] of 215 dB re 1 μPa²·s for tonal or broadband sources. However, the ship noise (assuming a stationary ship at maximum noise output representing a large dynamically position tanker holding station) resulted in a higher exposure under the assumed conditions than the sub-sea impact pile which has a much greater source level. It should be noted that the SEL values stated here are weighted according to the animal classifications defined by Southall et al [3], i.e. weighted for a LF cetacean in this instance. The sources have also been categorised as non-pulses and pulse/multi-pulses to maintain consistency with the source classifications used under Marine Mammal Noise Exposure Criteria [3].

![Graph](image)

*Fig. 4: Calculated cumulative SEL per third-octave band for a LF cetacean at 10 m depth traversing the sound source at different distances of closest approach.*

7. CONCLUSIONS

A method for calculating cumulative exposure based on SEL using a swim-by model has been outlined and results have been presented for a case study based on two representative noise sources. The first source, sub-sea impact piling, was used to represent a multi-pulse source with a very high source level and the second source, large vessel propeller noise, was chosen to represent a lower source level noise of longer duration. The cumulative exposure was calculated for each for predefined set of conditions. For the conditions considered in this paper, the ship noise resulted in a higher exposure than the sub-sea impact pile which had a much greater source level. However, both sources resulted in cumulated SEL’s below the injury criteria proposed by Marine Mammal Noise Exposure Criteria [3].

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A METHODOLOGY FOR THE MEASUREMENT OF RADIATED NOISE FROM MARINE PILING

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Abstract:

Marine piling is the most commonly-used method for offshore windfarm construction, and consists of steel monopiles being driven into the sea-bed using powerful hydraulic hammers. This is a source of high-level impulsive sound that can travel considerable distance in the water column.

This paper describes a methodology that has been developed for measuring marine piling noise, which is designed to record the temporal, spatial and spectral characteristics of the radiated sound field. In the method, a number of recording systems are simultaneously deployed at various ranges and depths. Fixed recording buoys allow the full piling sequence to be measured, and variations in the temporal and spectral characteristics to be assessed. This enables the effect of any source level variation with time to be determined, for example that due to a soft start procedure. To assess spatial variations in the radiated acoustic field, recording samples are also made using hydrophones deployed from a vessel which traverses the field along a radial transect from the pile location. This latter set of measurements allows an estimate of the effective source level to be made if a suitable transmission loss model is used. To illustrate the method, some results are presented of measurements made on marine piling in shallow coastal waters during the construction phase of offshore windfarms.

Keywords: Underwater noise, marine piling.
1. INTRODUCTION

Noise is often an unintended by-product of offshore activities, and the increasing levels of man-made sounds in the ocean (whether deliberately generated or not) have led to concern over marine noise pollution and its effect on marine life.[1] A significant source of impulsive underwater noise is marine piling where a pile is driven into the sea-bed using a hydraulic hammer. Such a technique is typically used to position piles in relatively shallow water for construction of offshore windfarms, bridge supports, and offshore structures associated with the oil and gas industry.

A methodology that has been developed for measuring marine piling noise, which is designed to record the temporal, spatial and spectral characteristics of the radiated sound field. In the method, fixed recording buoys allow the full piling sequence to be measured so that variations in the temporal and spectral characteristics of the acoustic field may be assessed. This enables the effect of any source level variation with time to be determined, for example that due to a soft start procedure.[2] To assess spatial variations in the radiated acoustic field, recorded samples are also made using hydrophones deployed from a vessel which traverses the field along a radial transect from the pile location. [3-8] This latter set of measurements allows an estimate of the effective source level to be made if a suitable transmission loss model is used. To illustrate the method, some results are presented of measurements made on marine piling in shallow coastal waters during the construction phase of offshore windfarms. Measurements made using the described methodology may be used to estimate the overall sound exposure of marine life using accepted exposure metrics and criteria for the threshold of bio-physical or behavioural effects. [9,10]

2. METHODOLOGY

2.1. Measurement method

The methodology used for measurements has two main features:

- fixed recording buoys recording the full piling sequence so that variations in the temporal and spectral characteristics of the acoustic field may be determined;
- recorded samples of the field using hydrophones deployed from a mobile vessel to determine the spatial variation of the acoustic field.

The fixed buoys are custom-designed, static recording buoys that are capable of recording the entire piling sequence at one location. The vessel-deployed recording systems consist of broadband hydrophone arrays operated from a work boat which is free to move along a transect in a radial direction away from the pile location. This combination provides simultaneous recording of the entire piling sequence from fixed locations to assess changes in the source over time. Such changes may be due to changes in hammer energy (due to a ‘soft
start’ procedure), or due to increasing pile penetration depth, changes in sediment composition, etc. The combination also provides an assessment of propagation losses within the water column by sampling the field at multiple ranges and depths along a specific radial transect. The exact configuration adopted depends on the particular requirements. In some cases, recordings have been made simultaneously using hydrophone systems at up to nine spatial positions within the acoustic field. On several occasions, a hydrophone system has been deployed from the piling vessel itself. Figure 1 shows a schematic diagram of the typical spatial arrangement of hydrophones employed for measurements.

![Schematic diagram of hydrophone arrangement](image)

**Fig. 1: A schematic diagram showing the methodology employed for measurements.**

With the buoy systems, either one or two hydrophones are deployed in a bottom-mounted configuration on a sub-surface buoy, with the hydrophones distributed vertically in the water column. Buoy systems are generally deployed at ranges of between 1 km and 22 km from the driven pile. At least one buoy, termed the calibration buoy, is deployed within 2 km of the pile being driven to provide a clean recording of the whole piling sequence with a good signal-to-noise ratio. In addition, recording buoys are positioned at other locations of interest, for example close to areas where sensitive marine species are present. Where possible, a hydrophone and recording system is deployed from the piling vessel itself. This allows measurements to be made at close range, between 10 m and 50 m from the pile.

In addition to the buoy systems, a work-boat is also used to deploy broadband hydrophone arrays with up to 200 kHz bandwidth. The hydrophone sensors are distributed within the water column and measurement samples are taken at various ranges from the pile. Typically, the work-boat starts at around 100 – 200 m away from the pile being driven and then a series of measurements are made on a radial transect away from the pile location using a “sprint/stop/measure” procedure. The transect used is chosen to pass through the location of the static calibration buoys. Measurements are made with the vessel quiet (engines off, echosounder off, and ideally with the generator off). Typical measured sequences last for a period of around 2 minutes. The vessel then moves to a new position along the transect. Using this
methodology to measure a piling sequence lasting 80 minutes, typically eight ranges can be used with a maximum range of 15 - 20 km.

The full piling sequence data from the calibration buoy is then used to correct for the variations in source level that occur between the times that the individual work-boat measurements were made. By this means, the measurements made as a function of range may be normalised to the same source level (typically the maximum value is used). For the measurements to be correlated, all recordings must be accurately time stamped. The measurement ranges and buoy locations are GPS position fixed, and a sound velocity profile is taken using a CTD sonde at the location of the calibration buoy. In the shallow coastal waters where offshore windfarms are constructed, the water is typically well mixed with no thermocline present.

2.2. Equipment and Instrumentation

For the recording systems deployed from the work boat, data acquisition is carried out using PC-based broadband analysis systems with sampling rates of 500 kHz or greater. This allows signals with frequencies greater than 200 kHz to be faithfully recorded. Three data acquisition systems have been employed for this work: an NI-DAQ 6062 E at 500 kS/s and 12 bit resolution; NI-DAQ-USB NI9162 at 500 kS/s and 12 bit resolution; and a dual channel Brüel and Kjær Pulse broadband analysis system capable of sampling at 524 kS/s with 24 bit resolution. Two models of hydrophone have been used for the vessel deployment: Reson TC4014 hydrophones (manufactured by Reson in Denmark), and HS150 hydrophones (manufactured by SRD Ltd in UK). These hydrophones are deployed at evenly distributed depths within the water column. Broadband, low-noise conditioning preamplifiers are used to amplify the signals from the HS150 hydrophones. The TC4014 hydrophones contain integral preamplifiers of fixed gain which can distort or even saturate if used to measure the high-amplitude acoustic pulses present in the vicinity of the pile. Data from these hydrophones are in general only used for measurements made at ranges greater than 2 km from the pile.

The buoy recording systems use two HS70 hydrophone elements (also from SRD Ltd). Data acquisition is made to solid state drives at up to 24-bits and a 48 kHz bandwidth. When an additional hydrophone is deployed from the piling vessel itself, this is an HS150 hydrophone, with data recorded digitally with a bandwidth from 20 Hz to 22 kHz with 16 bit resolution. In this case, no preamplifier gain is required.

All data acquisition electronics and amplifiers are calibrated before and after the trials. All hydrophones are calibrated by NPL over their complete frequency range of use, with calibrations traceable to UK national standards at NPL.

3. RESULTS

Some results are shown below for measurements of noise radiated from marine piling operations made using the above methodology. The pile diameter was 4.74 m and the sediment in the area mostly consists of hard chalk. The depth of water in the area varies from approximately 8 m to 15 m depending on local variation in bathymetry and the tide.

Figure 2 shows the time and spectral content of typical waveform recorded at a range of only 10 m from a 4.74 m diameter pile at full hammer energy (1900 kJ). In general, the pulse periodicity observed was approximately 2.5 seconds during the main piling sequences studied. Acoustic pulse durations were about 0.15 s close to the source, but could be as long as...
as 0.5 s at a range of 21 km. Primary frequency content is around 200-300 Hz, with a majority of the energy at frequencies of less than 10 kHz. However, close to the pile there are frequency components present at high tens of kilohertz.

Fig. 2: A typical pulse recorded using the hydrophone at a range of approximately 10 m. This recording system operated over audio band frequencies (maximum frequency: 22 kHz). The amplitude of the spectrogram is normalised and plotted in dB.

Fig. 3: Upper plot: time history of piling sequence measured at the calibration buoy. Lower plot: normalised peak-to-peak sound pressure levels for each measured pulse. This data may be used to correct the workboat data for source time-variation. Also shown with dotted lines are the time windows during which the workboat measurements were made, with the annotations indicating the range of the workboat from the pile in metres.

Figure 3 shows how the variation in source level determined from the complete piling sequence recorded by the static calibration buoy may be correlated with the data obtained from the workboat. The upper plot shows the time history of the received signals at the calibration buoy, with a gradually increasing amplitude as the hammer energy is slowly increased. Shown on the lower plot are the peak levels for the calibration buoy recording and the times during which the workboat measurements were made, with the annotations corresponding to the ranges from the pile in metres. In order to assess the transmission loss, the workboat data must be corrected to account for the time-variation. Since the maximum
source level is typically of greatest interest when considering impact on marine life, the workboat measurements made at times before the pile had reached maximum level must be increased to a value representative of the value corresponding to the maximum source level. Figure 4 shows the data measured at each of the ranges covered by the workboat after correction using the data from the static calibration buoy. This data may then be used to estimate received level at intermediate ranges, and an effective source level using an appropriate transmission loss model.[4-8]

![Figure 4](image1.png)

**Fig. 4:** Values of received peak levels and SEL plotted against range after correction for time variation using the calibration buoy data.

Figure 5 shows the received Sound Exposure Level (SEL) in relation to step increases in hammer energy for a soft-start period recorded at a range of 1.5 km.

![Figure 5](image2.png)

**Fig. 5:** The received SEL level in relation to step increases in hammer energy for a soft-start period recorded at a range of 1.5 km.

Figure 5 shows the received Sound Exposure Level (SEL) in relation to step increases in hammer energy for a soft-start. The SEL is a measure of the energy in the acoustic pulse and is obtained by integrating the square of the acoustic pressure waveform (the units are $\mu$Pa$^2$s). This metric has found to be more robust than peak pressure (which is more sensitive to transmission loss fluctuations) and RMS pressure (sensitive to uncertainty in pulse duration).[2,3] SEL also has the advantage that it may be used to evaluate the cumulative exposure for an animal over a prolonged duration. [9,10]
There are a number of reasons why the source level may appear to fluctuate with time, for example changes in sediment properties during seabed penetration, and changes in transmission loss due to local environmental changes (for example due to sea surface fluctuation in poor weather). However, a major reason is that the hammer energy is often increased during what is sometimes called a “soft start”. The energy in the acoustic pulse has been found to depend approximately linearly on the hammer energy. This is illustrated by Figure 6 which shows the results of plotting the mean acoustic pulse energy (expressed in units of J/m$^2$) against hammer energy (in kJ). The error bars represent the random uncertainties expressed for a confidence level of 95% (essentially twice the value of the standard deviation) and the straight line is a weighted least squares fit (weighted according the inverse of the variance). It should be noted that although the dependence is linear over the energy range shown, the fitted line does not quite go through the origin (the intercept is 0.08 mJ/m$^2$). This illustrates that there are aspects of the radiation mechanism which are poorly understood. A comprehensive physical model is required in order to predict the dependencies accurately.

4. CONCLUSIONS

This paper describes a methodology that has been developed for measuring marine piling noise, which is designed to record the temporal, spatial and spectral characteristics of the radiated sound field. Results are presented for measurements in a shallow water coastal site where measurements of the entire piling sequence were conducted at ranges from 10 m to 22 km for piles in 10 m to 20 m of water depth. To assess variations in the temporal, spatial and spectral characteristics, a number of recording systems were simultaneously deployed at various ranges and depths, allowing the full piling sequence to be measured. This allowed assessment of source level variation at fixed locations, and the effect of propagation within the water column. An approximately linear dependence of acoustic pulse energy with hammer energy is demonstrated.
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NATURAL AND ANTHROPOGENIC SOURCES OF SOUND IN THE NORTH SEA

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Abstract: This paper summarises the progress made by TNO on describing the most significant natural and anthropogenic sound sources in the North Sea (Dutch sector). An assessment of their relative importance is made by estimating an annual energy budget, not neglecting that details of the sound distribution in frequency, time and space might be equally important. Our approach to propagation modelling of the noise sources in the shallow waters near the Dutch coast is explained. Example noise maps are presented. Validation of the noise predictions requires a capability for long-term noise monitoring. Suitable monitoring solutions are discussed.

Keywords: Underwater sound sources, North Sea, energy budget, noise maps

1. INTRODUCTION

For a proper (national) implementation of the European Union’s ambitious Marine Strategy Framework Directive, it is required that the impact of anthropogenic activities on the North Sea environment be assessed thoroughly. One of the effects of the use of the North Sea by humans is the generation of underwater sound. Sound propagates over longer distances in water than in air. The impact of anthropogenic underwater sound sources could therefore be serious.

At this time, there is insufficient information on the underwater sound environment in the North Sea to make an impact assessment. Once this information becomes available, the next challenge is to predict correctly how the sound propagates in the shallow water of the North Sea, i.e. to find out the sound footprint of individual anthropogenic or natural sources of sound. The research reported on here assesses the existing knowledge on the underwater sound environment and identifies the gaps.
The first step is an inventory of all relevant natural and anthropogenic sources of sound, similar to Ref. [1], with specific information on source levels, frequency bands, etc. Sound sources in the air (e.g. aircraft) are excluded from the study. The study is limited to the Netherlands Continental Shelf (NCP), which covers an area of approximately 57,000 km$^2$.

2. NATURAL SOUND SOURCES

Ubiquitous natural underwater sound sources are wind and rain. Lightning is also subject to scrutiny because of the large amount of energy available in each individual strike. Compared to the sound levels due to these causes, the levels due to underwater fauna – marine mammals, fish, crustaceans and other biota – are small. As such, the animals do not substantially contribute to the total (time-averaged) sound levels in the North Sea. Other natural sources in the North Sea include precipitation other than rain, breaking gravity (surf) waves, wave-wave interactions and gravel noise.

For the main natural sources wind, rain and lightning, the total acoustic energy produced on an annual basis in the NCP has been estimated by Ref. [2]. Based on formulas from Ref. [3] for the spectral density of the wind’s dipole strength, the total radiated acoustic power on the NCP is estimated to first order. For wind speeds of 5-10 m/s at 10 m above the sea surface, the underwater acoustic power due to wind equals 2-9 GJ/y (gigajoules per year). Similarly, again using formulas from Ref. [3], the total radiated acoustic power of rain is estimated. In these simple formulas, the influence of drop size is neglected. For an annual rainfall of 800 mm, uniformly spread over the NCP area, the underwater acoustic power due to wind is estimated to be 0.3-1 GJ/y.

Only little has been published on underwater acoustics of lightning strikes. Ref. [4] estimates a source level of 260.5 dB re 1 $\mu$Pa$^2$m$^2$, which for a typical strike duration of 30 $\mu$s converts to a source energy of 30 kJ. Assuming 2 strikes per km$^2$ per year [5], this gives 114,000$\times$30 kJ/y $\approx$ 3.4 GJ/y for the underwater acoustic power due to lightning. Assuming further a total discharge energy of 500 MJ [5] (http://en.wikipedia.org/wiki/Lightning), this implies a 0.006% efficiency for conversion of electrical energy of the discharge to acoustic energy in the water. In practice, the estimation of source level, and the associated conversion efficiency, is speculative and subject to high uncertainty (several orders of magnitude).

3. ANTHROPOGENIC SOURCES

The anthropogenic sources are divided in two categories: intentional sound sources and unintentional ones. The latter are undesirable by-products, e.g. shipping noise, while for the former the production of sound is essential, e.g. sonar.

Relevant intentional sources are airgun arrays for seismic explorations, sonar equipment (single- and multi-beam echo sounders, sub-bottom profilers, side-scan, fish-finding, research and military search sonar) and acoustic deterrent devices. Other minor or unknown intentional sources are obstacle avoidance sonar, minesweeping equipment, Doppler current profilers, acoustic communications equipment, acoustic transponders and acoustic cameras.

Important unintentional sources include shipping (merchants, ferries, super/tankers, leisure craft, fishing vessels), pile driving for offshore construction, dredging, underwater explosions (mine & bomb clearance), operational oil & gas platforms and wind farms. Other minor or
unknown unintentional sources are pipe laying, cable laying, flow noise from pipelines, industrial/harbour noise and wind farm/offshore decommissioning.

The total acoustic energy produced on an annual basis on the NCP by the main anthropogenic sources has been estimated by Ref. [2], of which the results are summarised in column 2 of Table 1. From these values, it might seem that echo sounders contribute about 10 times more to the underwater sound than pile driving. However, this would be a misleading interpretation as the two sources have very different frequency ranges. Therefore, the total acoustic energy of the noise sources has been calculated, accounting for frequency-dependent absorption [2], as explained below.

For every source, the free-space energy density has been calculated, assuming a point source of the same power and frequency and assuming spherical spreading with frequency-dependent absorption. Estimates of the total acoustic energy are then obtained by integrating the energy density over all space. The results are given in column 4 of Table 1, which shows that the total sound energy due to pile driving exceeds that due to echo sounders by at least two orders of magnitude.

The table also shows that four of the sources considered, namely airguns, shipping, pile driving and explosions, result in significantly more sound energy than the remaining three. These four are all low-frequency sources, resulting in low absorption, long-range propagation and hence the high estimated values of total sound energy. It does not necessarily follow from this that they have the greatest impact, because some animals are most sensitive to high-frequency sounds that might propagate less effectively, resulting in lower estimates of total energy. A complete impact analysis should consider other aspects as well, such as temporal or spatial variability.

Table 1. Estimation of annual average acoustic power output of anthropogenic sound sources on the NCP (column 2) and order of magnitude estimation of the total (free-space) acoustic energy (column 4).

<table>
<thead>
<tr>
<th>Type of source</th>
<th>Estimated annual average of acoustic power output in the North Sea (NCP) [GJ/y]</th>
<th>Order of magnitude estimation of frequency [kHz]</th>
<th>Order of magnitude estimation of time-averaged total (free-space) acoustic energy [kJ]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Airgun arrays (3D seismic survey)</td>
<td>30-300</td>
<td>0.01-1</td>
<td>1000-10,000</td>
</tr>
<tr>
<td>Shipping</td>
<td>85-850</td>
<td>0.03-3</td>
<td>1000-10,000</td>
</tr>
<tr>
<td>Pile driving (wind farm construction)</td>
<td>2-20</td>
<td>0.01-1</td>
<td>100-1000</td>
</tr>
<tr>
<td>Explosions (clearance of historic munitions)</td>
<td>&lt; 14</td>
<td>0.01-1</td>
<td>100-1000</td>
</tr>
<tr>
<td>Navigation echo sounders</td>
<td>20-200</td>
<td>10-300</td>
<td>&lt; 1</td>
</tr>
<tr>
<td>Fish-finding sonar</td>
<td>3-30</td>
<td>10-300</td>
<td>&lt; 1</td>
</tr>
<tr>
<td>Military search sonar</td>
<td>&lt; 0.2</td>
<td>1-100</td>
<td>&lt; 1</td>
</tr>
</tbody>
</table>

4. NORTH SEA NOISE MAPS

For estimation of the spatial noise distribution due to sources in the North Sea, a simple and robust underwater sound propagation model has been implemented in a computer code. The applied modelling methodology is based on the work by D.E. Weston [6]. As an example, some noise maps are presented for a selection of cases in Fig. 1. It should be emphasized that these maps are mainly indicative, due to the uncertainties involved (measured source levels, modelling of environment).

For the dredger cases, the source spectrum of the trailing suction hopper dredger *Gerardus Mercator* is used (http://www.sakhalinenergy.com/en/documents/doc_33_cea_tbl4-7.pdf). The wind farm consists of 60 turbines. The source spectrum of each turbine was derived [2]...
from measurements of a Danish Bonus turbine (*Paludans Flak* [7]). The received noise levels are averaged over depth.

The source levels of wind and rain are computed using the formulas from [3] (and [2]), using 30-year averages for the wind speed distributions (ERA-40 database, accessible via http://climexp.knmi.nl) and a uniform precipitation density. The received noise level is computed at the seabed for these noise maps.

Looking at Fig. 1, the influence of the local water depth is clearly visible for the dredger and rain noise maps. Fig. 2 shows that the dredger noise levels are relatively small in the lower octaves, due to duct cut-off, and in the higher octaves, due to surface scattering and absorption.

**Fig. 1.** (a) Bathymetry of the North Sea (GEBCO 1-minute grid). Grey indicates areas above (mean) sea level (a-d). The other panels show simulated broadband noise distributions (received sound pressure level) for (b) a dredger north of the Dutch Wadden Islands, (c) the same dredger near the Rotterdam harbour, (d) a wind farm northwest of Amsterdam, (e) wind noise (at the seabed) in January, and (f) rain noise (at the seabed) for a uniform rain rate of 5 mm/h and an annual wind speed distribution.
5. MONITORING SOLUTIONS

The assessment of noise sources in Sections 2 and 3 is based on the scarce data that we have been able to find in publications or acquired through our professional contacts. Also, the demonstrated noise map capability relies on the availability of reliable source level distributions and noise measurements for validation of the modelling assumptions.

What is needed is a (semi) continuous and systematic underwater measurement effort that will reveal the actual noise levels and their variation in time throughout the NCP. This will require an extensive grid of autonomous underwater monitoring stations.

An inventory of long-term monitoring solutions was made recently [8] to be able to judge the present state-of-the-art of commercially available monitoring systems. A total of 12 (potentially) suitable systems has been identified, see Table 2. Which one of them is to be preferred depends strongly on the specific requirements regarding flexibility, technical specifications (energy consumption, storage capacity, dynamic and frequency range, recording scheme, sensitivity, etc.), ease of handling and price.

For most systems, the endurance is limited by storage capacity and not by battery life. The endurance for battery-fed systems is up to one year, when not continuously recording.

### Table 2. Inventory of monitoring systems [8].

<table>
<thead>
<tr>
<th>Monitoring system</th>
<th>Supplier</th>
</tr>
</thead>
<tbody>
<tr>
<td>PAL</td>
<td>APL-UW (US)</td>
</tr>
<tr>
<td>RUDAR</td>
<td>Cetecean Research Technology (US)</td>
</tr>
<tr>
<td>WAMS</td>
<td>High Tech (US)</td>
</tr>
<tr>
<td>SRB-16</td>
<td>High Tech (US)</td>
</tr>
<tr>
<td>IMARES</td>
<td>IMARES (NL)</td>
</tr>
<tr>
<td>BANAS</td>
<td>Istanbul Technical University (TU)</td>
</tr>
<tr>
<td>OBHS</td>
<td>KUM (GE)</td>
</tr>
<tr>
<td>AURAL-M2</td>
<td>Multi Électronique (CA)</td>
</tr>
<tr>
<td>RASP MT200</td>
<td>Nauta (IT)</td>
</tr>
<tr>
<td>EAR</td>
<td>Oceanwide Science Institute (US)</td>
</tr>
<tr>
<td>OptaMarine</td>
<td>OptaSense / QinetiQ (UK)</td>
</tr>
<tr>
<td>HARP</td>
<td>Scripps Whale Acoustic Lab (US)</td>
</tr>
</tbody>
</table>

Fig. 2. The noise distribution of Fig. 1b in 4 octave bands with centre freqs. of 31.5, 125, 500 and 2000 Hz.
6. CONCLUSIONS AND RECOMMENDATIONS

The main contributions to sound energy in the North Sea are estimated to come from shipping, seismic surveys (airguns), underwater explosions and pile driving. This energy estimation takes no account of the hearing sensitivity of individual species.

To find out the sound footprints of these sources, e.g. needed for the development of suitable mitigation measures [2], underwater sound maps are needed. In order to make reliable noise maps, a systematic measuring effort of relevant source level distributions and environmental noise levels should be set up. This requires suitable monitoring solutions and international agreement on guidelines and protocols for the measurement, processing and quantification of underwater sound.

Finally, there is a demand for research on the possible impact of underwater sound on diverse species. This refers to the individual physiology and the short-term dose-response relationship as well as to the long-term impact on the population. It is therefore recommended that experts from various disciplines (acousticians, ecologists, biologists) establish some sort of platform with the aim of improving the collaboration.

7. ACKNOWLEDGEMENTS

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REFERENCES

Structured Session 28

Tank Experiments

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EXPERIMENTAL EVIDENCE OF RANGE DEPENDENCE OF MODE CUT-ON FREQUENCY IN A SCALED WEDGE-LIKE ENVIRONMENT

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Abstract: In this paper, experimental results from previous laboratory scale measurements of acoustic propagation over a wedge-shaped oceanic bottom are used to show the range dependence of the mode cut-on frequency for one specific propagating mode. Numerical simulations performed using a three-dimensional parabolic equation based code confirm this effect for all the propagating modes present in the waveguide.

Keywords: Tank experiment, Sound propagation modeling, 3-D effects
1. INTRODUCTION

In a previous work, results of laboratory scale measurements of long-range, across-slope propagation of broadband pulses in a shallow-water wedge-shaped oceanic waveguide with a sandy bottom were reported and analyzed [1]. Strong 3-D effects like modal shadow zones and multiple modal arrivals were observed, in agreement with theoretical predictions. In the present work, we focus on the frequency dependence of the received signals along the cross-slope direction. The wave packets associated to mode 3 are extracted from the received signals at several ranges and their spectra are examined. It is observed that, as we move out in range across-slope, the low frequency part of the spectral content of the wave packets associated to mode 3 is progressively removed, giving evidence of a well-known 3-D effect referred to in the literature as the frequency dependence of the mode cut-off range [2], [3]. Numerical simulations performed using a 3-D model confirm these observations for all the existing propagating modes present in the wedge-shaped waveguide. The experimental set-up is recalled in Sec. 2. Spectra of mode 3 wave packets are shown and analyzed. The numerical results are presented in Sec. 3. The paper closes with a brief discussion.

2. EXPERIMENTAL RESULTS

The experimental results presented in this section come from a measurement campaign conducted in July 2007 using the tank facilities of the LMA-CNRS laboratory in Marseille (France). The shallow water tank (10-m-long and 3-m-wide) contains a thin layer of water overlying a thick layer of calibrated river sand simulating a bottom half-space. The experimental set-up, which is similar to the one described in [4], intends to simulate a shallow-water wedge-like environment, with the wedge apex aligned along the longer side of the tank, and a slope angle of approximately $4.5^\circ$. The source and receiver are cylindrical piezoelectric transducers both with diameters of 6.0 mm. The broadband signal used at the source was a 5-cycle Gaussian pulse with a duration of $\approx 40 \mu s$. Its spectrum was centered at $\approx 140$ kHz, with a bandwidth of 100 kHz.

During this measurement campaign, the source was fixed at a depth of 10 mm. The received signals were recorded along the cross-slope direction at several source/receiver distances (from 0.1 to 5 m, with an increment of 0.1 m), and, at each range step, they were recorded at depths between 1 and 48 mm with a depth increment of 1 mm. The water depth was measured to be 48 mm $[\pm 1 \text{ mm}]$ along that specific direction, leading to only few propagating modes at the central frequency of the source signal. As detailed in [1], the recorded time series exhibit typical and strong 3-D effects like mode shadow zones, multiple mode arrivals, and intra-mode interferences. In particular, a careful analysis of the transmission loss vs range curves (obtained from the experimental time series at specific frequencies by means of Fourier transform) showed the cut-off of each mode to occur at longer ranges with increasing frequency. This well-known effect (e.g., Refs. [2], [3], and [5]) becomes explicitly apparent by inspecting the spectral content of the arrivals of specific modes as a function of range. For instance, this is illustrated in Fig. 1 where the frequency spectra of the wave packets associated to
Fig. 1: Experimental spectra of arrivals associated to mode 3, at a receiver depth of 20 mm and several source/receiver distances ranging between 1.4 m and 2.4 m.

arrivals of mode 3 are displayed at several distances from the source and at a receiver depth of 20 mm. The spectra were obtained as Fourier transforms of the mode-3 arrivals at respective ranges. The mode-3 arrivals were extracted from the received time series and weighted with a Hanning window to smooth the edges prior to being Fourier transformed. Note that, although the amplitude scaling of the spectra in Fig. 1 is arbitrary, the same scaling has been used in each subplot. The frequency dependence of the cut-off range, or, equivalently, the range dependence of the cut-on frequency of mode 3 appears clearly in Fig. 1. More precisely, at the range of $r = 1.4$ m, no cut-off has taken place yet, and the cut-on frequency of mode 3 is approximately 100 kHz. As we move out in range across-slope, the lower frequency part of the spectrum energy is progressively removed due to downslope refraction. For instance, at $r = 2$ m, the cut-on frequency has moved to approximately 150 kHz, leaving the part of the spectrum up to that frequency in the shadow zone. Moreover, the increase of the peak amplitude observed, e.g., from $r = 1.4$ to 1.9 m, gives evidence of an additional arrival of mode 3. Additionally, the interference patterns observed at some ranges (e.g., at $r = 1.8$ m) are attributed to two distinct mode-3 arrivals occurring at different times. Finally, noting that the interference patterns depend on the relative time delay of the two arrivals, we conclude that, at ranges greater than 2.1 m, they are almost simultaneous and intra-mode interferences are very weakly observed.
3. NUMERICAL RESULTS

The numerical simulations described here were obtained using a 3-D parabolic equation based model [6]. A scale factor of 1/1000 was used in the simulations. The computations were carried out considering a Gaussian broadband source with a central frequency of 141.6 Hz with a 80 Hz bandwidth covering the band 100–180 Hz. The initial pulse length is approximately 50 ms. The propagation domain consists of a lossless homogeneous water layer with a sound speed of 1488.9 m/s (the temperature of the water layer in the tank was 22.19°C, [7]) and a density of 1 g/cm³, overlying a lossy half-space sediment bottom with a homogeneous sound speed of 1740 m/s, a density of 1.99 g/cm³, and an absorption of 0.5 dB/wavelength. The wedge-shaped waveguide has a tilted angle of 4.5°, with a water depth at the source position of 48 m. The source is positioned at 10 m below the water surface.

Stacked time series vs range are plotted in Figs. 2 and 3, for receiver depths of 20 and 26 m, respectively. The left panels illustrate the solution from a 3-D computation, while the right panels show, for comparison, the 2-D solutions for the equivalent 2-D environment. Additionally, stacked time series vs depth from the 3-D computation are plotted in Fig. 4, for three source/receiver distances. The 2-D computation gives rise to four distinct wave packets corresponding to signals carried by four propagating modes.

Fig. 2: Stacked time series versus range along the cross-slope direction at a depth of 20 m corresponding to a 3-D (left) and 2-D (right) computations. The source depth is 10 m.
Fig. 3: Stacked time series versus range along the cross-slope direction at a depth of 26 m corresponding to a 3-D (left) and 2-D (right) computations.

Note that the receiver depth of 26 m is in the close vicinity of nulls of modes 2 and 4. Hence, these two modes are not visible in the corresponding received signals of Fig. 3. By combining the 3-D results of Figs. 2–5 we are led to a clear identification of the multiple time arrivals of each mode, being distinct at some ranges, then merging together and progressively disappearing as we move out in range across-slope (modal shadow zone regions).

The above results from the numerical simulations are in good agreement with the experimental results reported in Ref. [1] (not shown here, cf. Figs. 4 an 5 therein). Furthermore, Fig. 5 displays the frequency spectra of the wave packets associated to each of the four modes as a function of range, and at a depth of 20 m. The respective modal wave packets were now isolated using a mode filtering technique (as described in Ref. [8]). The range dependence of the mode-3 spectral content obtained numerically compare favorably with the respective spectra of Fig. 5 obtained experimentally. Moreover, similar effects are observed in Fig. 5 for modes 2 and 4, though they occur at different ranges. More precisely, for higher-order modes they occur at shorter ranges than for lower-order modes. Note lastly that at the range of 4.75 km no cut-off has taken place for mode 1, as can be deduced from its bandwidth at that range. However, the stretch in the interference patterns of mode-1 spectrum suggest that mode 1 is close to entering its shadow zone.
4. DISCUSSION

Experimental measurements of long-range across-slope pulse propagation in a scaled wedge-like environment reported in a previous work, were used to illustrate a 3-D effect known as the frequency dependence of the mode cut-off range. The wave packets associated to mode 3 were extracted from the experimental time series and analyzed in the frequency domain. This effect, also known as the range dependence of the frequency mode cut-on becomes explicitly apparent by inspecting the spectral content of mode 3 as a function of range. Moreover, the patterns observed in mode-3 spectra give evidence of the multiple mode arrivals and intra-mode interferences. Time domain simulations using a 3-D PE model turned out to compare favorably to the experimental observations. The modal wave packets were isolated using a mode filtering technique and then analyzed in the frequency domain. Similar effects were also observed for modes 1, 2, and 4, although occurring, as expected, at longer ranges for modes 1 and 2, and at shorter ranges for mode 4. As interest in 3-D modeling is continuously growing due to increasing performance and capacity of computers, these experimental data turn out to be potentially promising for 3-D model validation. Current work is focusing on more detailed comparisons with a 3-D PE model.
Fig. 5: Numerical spectra of each mode at a receiver depth of 20 m and several source/receiver distances ranging between 1 km and 4.75 km.

REFERENCES


EXPERIMENTAL DETERMINATION OF RESONANCE MODES OF A STREAMLINE TUBE

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Abstract: The aim of this work is to obtain, experimentally, the resonance frequencies of waves circumnavigating around an aluminum streamline cross-section tube, immersed in water and filled with air. This tube is insonified, in far field, by a plane wave normally to the z-axis with the use of an impulse method. Different incidence directions are considered because of the non-axisymmetry cross section. The incident signal is a broadband short pulse and the scattered signal is the time-domain response of the tube obtained for a given azimuth angle. Only one transducer, acting alternatively as an emitter and as a receiver (monostatic set-up), enables us to obtain these responses in backscattering. After signal processing of the time-domain response, a resonance spectrum is obtained and resonances are thus isolated for a given incidence angle. In a last step, at resonance frequencies, the angular scattering patterns are analyzed. These patterns are obtained from a bistatic set-up, i.e. a set-up with two transducers, one acting as an emitter, and the other one as a receiver.

Keywords: acoustic scattering, arbitrary cross-section cylinder, streamline cross-section tube, experiment
1. INTRODUCTION

Considerable work has been done on the scattering by cylindrical objects having a circular cross section [1]. Comparatively, little attention has been given to the more general case of the noncircular cylindrical cylinders. Experimental results are even fewer for this type of scatterers. The scatterer considered in this paper is an aluminum streamline tube characterized by the dimensions of cross section: 15.7 mm in major axis (x-axis), 7.1 mm in minor axis (y-axis) and a wall thickness of 0.4 mm as presented in Fig. 1. For example, this shape of cross section can be the one of a marine current turbine blade.

The streamline tube is immersed in water and excited by a plane wave normally to the z-axis by a transducer Panametric. The transducers used are broadband with a central frequency of 500 kHz or 2.25 MHz. These transducers allow us to analyze the acoustic scattering in a wide range of frequency 100 kHz – 3000 kHz. The experiments are conducted in the far field of the transducers. As the diameter of the radiating surface of the transducers is 3.8 centimeters, this dimension is large enough to be sure that the cylinder is fully insonified.

The non-axisymmetry of this object involves the definition of an incidence angle $\theta_{\text{inc}}$. This angle is defined from the major axis $x$. The figure 2 presents three examples of incidence. The time-domain response of the target depends also on the azimuthal angle of observation $\theta_{\text{azim}}$. The backscattered signal is obtained by considering only an incidence angle $\theta_{\text{inc}}$ ($\theta_{\text{inc}} = \theta_{\text{azim}}$). In this case, one transducer (monostatic set-up) is used, acting alternatively as an emitter and as a receiver (monostatic set-up). In the case where $\theta_{\text{inc}}$ is different of $\theta_{\text{azim}}$, two transducers (bistatic set-up) are used, the emitter is defined by the fixed angle $\theta_{\text{inc}}$ and the receiver by the angle $\theta_{\text{azim}}$. This angle is defined from the angular position of the emitter-transducer.

An impulse measurement method is used in this work. This method is described in detail in reference 2. The excitation signal is a broadband short pulse and the scattered signal consists of a series of echoes. In this time signal, two parts appear: the specular echo (reflection) and reradiated echoes (elastic response). A data processing using a Fourier transform is applied to the totality or a part of the time signal. The modulus of this transform enables us to plot a spectrum. By considering the totality of the time signal, a backscattering
spectrum is obtained (monostatic set-up). As for the resonance spectrum, it is obtained from the time signal devoid of the specular echo and of the echoes related to wide resonances, i.e. the time windows (before the data processing) is translated along the time axis. Only thin resonances associated with circumferential waves are thus observable, on the resonance spectrum. The isolation of resonances is made by determining the frequential location of each peak [3].

2. STUDY IN LOW FREQUENCIES

The transducer characterized by a central frequency of 500 kHz turns round the target at a constant distance from its axis. For comparison purposes, the time signals are processed every degree, all the spectra are assembled in a matrix and after, the amplitude of the Fourier transform is normalized to 1. The backscattering spectrum as a function of the direction of the incident wave (θ_{inc}) is presented in Fig.3. The numerical results are then plotted as a gray level representation. The darker the gray shade is, bigger is the magnitude. Since no correction of the passband of the transducer is carried out, the amplitude of the spectrum is maximum in the vicinity of the central frequency of the transducer. Nevertheless, a dependence on shell orientation is noted. A comparison of the curves for the incident wave along the minor (Fig.3c, θ_{inc} = 90°) and major axes (Fig.3b, θ_{inc} = 180°) shows that the spectral amplitude is much more important when the curvature radius is largest. In these two cases, it appears, on the backscattering spectrum (Fig.3b-c), sharp transitions related to the resonances of the shell.

If we admit that circumferential waves travel round the shell in two opposite directions and form a standing wave at resonance frequency, it is possible to obtain a resonance spectrum by selecting a portion of the backscattering time signal with a gate and after, by applying a Fourier transform to the gated echo [2]. This method is used for each time signal recorded between θ_{inc}=0° to θ_{inc}=360° (angular step of 1°) with a receiver-transducer in rotation around this noncircular tube. The time position of the gate is common to the whole of the time signals and it is chosen correctly in order to isolate the elastic time response only (Fig.4b-f). All the resonance spectra are assembled and presented in Fig.4a. It appears, in this figure, “vertical” lines related to resonances, in particular, with the values 255 kHz, 304 kHz, 352 kHz, 407 kHz, 457 kHz, 510 kHz. This series of resonances is characterized by a quasi-constant space between two consecutive peaks; its value is between 48 and 55 (average ΔF_1 = 51 kHz). It is noted also for each frequency resonance, a modulation of amplitude as shown in Fig.4a. The mechanism of generation of these resonances according to the incidence angle is not well-known but in Fig.4a, it appears that, at resonance frequencies and for some values of θ_{inc}, the resonance is not formed (amplitude close to zero).

These early results obtained from a monostatic method show that resonances exist at particular frequencies. By using the bistatic method, it is possible to plot the angular scattered pressure for these particular frequencies. The experimental results depend on the shell orientation (θ_{inc}) because of the non-axisymmetry of the object. For this, three examples of incidence angle, θ_{inc} =45°, θ_{inc} =90°, θ_{inc} =135°, are only considered. The gray level representation of resonance spectrum (bistatic method) as a function of the observation angle θ_{azim} is presented for these angles in Fig.5.
Let us consider, for example, the resonance frequency $F = 457$ kHz. For this frequency, the same vertical sequence of black spots appears on the three gray level representations (Fig.5). This similarity, at $F = 457$ kHz, is confirmed with the angular diagrams for the three incidence angles. Indeed, in Fig.6a, a good agreement is noted between the angular diagrams obtained for $\theta_{inc}=45^\circ$ and for $\theta_{inc}=90^\circ$. The same thing is noted in Fig.6b for $\theta_{inc}=45^\circ$ and for $\theta_{inc}=135^\circ$. Moreover, it appears in Fig.6, and clearly in Fig.5, an angular area, in the vicinity of the major axis, where the amplitude diminishes strongly. This diminution of the amplitude is more important in the direction of the sharp part of the cross-section (Fig.6a-b). For this angular position, the variation of the curvature radius is extremely fast, and during the
circular rotation of the receiver-transducer, the waves radiated from the tube are not observed in the vicinity of this angle.

Fig. 4: Resonance spectra (monostatic method) in low frequencies
(a) gray level representation of resonance spectrum as a function of the incident angle $\theta_{\text{inc}}$
(b), (c), (d), (e), (f), backscattering spectrum respectively for $\theta_{\text{inc}} = 0^\circ, 45^\circ, 90^\circ, 135^\circ, 180^\circ$

3. STUDY IN HIGH FREQUENCIES

Experiments has been realized with a transducer of central frequency 2.25 MHz. Monostatic method and bistatic method are used to obtain backscattering spectra and resonance spectra. One of these results is presented in Fig. 7. The resonance spectrum as a function of the direction of the incident angle ($\theta_{\text{inc}}$) is plotted as a gray level representation.
In this figure, three areas where resonances are observable can be pointed. These areas are named Ar1, Ar2 and Ar3. The first area, defined by the frequency range 100 kHz-600 kHz and the incident angle range 0°-360°, concerns the resonances studied in the previous paragraph. This series of resonances presented in all range of the incident angle is characterized by the average frequential space equal to 51 kHz. Contrary to the area Ar1, the resonances selected by the areas Ar2 and Ar3 are isolated in a limited range of the incident angle, 0°-100° and 260°-360° for Ar2, 140°-220° for Ar3. The range of frequency is equivalent for these two areas, approximately 600 kHz-2500 kHz but the resonance frequencies are different and are equal to:

- 634 kHz, 786 kHz, 937 kHz, 1088 kHz, 1239 kHz, 1390 kHz, 1540 kHz, 1694 kHz, 1846 kHz, 1995 kHz for the area Ar2,
- 703 kHz, 840 kHz, 991 kHz, 1141 kHz, 1294 kHz, 1443 kHz, 1590 kHz, 1737 kHz, 1883 kHz for the area Ar3.

Only the resonances correctly isolated are listed above. The frequential space between two consecutive peaks of resonance (Fig.7) is quasi-constant for the resonances of the areas Ar2 and Ar3. The average frequential space of the two series of resonance are close: $\Delta F_2 = 151.2$ kHz for Ar2 and $\Delta F_3 = 147.5$ kHz for Ar2. The frequential space of two successive resonances of a given wave is proportional to the group velocity of this wave. Thus, the first series of resonances corresponds to a surface wave with a smaller group velocity, while the second and third series to waves with a higher group velocity.

4. DISCUSSION AND CONCLUSION

In the case of a circular cylindrical shell, it is well known that the experimental measurement of the mode number $n$ gives an identification of resonances [2,3]. This number is obtained from the angular diagram at a frequency of resonance. This diagram is a quasi-perfect “daisy” pattern where the number of lobes is equal to 2n. This datum is important because the knowledge of this mode for each resonance gives the experimental phase velocity and consequently, allows us to determine the type of wave among all the known waves (antisymmetrical Scholte waves $A_0$, symmetrical waves $S_0$ ...). For a noncircular cylinder, the identification (determination of the mode $n$) from the angular diagram of the scattered pressure becomes difficult (Fig.6a-b) because of the non-axisymmetry cross section.

In order to identify the waves propagating around the shell, the group velocities of these waves are considered. From the frequential space between two consecutive resonances and the perimeter of the tube $P = 36.5$ mm, the experimental group velocity is calculated by applying $C_G = \Delta F P$ [4]. Three series of resonances are isolated as shown in the previous paragraphs. The experimental group velocities of these waves are placed in Fig.8 and symbolized by $\bullet$, $+$ and $\times$ in this figure. Considering that the thickness of the streamline tube is thin, the experimental data are compared with the theoretical group velocities of the waves $A_0$ and $S_0$ propagating along an aluminum plate (thickness of 0.4 mm, $C_L = 6380$ m/s $C_T = 3100$ m/s) placed in vacuum. The second series ($+$) and the third series ($\times$) of resonances corresponds to the symmetrical waves $S_0$. The points $\bullet$ are close to the $A_0$ curve but it is well-known that, for thin circular tubes immersed in water [1], the wave $A_0$ is not observed experimentally because of its coefficient of reradiation in water. By comparison with a thin circular tube, we think that this wave is the antisymmetrical Scholte waves $A$. This wave is also observed in low frequencies for a circular cross-section tube immersed in water.
Fig. 5: gray level representation of resonance spectrum (bistatic method) as a function of the observation angle $\theta_{\text{azim}}$ for (a) $\theta_{\text{inc}} = 45^\circ$, (b) $\theta_{\text{inc}} = 90^\circ$, (c) $\theta_{\text{inc}} = 135^\circ$.

Fig. 6: Resonant angular diagrams (bistatic method), $F = 457$ kHz
Fig.7: Resonance spectra (monostatic method) in high frequencies gray level representation of resonance spectrum as a function of the incident angle $\theta_{inc}$.

Fig.8: — : Theoretical group velocities of the $A_0$ wave and the $S_0$ wave on a aluminium plate of thickness $e = 0.4\text{mm}$.
+ × ● : Experimental group velocities of waves on a streamline tube.

REFERENCES

Abstract: Squid are important organisms both ecologically and commercially. Acoustic scattering techniques can provide synoptic data on their distribution and abundance, and have the advantage of being efficient compared with traditional net sampling methods. However, knowledge of the scattering properties of squid is required to accurately convert the acoustic data into meaningful biological information. Tank experiments, compared with field measurements, allow better control over both the environment and the experimental animals, which are necessary for understanding the scattering of sound by individual squid, verifying existing models, and guiding new model development. A controlled laboratory backscattering experiment was conducted on live squid (Loligo pealeii) using broadband linear chirp signals (45-105 kHz) with data collected over the full 360 degrees of orientation in the lateral plane, in 1 degree increments. The scattered spectra often showed significant structure over the frequency band available at different angle of orientation, in addition to high levels of ping-to-ping variability. Pulse compression signal processing techniques were also used, revealing the following dominant scattering features: 1) at normal incidence, the front and back interface of the animal were resolved and 2) at off-normal incidence, the anatomical features in the head region were found to dominate the scattering. This information can serve as the basis for an accurate acoustic scattering model for squid.
Predictive Models for Low Frequency Acoustic Calibration Systems at the USRD

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Abstract: The state of an art for calibrating acoustic transducers at very low frequencies is by way of a confined and well understood environment. The Underwater Sound Reference Division (USRD) has three such calibration systems, called system K, J and L respectively. Each system has a cylindrical tube, of certain length and diameter that determine a cut off calibration frequency. System K operates in a standing wave mode condition. Systems J and L both operate in traveling wave mode conditions, where plane waves propagate from one end of the tube to the other. Optimally locating a calibrated transducer and an unknown within the tube provides the proper configuration for calibration. This paper documents the simulation tools to predict performance of aforementioned low frequency calibration systems. The mathematical model, the numerical coding and simulation results will be presented.
APPLICATION OF THE DECOMPOSITION OF THE TIME-REVERSAL OPERATOR IN NON-STATIONARY MEDIA

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Abstract: In shallow water acoustics, Time Reversal based techniques that assume a stationary medium, are often invalidated because of medium fluctuations. With the DORT method (French acronym for Decomposition of the Time-Reversal Operator), it is all the more true as the acquisition of the full array response matrix requires a lot of time. The DORT method uses the singular value decomposition (SVD) of the impulse response matrix to detect targets and retrieve their Green's functions. It was shown that in ideal conditions (stationary medium and point-like targets), the number of singular values of the K matrix is equal to the number of point-like targets in the medium and the singular vectors provide the phase laws focusing on each of them. Tank experiments involving a moving source or a waveguide perturbed by surface waves show that this assertion is no longer valid in non-stationary media. In particular, it is observed that several singular values can be associated with one target. A theory explaining the behaviour of singular values in non-stationary media as a function of the standard variation of the perturbation is presented. Analytical formulations of the singular values are in good agreement with experimental values acquired with a 64 elements ultrasonic array of 3.5MHz central frequency in a waveguide 500 mm long and 40 mm deep.

It is shown that the DORT method can be used in a non-stationary medium, in the case of a target moving vertically or a waveguide perturbed by surface waves. Furthermore, the analysis of the eigenvectors allows then to retrieve the positions of the target during the acquisition or the surface state above the target.

Keywords: time reversal, DORT method, non stationary medium, sea surface wave, moving target.
INTRODUCTION

Time reversal focusing in a wave-guide using a source receiver array (SRA) was demonstrated by several underwater experiments [1]-[3]. Taking advantage of the multiple reflections at the wave-guide interfaces, time reversal allows high resolution focusing. This high resolution has also been exploited for detection and separation of scatterer echoes in an ultrasonic wave-guide using the Decomposition of the Time Reversal Operator (DORT method). This method was applied in an ultrasonic water wave-guide demonstrating multi-target detection, selective focusing with high resolution [4] and separation of target echo from bottom reverberation [5]. It has also been the object of several studies for ocean applications [5]. As the DORT method inherits the strengths of the time-reversal in wave-guide environments, it inherits also its weakness to non-stationary media. Moreover, as the DORT method uses the decomposition of the full multistatic data matrix (MDM), it is all the more sensitive to perturbations of the media during the acquisition of the transfer matrix.

Numerous studies have investigated into the effects of non stationary medium on time reversal [6]-[8]. For example, Roux et al. show the degradation of the focal spot in a tank time reversal experiment when in presence of surface waves and Sabra et al. study the effect of the deformation of the time reversal mirror (TRM) on the quality of the focalization using modal decomposition.

On the other hand, no theoretical studies have been made on the effect of medium fluctuations during the MDM acquisition on the invariants of the Time Reversal Operator (TRO). Here we present a theory predicting the evolution of the eigenvectors (EV) of the TRO when the target is moving in the waveguide. This theory is then extended to the case where the surface of the waveguide is perturbed by surface waves. These results are validated by simulations and tank experiments at 3.5 MHz.

1. TARGET IN MOTION IN A WAVEGUIDE

We consider the case of a target in a waveguide at a distance L from the array moving vertically during the acquisition of the MDM. In order to simplify the problem, the study is limited to the quasi-static case. In general, the acquisition of the MDM is made column by column so, for an N transducer array, N acquisitions are needed. It is assumed that for each acquisition the target is at a different position. The position of the target during acquisition \( j \) is then given by:

\[
S^{(j)} = S + \sigma(0,0,Z^{(j)}) = (L,0,z_s + \sigma Z^{(j)})
\]

(1)

Where S is the mean position of the target in the waveguide, \( \sigma Z^{(j)} \) is the \( j \)\(^{th} \) displacement of the target around depth \( z_s \). We assume that the \( Z^{(j)} \) are independent and identically distributed and that their distribution follows a gaussian law with unitary standard deviation and zero mean.
Fig. 1: Experimental conditions of the acquisition of the MDM. The signal received on the array after the emission by transducer \( j \) gives the column \( j \) of the MDM after Fourier transform.

Then, in a waveguide, the pressure field on the transducer at depth \( z_s \), coming from a source at depth \( z_j + \sigma Z^{(j)} \) and distance \( L \) is written by using normal mode decomposition:

\[
p(L, z_j, z_s + \sigma Z^{(j)}) = P_0 \sum_{m=0}^{M-1} \Psi_m(z_j) \Psi_m(z_s + \sigma Z^{(j)}) e^{ik_m L} / \sqrt{k_m}
\]

with \( P_0 = \frac{ie^{-i\pi/4}}{\rho(z_0)\sqrt{8\pi}} \).

For simplicity, \( P_0 \) is omitted in the following equations and an element of the transfer matrix is written:

\[
K_{ij} = \sum_{m=0}^{M-1} \Psi_m(z_j) \Psi_m(z_s + \sigma Z^{(j)}) e^{ik_m L} / \sqrt{k_m} \times \sum_{m=0}^{M-1} \Psi_m(z_j) \Psi_m(z_s + \sigma Z^{(j)}) e^{ik_m L} / \sqrt{k_m}
\]

For small variances \( \sigma \), the Time Reversal Operator (TRO) \( H = K^\dagger K \) separated into two terms using a Taylor series expansion:

\[
H = \frac{1}{2\delta k_0} [A + B]
\]

where the dagger superscript denotes transpose conjugation and \( A \) corresponds to the zero\(^{th} \) order of \( H \) and \( B \) to the second order[10]. The zero\(^{th} \) order matrix corresponds to the motionless target at depth \( z_0 \). The eigenvector \( V_0 \) and eigenvalue \( \Lambda_0 \) of \( A \) have been derived in Philippe \textit{et al.} for a perfect shallow water waveguide:

\[
\Lambda_0 = \left( M - f \left( \frac{\pi z_s}{D} \right) \right)^2
\]

and
\[ V_0 = \frac{V_{\text{moy}}}{\|V_{\text{moy}}\|} = \frac{1}{\sqrt{M - f\left(\frac{\pi z_i}{D}\right)}} \left( \sum_{m=0}^{M-1} \Psi_m(z_i) * \Psi_m(z_j) * e^{i k_{mn} L} \right) \]

where \( V_{\text{moy}} \) is the Green’s function between the array and the mean position of the target [11].

The EV of \( B \) is then given by the formula

\[ V_i = \left[ (V_0)_j Z^{(i)} \right] \text{ with eigenvalue:} \]

\[ \Lambda_i = N^2 \pi^2 \sigma^2 \left[ \frac{1}{12} M + \frac{1}{3} M^3 + g\left(\frac{\pi z_i}{D}\right) \right] \|V_i\|^2 \]

\[ = N^2 \pi^2 \sigma^2 \left[ \frac{1}{12} M + \frac{1}{3} M^3 + g\left(\frac{\pi z_i}{D}\right) \right] \left[ M - f\left(\frac{\pi z_i}{D}\right) \right] \]

The same derivation can be done for the superior order Taylor series of the TRO, giving eigenvalues with a dependence on \( \sigma \) with a power law corresponding to the index of the eigenvalue, i.e., the first eigenvalue depends on \( \sigma \) with power zero, the second eigenvalue on \( \sigma \) with power 2, the third eigenvalue on \( \sigma \) with power 4, etc… This behaviour is confirmed by simple simulations presented in the next section. Moreover, the analytical formulation of the eigenvector gives us a mean to simply extract the positions of the target during the acquisition using only an estimation of the number of modes in the waveguide.

2. TARGET IN A WAVEGUIDE PERTURBED BY SURFACE WAVE

We want to extend the results presented in the previous section to the case of a waveguide perturbed by surface waves.

In the case of a shallow water waveguide with perfect interfaces, the formulation of the normal modes is simple.

\[ \Psi_m(z) = \sqrt{\frac{2 \rho_0}{D}} \sin(k_{zm} z) \quad \text{and} \quad k_{zm} = \frac{\pi}{2D} + m \frac{\pi}{D}. \]

with \( D \) the depth of the waveguide. To keep this simple formulation, the approximation of an adiabatic perturbation of the surface is necessary. This approximation is commonly used in modal theory and simply indicates that there is no transfer of energy from one mode to
another in the transition region [12]. Then the normal modes can be expressed as a function of the guide’s depth \(D(r)\) at the position of the target:

\[
\Psi_m(z, k_{zm}(r)) = \sqrt{\frac{2\rho}{D}} \sin(k_{zm}(r)z) \quad \text{and} \quad k_{zm}(r) = \frac{(m+1/2)\pi}{D(r)}.
\]

As is easily seen in these equations, the displacement of the surface directly above the target is equivalent to a displacement of the relative position of the target in a stationary waveguide.

\[
k_{zm}(r)z = \frac{(m+1/2)\pi z}{D(r)} = \frac{(m+1/2)\pi z}{D_0 + \Delta D(r)} \left(1 - \frac{\Delta D(r)}{D_0}\right)
\]

The derivation of the analytical formulation of the eigenvectors and eigenvalues of the TRO remain the same with 

\[
\sigma Z^{(i)} = -z_i \frac{\Delta D(L)^{(i)}}{D_0}
\]

with \(D_0\) the mean height of the waveguide and \(\Delta D(L)^{(i)}\) the displacement of the surface at distance \(L\) during acquisition \(i\).

3. SIMULATION AND EXPERIMENTAL RESULTS

Moving target: experiment compared to theory

Simulations and tank experiments have been performed to confirm the analytical formulation of the eigenvalues and eigenvectors given in the previous part. The simulation is based on modal theory in a perfect waveguide with Dirichlet limit condition for the surface and Neumann condition for the bottom. The number of modes taken into account in the simulation is equal to the number of modes in the tank experiment. This number is evaluated from the number of significant paths in the received signal. The parameters of the waveguide are \(D=40\) mm, \(L=500\) mm and a bottom made of Plexiglas. The MDM is acquired with a 64 transducer array with a pitch of 0.417 mm and 3.4 MHz central frequency. In this configuration, the number of different paths is equal to 6 meaning an approximate number of 44 propagating modes in the waveguide. When using the DORT method, the eigenvalues and eigenvectors of the TRO are simply obtained from the singular value decomposition (SVD) of the MDM. The two sets of different eigenvectors are then the eigenvectors of the TRO reception-wise and emission-wise and the singular values of the MDM are the square root of the eigenvalues of the TRO.

As can be seen on Fig.2, the analytical formulation of the singular values of the TRO given by Eq.6 fits the simulated results perfectly for small \(\sigma\). In this method, the limit for the perturbation is given by the lateral size of the focalization spot. Past this limit, the distribution of eigenvalues reaches an asymptote which is the distribution of white noise. In this case, the
focal spot has a width equal to $\frac{\lambda L}{6D} = 0.9$ mm. Although the non perfect reflection at the bottom of the waveguide changes the expression of the normal modes, the experimental results are in good agreement with the analytical results.

**Fig.2:** Evolution of the singular values of the MDM as a function of the variance of the displacement a) in a modal simulation b) in a tank experiment.

**Fig.3:** Comparison of the normalized displacement of the target with the norm of the corresponding elements of $\frac{Vp_h}{Vp_0}$.

Experimental results in the presence of surface waves

In the case of a waveguide perturbed by surface waves, the results are also very good. In the following, no numerical model taking surface waves into account was available so only tank experiment results are presented. The parameters of the waveguide are the same as the previous waveguide but the time window used for the DORT method is larger and comprises 9 different paths meaning a maximum number of modes $M=54$ at 3MHz. The surface is perturbed by a vibrator with controllable frequency and amplitude. Fig.4 a) shows the comparison between the evolution of the singular vector of the MDM and the analytical results while Fig.4 b) shows the frequency dependence for a given $\sigma$ implicitly due to $M$. It is important to see that in order to get rid of the unknown constant terms as the reflectivity of the target and $P_0$, the formulation of Eq.6 is normalised by the first singular value of the unperturbed MDM.
As in the previous case, the eigenvector contains the information concerning the relative displacement of the target and thus the surface state above the still target. The vibrator generates a sinusoidal wave of frequency 3 Hz and the frequency of acquisition is 1 kHz. In Fig.5, we compare the normalized displacement of the surface during the $i$th acquisition with the $i$th element of $\frac{V_{P_i}}{V_{P_0}}$. The conclusion is that we can retrieve the information of the surface state during the acquisition from the eigenvector of the perturbed MDM.

4. CONCLUSIONS

A theory allowing the derivation of the invariant of the time reversal operator in a non-stationary medium was introduced. It was applied to a moving target in a waveguide and generalized to a waveguide perturbed with surface waves. The analytical results were confirmed by simulations and tank experiments. It was shown that the eigenvector decomposition of the TRO separates the different order of the Taylor series of the elements of the MDM and that the analytical formulation of the second eigenvector allows for tracking of the target or the displacement of the surface of the waveguide above the target.
Thanks to the tank experiments the relatively strong approximations used in the derivations were validated for a Pekeris waveguides.

REFERENCES


CHARACTERIZATION AND CLASSIFICATION OF UNDERWATER ACOUSTIC MATERIALS

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\textbf{Abstract:} Passive acoustic materials play a major role in underwater acoustical system in different forms as sound absorbers, window materials, de-couplers etc. This paper deals with acoustical evaluation procedures of passive materials in the form of discs for its transmission and reflection properties and classification on the basis of evaluation.

\textbf{Keywords:} Acoustic material, impedance tube, echo reduction, insertion loss
1.0 INTRODUCTION

Acoustic materials have an important role in various systems like room acoustics, underwater acoustics, etc. This paper describes the methods and procedures for evaluating the passive materials used for underwater purposes. Some of the applications of passive materials in underwater acoustics are window materials for encapsulating the transducers, for fabricating baffles, domes, de-couplers, sound absorbers, etc. The evaluation includes measurement of transmission and reflection coefficients from which the absorption coefficient is derived.

There are two methods of evaluating the material underwater. One is in the form of panel in an open tank and the other is of disc type using water-filled impedance tube. The panel size required is about one wavelength at the lowest frequency of interest for reasonably diffraction error free measurements. Also, when the absorption coefficient is computed from the panel measurement, it may not be accurate because some of the sound energy is scattered or diffracted. At the development stage, it is neither worth nor economical to fabricate a panel for the evaluation at low frequencies. The alternative is to evaluate small size samples in water-filled impedance tube. Another advantage in using the impedance tube is that there are no losses due to scattering or diffraction and hence the absorption coefficient can be computed more accurately.

2.0 CHARACTERIZATION OF UNDERWATER ACOUSTIC MATERIALS

2.1 Impedance tube method

An impedance tube is a thick walled steel tube filled with water, instrumented with acoustic transducer positioned at the bottom. The material sample is made in the form of disc and is inserted inside the tube for the evaluation. The tube is positioned vertically and is filled with de-gassed distilled water. A transducer is positioned at the bottom of the tube. The transducer has many piezo ceramic elements stacked as a single unit and the tonpilz type design is used. The transducer is operated in a trans-receiver mode so that the same is used for generating the acoustic signal and for receiving the reflected signal. By this arrangement flushing of hydrophones along the tube wall is avoided and the rigidity of the tube is not compromised. The transducer is mounted in a mechanically de-coupled fashion to avoid the structural vibrations that could be transmitted to the medium inside through the tube wall. Any air bubble inside the tube will increase the compliance and acoustical pressure drops at that point. High internal finish of the tube helps in avoiding entrapping of air bubbles at the wall surface.

For the impedance measurement, it is necessary to have the speed of sound within the tube medium equivalent to that of free-field conditions. This can be approached to a closer value by using a thick walled tube and is calculated from the equation

\[
\frac{c'}{c} = \left[ 1 + \frac{2E_0 r}{E_w h} \right]^{-1/2}
\]

where \( c' \) is the corrected speed of sound, \( c \) is the speed of sound in the free-field conditions, \( E_0 \) is the bulk modulus of water in the tube, \( E_w \) is the Young’s modulus of tube wall material, \( r, h \) is the radius and the thickness of the tube.
The impedance tube has an upper frequency limit dictated by its diameter. Only plane wave exists up to this frequency and above which other modes are also generated. The measurement is restricted to plane wave propagation only. The upper frequency limit is calculated from the formula:

\[ Freq_{upper} = \frac{0.586 \times c'}{2a} \]

where \( c' \) is the speed of sound in the medium and \( a \) is the radius of the tube.

The tube at Materials and transducers simulated test centre (MATS) of NPOL has a length of 7000 mm and internal diameter of 200 mm. The upper frequency limit for this tube is 4000 Hz. The impedance tube with instrumentation set-up is shown in Figure 1.

![Impedance tube set-up for material evaluation](image)

There are two methods by which the material can be evaluated using impedance tube. The first is Standing Wave Ratio (SWR) method and the other is Transfer Function (TF) method. The standing wave method is the older of the two methods and is described in detail in reference [3].

The second one is Transfer Function method. In this method, continuous signals are converted into tone bursts and are used for evaluation. The acoustic waves reflected from or transmitted through the material are acquired in an oscilloscope and are processed subsequently. The multiple reflections are separated in time domain and the direct signals only are measured. The ratio of incident to reflected signal gives the echo reduction and the ratio of incident to transmitted signal describes the insertion loss of the sample. The other alternate is to use an impulse signal [4] and exploit the capabilities of commercially available FFT analyzers. The second option is easy to use and the phase also can be easily measured.
2.1.1 Echo Reduction (ER)

For echo reduction measurement, the sample is inserted into the tube from the top. The topside of the sample is aligned to the top surface of the tube. The water layer on top of the sample is removed carefully. The measurement is carried out employing impulse method. For this, a time and amplitude limited signal is used. A half-sine signal of 40-microsec width through a power amplifier is used for excitation. The same transducer receives the reflected pressure signal and is measured through a signal-conditioning amplifier. It is windowed in time domain to remove multiple reflections. The signal is de-composed in the frequency domain using the FFT analyzer and is stored in the memory. The sample is then removed and the reflection from the water-air interface is measured. Ratio of these two measurements is the echo reduction of the test sample. Time domain averaging is used for achieving better signal to noise ratio. In this method air medium on top of the tube is used as reference and the phase value is corrected by multiplying $j\omega$ twice with the final result.

2.1.2 Insertion Loss (IL)

For insertion loss measurement, the sample is positioned inside the tube at a depth of two metres from the top. The sample is suspended through a single stranded nylon string. A probe hydrophone is positioned just behind the sample to receive the transmitted signal. The hydrophone receives the transmitted signal through the test sample. The transmitted signal is de-composed in the frequency domain. Then the sample is removed and the hydrophone is positioned again in the same place to receive the signal in the absence of sample. The signal is measured again. Time averaging is used for achieving better signal to noise ratio. The ratio of these two is the insertion loss. A smaller size hydrophone is preferred to avoid the disturbance of pressure field.

3.0 CLASSIFICATION OF UNDERWATER ACOUSTIC MATERIALS

The material samples used for acoustical purpose can be classified into three categories namely, acoustically transparent, reflective or an absorber. Acoustically transparent material or a reflector will give same echo reduction values when measured in impedance tube since the backing medium is good reflector. To characterize the material fully, both echo reduction and insertion loss is to be measured. A procedure was formulated to classify the material from the result of the impedance tube measurement.

For a given sample, two measurements are made namely echo reduction and insertion loss in the frequency range of 500 Hz to 4000 Hz. The results are plotted on a single graph. The procedure for classification is described below.

3.1 Acoustically transparent material

For echo reduction measurement, the values for these types of samples will be close to zero dB. In the impedance tube measurement, the result will be similar to a reflector since the backing is air medium, which is a perfect reflector. However it can be identified as acoustically transparent from the insertion loss measurement. The insertion loss values also will be close to zero dB for these types of materials. The echo reduction and insertion loss measurement is plotted in the same graph and if both the results are close to zero dB, it is classified as acoustically transparent material.
3.2 Reflective type material

For this type of materials the values of echo reduction measurement will be close to zero dB. However the material provides higher insertion loss. If echo reduction is close to zero dB and the insertion loss is high then the material can be classified as acoustically reflective type.

3.3 Absorber type material

For this type of material, the sound energy is absorbed and hence the amount of reflection and transmission will be less. Hence the echo reduction and insertion loss values will be greater than zero dB depending on the amount of absorption.
The absorption coefficient can be easily computed from the IL and ER values. The material can be easily classified by the above mentioned procedure.

4.0 CONCLUSION

This paper describes the acoustic evaluation procedures of passive acoustic materials in a water filled impedance tube and a method to classify the same. This method is more suitable during the development stage and can be used to quickly compare between samples. Small samples with macro inhomogeneous contents cannot fully represent effective damping mechanism and hence such materials are to be evaluated using larger diameter tube with reduced higher cut-off frequency or using panel methods.

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Structured Session 29

Multibeam Echo Sounders and their Applications

Organizers: Mirjam Snellen & Dick Simons
USING MULTI-BEAM ECHO SOUNDER BACKSCATTER DATA FOR SEDIMENT CLASSIFICATION IN VERY SHALLOW WATER ENVIRONMENTS

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Abstract: In a recent work described in Ref. [1], an angle-independent methodology was developed to use the multi-beam echo sounder backscatter (MBES) data for the seabed sediment classification. The method employs the backscatter data at a certain angle to obtain the number of sediment classes and to discriminate between them by applying the Bayes decision rule to multiple hypotheses. This method is adopted and applied to very shallow-water applications. There are two issues when dealing with riverbed classification in shallow water. Shallow water depth results in a small beam-footprints and hence a small number of scatter pixels, which makes the classification results to be less discriminative. The significant bottom slopes will also affect the backscatter data and hence the classification results. We aim to handle these issues using the high resolution bathymetry and backscatter data. A methodology is developed to estimate the precise bottom slopes using the high resolution bathymetry data. Corrections are then applied to convert the arrival angle of the signal into the true incident angle and to compensate for the effect of the ensonified area. The high resolution backscatter data allows one to reduce the statistical fluctuations using an averaging procedure. The methodology will then be tested on a MBES data set from the river Waal, the Netherlands. The acoustic classification results are correlated with the mean grain sizes of the sediment obtained from core analysis of the grab samples. The dependence of acoustic backscatter classification on sediment physical properties is verified by observing a significant positive correlation coefficient of 0.70 between the classification results and sediment mean grain sizes of the grab samples.

Keywords: Multi-beam echo-sounders (MBES), Backscatter data, Slope correction, Bayes decision rule, Gaussian distribution.
1. INTRODUCTION

Multi-beam echo sounder (MBES) systems produce high-resolution bathymetry and backscatter data throughout the survey area. The bathymetry data is used to locate topographical features on the seafloor and to make topographical maps (e.g. harbour charts), which are the task of many hydrographic surveying institutes. The MBES backscatter data can be used to obtain information about the sediment composition and physical properties of the riverbed and seabed. Proper analysis and subsequent interpretation of the backscatter data is currently the task of many research institutes of acoustic remote sensing. The ultimate goal of acoustic classification methods is to remotely measure the physical properties of the surficial sediments such as porosity and mean grain size.

We use a classification method, which was developed by [1] for the seabed sediment classification. The method, based on the Bayesian decision rule, was applied to MBES backscatter data for the classification in a test area in the North Sea with well-known lithology. In order to adopt the method for shallow water applications, two issues need to be addressed. 1) The lower water depths result in smaller beam footprints and hence higher variances for the backscatter data. The discriminating power between sediments will accordingly decrease. 2) There exist significant bottom slopes which affect the backscatter data and hence the classification results. We elaborate these issues in detail and improve the results of the classification method for a shallow-water environment.

This contribution is organized as follows. In section 2, we briefly describe MBES classification method proposed by [1] and discuss our methodology to compute the bottom slopes using the precise bathymetry data and then to apply corrections to the backscatter data. We explain how to combine the classification results at different angles. In section 3, the acoustic classification results are presented for a recent data set carried out at Waal river. The classification results are then correlated with the mean grain size of grab samples. Section 4 concludes this paper.

2. MBES BACKSCATTER DATA

2.1 Classification method

In Ref. [1] a method was developed for seafloor sediment classification in deep water applications. The method fits a few Gaussian PDFs (normal distributions \(N\)) to the histogram of the backscatter data (\(BS\)) at a given grazing angle (each Gaussian PDF then represents one acoustic class):

\[
BS - f_{BS}(BS) = \sum_{i=1}^{n} c_i N(BS; \mu_i, \sigma_i^2)
\]

This is achieved by consecutively increasing the number of PDFs until a chi-square distributed test-statistic (on the residuals) becomes less than a certain critical value. In the preceding equation, \(\mu_i\) and \(\sigma_i^2\) are the mean and variance of the \(i^{th}\) PDF, respectively, and \(c_i\) is the contribution of the individual Gaussian functions to the total PDF. For further description of the method and the steps involved we refer to Ref. [1].

The main issue regarding the classification method is the normality assumption. This is valid for deep-water environments like seas and oceans where the beam footprint is large—it is proportional to depth—and hence many scatter pixels fall within the beam footprint. The central limit theorem states that the distribution of the averaged (over scatter pixels)
backscatter data in the beam footprint tends to a normal distribution if the number of scatter pixels $N$ is large enough. This holds obviously in deep water, while, in shallow water, $N$ is not large enough to use the central limit theorem. Therefore, we may use the averaged backscatter strength over the small surface patches. In addition, bottom slopes can be significant in the river environment considered in this paper. Therefore, two intermediate steps are added to the approach in Ref. [1]. These steps are as follows:

**Step I (correcting and averaging procedure):** In shallow water environments such as rivers, the number $N$ of the scatter pixels inside the beam footprint is not large because $N$ is proportional to the water depth. In order to restore the normality of the backscatter strength by means of the central limit theorem, one can use the average backscatter values over the small surface patches. Each patch consists of a few beams in the across-track direction and a few pings in the along-track direction. It also allows one to apply the slope corrections to the backscatter data, namely, correction due to the changes of the area of the signal footprint, and correction due to the true beam grazing angle. Therefore, for angle $\theta$ the ‘averaged corrected’ (over patches) backscatter data will be used. Further explanation is given in Section 2.2.

**Step II (combination of different angles):** The method in Ref. [1] takes observations from one single angle only. In practice, to use the full high-resolution mapping potential of the method, we consider multiple beams and individually perform the classification. This consequently allows one to obtain a continuous map over the whole area. The classification method at angles close to nadir (e.g. $\theta = 20^\circ$), however, becomes less efficient as the backscatter values of different sediment types have values close to each other. One remedy, followed in this contribution, is to first use the backscatter data at a few low grazing angles (e.g. reference angles of $\theta = 64^\circ, 62^\circ, 60^\circ$) and apply the classification method. This analysis gives the number $r$ of the sediment types, the means $\mu_i$, the variances $\sigma_i^2$ and the coefficients $c_i$. The nonlinear curve fitting in Eq. (1) is based on the bounds on the variables. Based on this information the curve fitting procedure is then executed and extended to all other angles ranging from $\theta = 60^\circ, 58^\circ, \ldots, 20^\circ$, but now i) for a fixed number $r$ of the Gaussian PDFs, where $r$ has been determined from the application of the classification method to the backscatter data of the low grazing reference angles (say $\theta = 64^\circ, 62^\circ, 60^\circ$), ii) by obtaining a good initial guess for the mean parameters, i.e. $\mu_i^0$ ($i = 1, \ldots, r$), of the backscatter data at the angle under study. This is achieved by using the mean values $\mu_i$ ($i = 1, \ldots, r$) of the reference angles, and equally shifted by the difference between the mean backscatter values at the angle under study (of entire histogram) and the mean backscatter values at the reference angles, and iii) by using more strict bounds on the mean parameters $\mu_i$ ($i = 1, \ldots, r$) for the classification of backscatter data at the angle under study (e.g. $\mu_i^l = \mu_i^0 - 0.5$ and $\mu_i^u = \mu_i^0 + 0.5$ dB). The bounds considered are still wide enough to compensate for the angular dependence of the statistical distributions for the backscatter data.

### 2.2 Local slope correction

The significant local slopes of the riverbed will affect the classification results. To compensate for these effects one has to estimate the along- and across-track slopes. Multi-beam echo-sounders (MBES) provide detailed and precise bathymetry information from which the local slopes (along- and across-track slopes) can be estimated using the least-squares method. This allows one to improve the seabed classification results by applying the corrections to the backscatter data. The literature has paid little attention to the question of
how such corrections should be taken into account. Two effects are discussed: 1) correction
due to the changes in the signal footprint (esonified area) to which the backscatter data
refers, and 2) correction due to the true beam grazing angle. Both corrections can be applied
when the along- and across-track slopes of the seafloor (riverbed) are available.

Estimation of slopes:
A discrete surface patch \( z_i = f(x_i, y_i), \ i = 1, ..., m \) includes a few angles around the central beam
angle (e.g. with deviation of one degree), where the angular dependence of the statistical
distribution of the backscatter data is negligible. Also, because the ping rate is high (40 Hz),
we may in addition include a set of neighboring pings to make a surface patch and hence to
be able to estimate the along- and across-track slopes. This results in a window (e.g. 0.5×0.5
m) that contains, say, \( m = 8 \times 7 = 56 \) beams. The average backscatter data and the average depth
in this small patch will be used. Using this strategy, to divide the area under survey into small
surface patches and to use the average values, one can i) compute the along- and across-track
slopes and correct for the true grazing angle and the backscatter data, ii) assure that the
normality assumption is achieved by means of the central limit theorem. This is a prerequisite
for using the classification method, and iii) decrease the variance and hence increase the
discriminating power between sediments. This makes the classification method more
discriminative.

A bi-quadratic function consisting of six unknown coefficients is used to model (estimate)
the surface patch: \( z = f(x, y) = a_0 + a_1 x + a_2 y + a_3 x^2 + a_4 y^2 + a_5 x y \). The depth measurements \( z_i \)
at the discrete points of the surface patch allow one to compute the unknown coefficients
\( [a_0, a_1, a_2, a_3, a_4, a_5] \) using the least-squares method. And a procedure called datasnooping
can be used to test for the presence of outliers in the bathymetry data ([2, 3]). This subsequently
allows one to obtain the along-track (\( x \)) and across-track (\( y \)) bottom slopes \( a_x \) and \( a_y \) (take
partial derivatives with respect to \( x \) and \( y \)). The slope angles \( \alpha_x \) and \( \alpha_y \) that the tangent plane
makes with the positive directions of \( x \) and \( y \) axes can accordingly be obtained. For more
information we refer to Ref. [4].

Grazing angle and backscatter corrections:
Suppose that the local surface is estimated as \( \hat{z} = \hat{f}(x, y) \). The angle between the normal
vector to this surface patch and the nominal receiving-beam direction (based on the flat
surface) is the true incident angle \( \theta_i \), which is a function of both \( \alpha_x \) and \( \alpha_y \). A general
formulation can be given for \( \theta_i \) as a function of \( \varphi \) (the nominal grazing angle), \( \alpha_x \) and \( \alpha_y \).
In a special case when \( \alpha_x = 0 \) it follows that \( \theta_i = 90^\circ - (\varphi + \alpha_y) \).

Another correction due to the local slopes \( a_x \) and \( a_y \) is the fact that the signal footprint
(esonified area) will change if the surface is not flat. For a sloping surface the signal
footprint is obtained as
\[
A_f = \frac{c TR\Omega_x}{2 \sin(\theta - \alpha_y) \cos \alpha_x} \tag{2}
\]
where \( R\Omega_x \) is the along-track resolution, \( c \) is the sound speed in water, and \( T \) is the
transmitted pulse length. The correction for the backscatter strength (in dB) can accordingly
be obtained (see Ref. [4] for further information).

3. CLASSIFICATION RESULTS AND DISCUSSIONS
The Waal river is a major river that serves as the main waterway connecting the Rotterdam harbor and Germany. For keeping the Waal river suitable for commercial activities bottom stabilizing measures are planned to counteract the subsidence and to keep the bottom more stable. To monitor the suppletion effectiveness at the river Waal, multi-beam echo sounder (MBES) bathymetry and backscatter measurements accompanied with extensive sediment grabbing were carried out in May 2008. The MBES used for the measurements is an EM3002, typically working at a frequency of 300 kHz for shallow water; the pulse length is 150 $\mu$s; the maximum number of beams per ping is 254; and the maximum ping rate is 40 Hz.

The bathymetry of this study area is shown in Fig. 1. Except for the flat area (sediment suppletion to prevent deformation in the outer part of the bend) in the middle of the area, the river exhibits significant bottom slopes. This section presents the results of the acoustic sediment classification based on the methodology developed by [1], which was modified in section 2 for shallow water applications. To assess the MBES classification results a comparison is made with the analysis of the grab samples.

We apply the classification method of section 2 to the averaged backscatter strength (over the small surface patches). The surface patches include a few angles around the central beam angle (with deviation of 1° as $\theta - 1° < \theta < \theta + 1°$). For such close angles, the angular dependence of the backscatter distribution can be ignored. Also a few consecutive pings (e.g. 7 pings) have been included, because the ping rate (40 Hz) is high in shallow water. This results in a small surface patch that contains, say, 56 beams. The precise bathymetry data over the patches allows correcting the backscatter data for the bottom slope effects. The ‘averaged corrected’ (over patches) backscatter data will then be used.

The number of seafloor types is unknown and needs to be determined. This is achieved by increasing the number of Gaussian functions to well describe the histogram of the averaged backscatter strength. A plot (not presented here) of a chi-square distributed test statistic versus the number $r$ of the Gaussian PDFs shows that this value is $r = 3$ (the ‘real’ number of riverbed types). Figure 2 shows the histogram and its best Gaussian fit for the averaged backscatter values at $\theta = 60°$. Three Gaussian PDFs, indicating three acoustic classes, are identified. The contribution of the PDFs is roughly 5%, 30%, and 65%. It is worthwhile mentioning that the classification method is independent of the absolute values of the backscatter data. In this respect, one may for instance think of the angular dependence of the backscatter data, or the intrinsic difference between the backscatter data of the left and right
transducers due to their calibration effects.

In order to explore the full high-resolution mapping potential of the method, one may consider using multiple beams instead of only one (section 2, step II). The ultimate goal of the acoustic classification method is to obtain a continuous map over the whole region, as for the bathymetry map. The classification map obtained from the averaged backscatter data using beam angles at $\theta = 64, 62, \ldots, 20^\circ$ is shown in Fig. 3, where the three sediment classes are presented by the colours red, yellow, and green. The green represents low values, the yellow represents intermediate values, and the red represents high values of the backscatter data. At a typical angle $\theta = 60^\circ$, the acceptance regions are as follows: $[-\infty \text{ to } -18.5] \text{ dB (class I), } [-18.5 \text{ to } -15] \text{ dB (class II), and } [-15 \text{ to } +\infty] \text{ dB (class III).}$

The ultimate goal of MBES data analysis is to transform the backscatter classification results into estimates of seafloor sediment properties such as mean grain size. The goal of the sediment grab sampling and grain-size analysis is to evaluate the potential correlation between the mean grain size and the results from acoustic classification. 28 grab samples taken at the central axis of the river and at both sides (70 m apart from the central path) were collected and analyzed for grain size distribution. The grab samples were washed, dried, and sieved through a series of mesh sizes ranging from 30 mm to 0.1 mm. The sieve sizes were converted into $\phi$ (phi) units using the equation $\phi = -\log_2 d$, where $d$ is diameter of grain in mm. Note that fine sediments have large $\phi$ values. Based on the comparison with the acoustic classification results it can be concluded that the areas of high backscatter values correspond to gravel (class III) and lower backscatter values correspond to sand (class I).

We now make a comparison between the classification results and the mean grain size of the samples. Our strategy is to use the results of core analysis for comparison and to perform a correlation analysis afterward. The mean grain sizes were sorted from fine to coarse sediment. Considering the grab samples as an unbiased representative for the whole area, the percentages of 5%, 30%, and 65% were then applied to the 28 samples. This corresponds to 1, 8, and 19 samples, respectively for sand, sandy gravel, and gravel areas. The classification results show good overall agreement with the ground truth information obtained from the

![Fig 2. Histogram (light bar) of averaged (over small surface patches) backscatter data corrected for local slopes, its three Gaussians (solid line), and its best fit (dashed line) at angle $\theta = 60^\circ$ over the whole area; left transducer (left); right transducer (right); number of Gaussian PDFs $r = 3$.](image)
core analysis (Fig 3 zoom-in part).

Most of the differences belong to the areas where the grab samples are in the boundary region of two classes. The dependence of acoustic backscatter techniques on sediment physical properties is examined using the Pearson correlation between mean grain size of the samples and the classification results. Larger grain sizes are expected to produce stronger backscatter for sandy and gravelly sediment. The Pearson correlation coefficient between the mean grain size and the results of the classification is 0.70. It indicates a high positive correlation (it is negatively correlated with $\phi$ values).

Due to the river currents interaction with bottom sediments, the rivers are dynamic environments and hence sediment distribution is highly heterogeneous. Ground truthing our classification results from core analysis of the sediments is prone to a few sources of uncertainty. We can at least mention: i) positioning error of the grab samples which is considered to be about 4-5 m, ii) the complexity inherent in ascertaining whether a single sample is representative of a larger region. This originates from the heterogeneity of the river sediment distribution, iii) a finite number of grab samples when assigning sediment types to acoustic classes. For example, the percentage of the class 1 (green) is 5% which leads to just one sample (if any) from 28 samples, iv) large standard deviation of backscatter data due to the shallowness of water, which leads to a small beam-footprint. This has been accounted for, to a large extent, due to the averaging procedure, and v) considering other physical properties of sediments rather than just the mean grain size.

4. SUMMARY AND CONCLUSIONS

Riverbed sediment classification using multi-beam echo-sounders (MBES) backscatter data is a promising approach. This contribution presented the methodology, developed in Ref. [1], to use the MBES backscatter data for the sediment classification in shallow water applications. The method employs the backscatter data to obtain the number of acoustic classes and to discriminate between them by applying the Bayes decision rule for multiple hypotheses. This
is achieved by fitting a series of Gaussian PDFs to the backscatter strength histogram. Since
the classification is done per beam, the method is considered to be independent of the
possible incorrect calibration effects and the angular behaviour of the backscatter data.

Shallow water depth results in a small beam-footprints and hence a small number of
scatter pixels per beam. That makes the backscatter data highly variable and consequently
the classification method becomes less efficient. In order to increase the discriminating power of
the classification results we used an averaging procedure over small surface patches (0.5 × 0.5
m). The high resolution bathymetry data provides precise bottom slope corrections to convert
the arrival angle of the signal into the true incident angle, and the high resolution backscatter
data allows one to reduce the statistical fluctuation in backscatter strength. Both make the
classification method more efficient.

The performance of the method was tested by using the backscatter data acquired in the
river Waal, the Netherlands (Fig. 3). For keeping the Dutch rivers suitable for commercial
activities bottom stabilizing measures are planned to counteract the subsidence. To monitor
the suppletion effectiveness, MBES measurements were used to apply the classification
method. Extensive sediment grab samples analyzed for the grain-size distribution were used
to evaluate the performance of the classification results.

We performed a correlation analysis. The dependence of acoustic backscatter
classification on sediment physical properties was verified by obtaining a Pearson correlation
coefficient of 0.70 between the classification results and sediment mean grain-size. Because
ground truthing the classification results from core analysis of the sediments is prone to a few
sources of uncertainty, further analysis of the correlation coefficient is required. We can in
particular think of a disattenuated correlation coefficient.

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INVERTING FOR BOTTOM PARAMETERS IN SHALLOW-WATER SOFT SEDIMENT ENVIRONMENTS USING MBES BACKSCATTER STRENGTH

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Abstract: Shallow water naval operations require detailed knowledge of the environmental properties. In addition to parameters such as water depth, knowledge about the sediment properties is of high importance for a wide range of operations. In this context, the MREA BP'07 experiment was carried out in the Mediterranean Sea in 2007. Measurements employed a large set of sensors, thereby providing all information required to fully describe the environment. This paper focuses on multibeam echosounder (MBES) measurements, which were taken not only to provide information about the water depths, but also to provide the backscatter strength as a function of angle. These backscatter data are employed to infer sediment parameters. To this end, a comparison of measured and modeled backscatter strengths is conducted. Use is made of a model that accounts for both scattering at the water-sediment interface and scattering and scattering at the inhomogeneities in the sediment, i.e. volume scattering. In practice, the measured backscatter strength values are affected by the imperfect MBES calibration. This impedes a direct model-data comparison, unless the effects of miscalibration are eliminated. Therefore, a calibration curve is derived by optimizing spectral strength and volume parameter, according to known mean grain sizes at bottom grab positions. After having corrected the measured backscatter strength for these systematic effects, inversions are carried out to estimate the sediment parameters (grain size, spectral strength, and volume parameter) at various locations in the research area.

Keywords: backscattering, seafloor classification, inversion, multibeam echosounder
1. INTRODUCTION

Shallow water naval operations have become highly important for a wide range of applications, for instance mine hunting operations. Such applications require detailed knowledge about the characteristics of the underwater environment, including bottom properties.

Mainly, two types of bottom properties can be discriminated according to [1]: physical properties (e.g., grain size) and geoacoustical properties (e.g., backscatter strength). A direct measurement of the physical bottom parameters requires bottom samples. Collecting these samples is a very time consuming process. Therefore, often alternative methods are followed. They commonly employ measured geoacoustic properties, for example, obtained by a multibeam echosounder (MBES), and assign these to physical properties by, for instance, modelling [2]. Physical bottom parameters then can be inverted for by optimizing the fit between the model predictions and the measured backscatter strength. Alternatively, a relatively limited set of bottom samples can be employed to deduce the relation between measured geoacoustic properties and, e.g., mean grain size.

In this paper the measurements consist of backscatter strength as a function of angle. Backscatter models are well-proven in areas with common sediment types (e.g., sand, gravel). However, research on the backscatter modelling in areas consisting of very fine grained sediments and colloids is currently lacking.

One such site of very fine grained sediments is the shallow water region in the Mediterranean Sea, south-east of Elba Island. Data analysed in this paper are collected during the multidisciplinary MREA BP’07 experiment (as described in [3]) that was carried out in this region.

This paper is organized as follows. First, a brief description of the data, taken during the MREA BP’07 experiment, is given in section 2. Section 3 focuses on the modelling of the MBES backscatter strength measurements for inverting physical parameters of the seafloor. Thereafter, results are discussed in section 4. Finally, results are summarized and embedded in the context of the research project in section 5.

2. THE MREA BP’07 EXPERIMENT

In the context of the Marine Rapid Environmental Assessment (MREA) project, the MREA BP’07 experiment was carried out in the Mediterranean Sea during two weeks in the spring of 2007. The survey focussed on a shallow-water environment between 10.7° and 11.0° eastern longitude and between 42.5° and 42.8° northern latitude. For survey details we refer to [3]. A part of this area was already surveyed during former experiments, such as the Yellow Shark (YS) experiments (see [4], [5]). These experiments provide additional information for interpreting the data.

Measurements that were taken during the MREA BP’07 experiment employ a large set of sensors (e.g. multibeam and three-frequency single beam echosounders, seismic profilers, bottom grabber), thereby providing all information required to fully describe the environment. In this paper we focus on MBES data and bottom grab samples.
MBES data were taken by a SIMRAD EM3000D that is mounted on the HNLMS Snellius, a hydrographic vessel of the Royal Netherlands Navy. This MBES operates at 300 kHz with a ping rate of 0.2 s to 0.3 s. Within an opening angle of 130° up to 245 beams are formed, dependent on the water depth. Per beam information about the depth and backscatter strength is obtained.

 Depths values vary within the wide range from 0 to 130 m. Shallowest parts occur in the north-east, near to the Italian coast. From these regions on, depth increases with distance to the coast, resulting in isobaths that tend to run parallel to the coastline.

 MBES backscatter strength data also tend to vary with the distance to the coast. From the shallow to the deep part, backscatter strength first decreases up to about 50 m depth, before it increases again. This behavior can be seen in the map of backscatter classes (Fig.1), as presented in [6]. The classes represent backscatter strengths at 44° beam angle, an angle that is sufficient small to obtain measurements at all present water depths, but still sufficient large to cover a large footprint area.

![Fig. 1: Backscatter strength classes in the MREA BP’07 area.](image)

Additionally, 24 bottom grabs were taken with a Hamon grabber to provide ground-truthing information. These bottom grabs were analyzed at TNO, in the Netherlands. They show the occurrence of very fine grained sediments over the entire area. Mean grain sizes $M_z$ are calculated according to [6] as the average of the three proportions $d_{16}$, $d_{50}$, and $d_{84}$. Hereby, $d_x$ denotes the grain size in $\phi$ units, at which $x$ % of the grains in the sample are smaller. A map of mean grain sizes is given in Fig. 2.
3. INVERSION OF SEAFLOOR PARAMETERS FROM MBES BACKSCATTER STRENGTH

The backscatter strength of an acoustic signal that is returned from the seafloor contains information about the properties (e.g., density and roughness) of the sediments. This information is expressed in the varying shape and value of the measured MBES backscatter strength curves per swathe. Therefore, MBES backscatter strength measurements can be used for assessing the sediment composition of the seafloor.

In order to link acoustic measurements to a set of related seafloor parameters, a geoacoustic model is employed according to [7].

3.1. The backscatter strength model

The present model [7] of the backscatter strength \( BS_{\text{mod}} \) at a certain angle \( \theta \) distinguishes between a contribution from a backscatter cross section that is due to interface roughness \( \sigma_r \) and one that is due to volume scattering \( \sigma_v \).

\[
BS_{\text{mod}}(\theta) = 10\log_{10}[\sigma_r(\theta) + \sigma_v(\theta)]
\] (1)
Parameters that contribute to $BS_{mod}$ are listed in Table 1. While the interface roughness scattering mainly varies with the spectral strength $w_2$, the volume parameter $\sigma_2$ is the crucial parameter for the volume scattering.

<table>
<thead>
<tr>
<th>Seafloor parameter</th>
<th>Symbol</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sediment – water ratio of mass density</td>
<td>$\rho$</td>
</tr>
<tr>
<td>Sediment – water ratio of sound speed</td>
<td>$\nu$</td>
</tr>
<tr>
<td>Imaginary to real wavenumber ratio</td>
<td>$\delta$</td>
</tr>
<tr>
<td>Sediment volume scattering cross section to attenuation coefficient ratio</td>
<td>$\sigma_2$</td>
</tr>
<tr>
<td>Exponent of the bottom relief spectrum</td>
<td>$\gamma$</td>
</tr>
<tr>
<td>Strength of the bottom relief spectrum</td>
<td>$w_2$</td>
</tr>
</tbody>
</table>

Table 1: Seafloor parameters.

The interface roughness cross section differs according to changes both in sediment softness and angle. Therefore, a set of three approximations is applied, which all cover a certain angular range and sediment type: 1) the Kirchhoff approximation for smooth and moderate rough bottoms at grazing angles between 40° and 90°; 2) the composite roughness approximation for smooth and moderate rough bottoms at grazing angles below 40°; and 3) the large-roughness approximation for gravel and rock bottoms. The last approximation, however, does not contribute to the model used here, since only soft sediments are present in the MREA BP’07 area. All contributions are a function of the roughness spectrum. Hereeto, [7] uses the following isotropic relief spectrum $W_2(K) = (h_0K)^{\gamma}w_2$, with $K$ denoting the wave number of the bottom relief and $h_0$ a reference length of 1 cm.

The sediment volume scattering cross section is calculated from Eq. 2 by applying corrections for slope effects and shadowing. Hereby, $R(\theta)$ denotes the amplitude refraction coefficient at grazing angle $\theta$. Furthermore, $P(\theta)$ is calculated according to $P(\theta) = [\kappa^2 - \cos^2(\theta)]^{1/2}$ with $\kappa = (1+i\delta)/\nu$ and $\text{Im}\{P(\theta)\}$ is its imaginary part.

$$\sigma_{\rho_\theta}(\theta) = \frac{5\sigma_2\Big|1 - R^2(\theta)\Big|^{2}\sin^2(\theta)}{\nu \ln10|P(\theta)|^2 \text{Im}\{P(\theta)\}}$$

(2)

3.2. Calibration

Before employing a comparison between model and measurements for the purpose of inverting bottom parameters, calibration has to be applied. The calibration of the measured backscatter strength data is necessary, since these data show systematic effects over the entire angle range. In order to eliminate these systematic effects, a calibration curve is derived and is added to the MBES backscatter strength measurements $BS_{meas}$.

To this end, backscatter measurements are selected from regions in which sufficient groundtruthing information is available. These are regions in which bottom grabs are taken. Around each grab position a sequence of 100 pings is selected. The average of these pings is then employed in an optimization step. In this step the mismatch between the model and each of the averaged measurements is minimized with regard to the spectral strength $w_2$ and the volume parameter $\sigma_2$. The other model parameters of $BS_{mod}$, thereby, are adapted to the measured mean grain size $M_2$ at the current bottom grab position.
The resulting calibration curve should reflect systematic effects only and should therefore be independent of the area, i.e. bottom type.

For the reason of reliability, only grabs with a mean grain size of $M_z = 9\phi$ or less are considered for calibration purposes. This restriction is made, since the model was basically introduced for an $M_z$-range of -1 to $9\phi$. For the MREA BP’07 experiment this leads to a selection of four bottom grab positions (see Fig. 2).

In the case no systematic effects are present the difference between the optimal model, including the optimized parameters $w_2$ and $\sigma_2$ as well as the measured $M_z$ value, and the measurements should be zero. From Fig. 3 it can be seen that this is not the case. At all four considered grab positions the difference curves (thin, colored lines) are shifted from zero and vary strongly, but in a comparable way. Therefore, the calibration curve $C$ is taken as the average of these difference curves (thick, black line). The calibration curve itself is independent of grain size.

![Fig. 3: Differences between modelled and measured backscatter strengths at bottom grab positions for $M_z = 7\phi$ (thin, gray line) and $M_z = 9\phi$ (thin, colored lines). The calibration curve (thick, black line) is taken as the average of these curves.](image)

3.3. Inversion

Estimating seafloor properties by inversion is conducted for three parameters, which are assumed to vary strongly with seafloor type. These parameters are the mean grain size $M_z$, a scaling factor $w$ for the spectral strength $w_2$, and the volume parameter $\sigma_2$. These parameters are known to show strong variations (see [7]). Consequently, actual values might deviate significantly from default values. For the remaining parameters empirical relations, expressing them as a function of mean grain size, are employed.
A maximum agreement between the calibrated measurements $BS_{meas} + C$ and the model $BS_{mod}$ at interpolated angular positions $k$ is obtained by optimization. Hereto, the following energy function $E$ is minimized:

$$E = \sqrt{\frac{\sum_k \left( BS_{mod,k} - (BS_{meas,k} + C) \right)^2}{\sum_k (BS_{meas,k} + C)^2}}.$$  \hspace{1cm} (3)

4. RESULTS

Inversion of the three parameters, mean grain size, spectral strength, and volume parameter, has been conducted by employing the model of backscatter strength values as described in section 3. Hereto, several regions in the MREA BP’07 research area have been selected based on the classification of the backscatter strength at a beam angle of 44° (Fig. 1) and the mean grain size distribution of the bottom grabs (Fig. 2).

Figure 4 shows the map of the inverted volume parameter $\sigma_2$. Patches of comparable $\sigma_2$ values persist throughout the entire area. They tend to follow the pattern of the bottom grab samples and the backscatter strength classes. Overall, $\sigma_2$ values are slightly higher than expected. The default value for $\sigma_2$ in the model is 0.002 for fine grained sediments [7] such as in the experimental area. Extreme high volume parameters occur in the shallowest parts close to the Italian coast. Such high values might result from the occurrence of gas deposits in the sediments as observed on seismic profiles [3].

Fig. 4: Inverted volume parameter at several spots in the MREA BP’07 area.
However, the two other parameters $M_z$ and $w_2$ (not shown here) do not show such clearly separable patches as the volume parameter does. The inverted mean grain size is scattered throughout the entire area, thereby underestimating the actual mean grain size values of the slightly coarser sediments and overestimating the actual mean grain size values of the softest sediments. The inverted spectral strength appears to be less scattered, but shows values at the outer search boundary for almost each measurement taken in regions that feature very soft sediments ($M_z = 9\phi$ or less). Only in the shallowest part, where sediments with mean grain sizes of 7 to 9\phi occur, two patches of different $w_2$ values are distinguishable. Research on these parameters is ongoing.

5. CONCLUSION

Three bottom parameters mean grain size $M_z$, spectral strength $w_2$, and volume parameter $\sigma_2$ have been inverted for, using a model that accounts both for the interface roughness scattering and the volume scattering of the seafloor sediments. Thereby, valuable results are obtained for $\sigma_2$, which match the pattern of backscatter strength classes. The range of applicability of the model can therefore be extended to grains even smaller than $M_z = 9\phi$, at least when considering the inversion of the volume parameter.

Neither $M_z$ nor $w_2$ can be inverted reasonably for the MREA BP'07 soft sediment environment. These model parameters seem not be suited to properly describe (resolve) small differences in the backscatter properties of such fine grains. From this we can conclude that volume scattering plays a dominant role in interpreting the backscatter strengths in soft sediments.

Additional research has to be carried out with regard to the search boundaries of the spectral strength in order to make the model capable to discriminate between the softest sediments. Furthermore, fine grained regions might also be considered for groundtruthing.

6. ACKNOWLEDGEMENTS

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DISCRIMINANT ANALYSIS IN IMAGE-BASED SEABED CLASSIFICATION

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Abstract: Acoustic seabed classification of images from multibeam or sidescan sonar systems usually starts with dividing the image into regions or sub-images. The backscatter amplitudes in these regions can be used directly for classification, but discrimination can be improved if features (statistical attributes) are calculated from them. Classifying for the first time with a particular sonar and a particular set of seabed types is done either manually by expert interpretation or by an unsupervised computer-based process. Unsupervised classification isolates the naturally occurring acoustic classes in the set of survey images by a clustering method, optionally preceded by dimension reduction. If ground truth is available, from sediment samples or photographs, it can be used to label the acoustic classes or to confirm the expert interpretation. Discriminant analysis is a process for extending this initial classification to new areas. Sets or combinations of features from each sediment class are identified – this can be called the training set of classes. Each new record is then assigned to the class to whose centre it is closest, typically using a Mahalanobis metric. We report on discriminant analysis of a set of multibeam images, in each case after compensating the image for geometrical effects with the methods in QTC MULTIVIEW\textsuperscript{\textcopyright} (Quester Tangent Corporation). Some features were found that offer good discrimination among the seabed types, while others were highly correlated or less powerful. The list of best features varies among data sets. Records that do not fit well into any class are readily identified – if they are geographically cohesive this indicates that another seabed type should be added to the training set.

Keywords: acoustic classification, seabed classification, discriminant analysis, Mahalanobis
1. INTRODUCTION

Image-based acoustic seabed classification with multibeam and sidescan sonar systems starts with dividing the image into regions or sub-images. Each region is a set of echo amplitudes from which first- or second-order features (statistical attributes) are calculated. First-order features such as the mean are independent of data order, while second-order features rely on the arrangements among neighbouring values. Examples of second-order statistics are those derived from grey-level correlation matrices. Using features rather than the amplitudes themselves is an aid to discrimination.

There are two routes from features to class maps. Unsupervised classification isolates the naturally occurring classes in the set of survey images by a clustering method, optionally preceded by dimension reduction in which only a few principal components are usually kept. No ground truth is required or used on this route. Another route becomes available if some images have been classified and ground truth is available, because they can constitute a training set.

Developing a training set is more awkward with imaging sonars than it is with single-beam sounders. A sounder-equipped boat can hover atop a region of known seabed type, generating echoes and either recording their time series or calculating features from them with analogue or digital processing. Images cannot be acquired in this way because ship motion is essential to the imaging process. It would be rare to find a patch of seabed large enough to fill an image yet homogenous enough to serve as representative of a seabed type. Thus, training sets for image-based seabed classification are generally obtained from images that include several seabed types. Acoustic classes in these images have to be assigned by unsupervised classification or by expert interpretation. This paper describes two methods of classifying other areas based on the classes in a suitable training set.

Discriminant analysis is a statistical method that finds sets of, or linear combinations of, features that best separate a data set into classes. It operates in feature space, the space in which each feature has its own axis and the records of the data set, one from each sub-image, are a cloud of points. With the simplest classification methods, using only one or two

![Diagram of classification processes in this paper. QC stands for quadratic classification, which is the assignment of points to classes using a squared metric in feature space.](image)
features, discriminant analysis could be nothing more than choosing borders on axes in a one-
or two-dimensional plot. A more complicated method is selecting features from a larger
feature set based on how well they discriminate with a particular data set. The general method
involves rotations in feature space, that is, finding weights for each feature that optimize class
assignments. The most popular method of finding these weights is principal components
analysis (PCA).

In this paper, two methods of discriminant analysis are compared using a multibeam data
set. The features used here are those of QTC MULTIVIEW™ (Quester Tangent Corporation).
They were designed to capture image texture and amplitude [1]. The MULTIVIEW software
suite emphasizes quality control and compensation of every pixel for range and grazing angle
before features are made from rectangular sub-images. Records, each with their vectors of
features, are then assigned to classes by unsupervised classification using PCA to reduce
dimensionality and QTC’s objective clustering method [2]. Assignment is done iteratively
with quadratic classification and a Bayesian metric (these terms are explained below).

Fig. 1 is a flowchart of the classification methods used here. Unsupervised classification
with MULTIVIEW is above the dashed line, and the two methods of discriminant analysis
are beneath it. Both start with a training set that is actually a subset of the full survey (three
different sets were used). This training set is classified afresh, unsupervised. The first of the
two methods uses the lists of records in the training set with their assigned classes. This is the
method that classifies with just a few features in a simple feature space (left side of Fig. 1).
The second method uses the PCA weights and the positions of the cluster centres in feature
space, both of which were produced by classifying the training set. Both methods use
quadratic classification to assign all the points in the full data set to classes. These class
assignments are then compared with the initial unsupervised classification.

The method that uses only a few features has the benefit that its operations can be readily
interpreted. For example, if the small feature set it uses in a particular case includes the
amplitude standard deviation, it is easy to appreciate an image characteristic that is driving
classification. Does making good class maps with a small feature set mean that classification
of unknown areas can be done that easily? No, because the features chosen for the small set
vary between surveys. It is not possible to predict in advance which particular features will
discriminate well.

2. SURVEY DATA AND INITIAL CLASSIFICATION

A multibeam survey in the approaches to Portsmouth Harbor, New Hampshire, USA, was
done on 5 and 6 Nov 2000. Bathymetric and image data were collected along 90 lines
covering about 1.6 km². A Reson 8101, which transmits at 240 kHz, produced images in
sidescan mode with a range setting of 100 m (swath width 200 m).

The initial unsupervised classification was done with QTC MULTIVIEW, giving the map
of acoustic classes in the middle panel of Fig. 2. This set of classes serves as the reference set
against which classes generated using training sets are assessed. In Fig. 1, these two types of
classification are separated by a dashed line.

This MULTIVIEW process does not use any information about sediments in the survey
area, and is therefore an example of unsupervised segmentation of an area into acoustic
classes. Often the class map is more useful with geological or biological labels attached to the
acoustic classes. Some non-acoustic information, such as photographs, diver observations, or
grab samples are required to attach labels. The benefit of acoustic classification, and the
primary motive for it, is that far fewer non-acoustic samples are needed to label acoustic
classes than would be needed to make a detailed class map of an area without acoustics.
Grab samples of the surficial sediments within the survey area were obtained from 22 sites in 1995. Based on size fractions, each was assigned to a Folk class [3]. Four of the six acoustic classes could be labelled with grab results because they had grab sites in their areas. Class 1’s area contained no grab samples, but its location on sun-illuminated bathymetry (Fig. 2) showed that class 1 is rock. There is no information with which to assign a geological label to class 3. Table 2 shows these geological labels. A dominant feature of this area is the large field of sand and gravel waves centred on (361200, 4769600) in which classes 4 and 5 both appear. At frequencies as high as 240 kHz, acoustic backscatter is controlled by surface roughness, and areas with similar grain sizes can be put into different acoustic classes if their surfaces differ.

![Fig. 2: Reson 8101 sun-illuminated bathymetry (left) of approaches to Portsmouth Harbor, New Hampshire, USA. Acoustic classes are from QTC MULTIVIEW (unsupervised, middle panel) and from quadratic classification using only features 2 and 3 with 25% of the data for training (right panel). The two class maps disagree at 9% of the points, mainly near class boundaries. Locations are in WGS84 and UTM zone 19.](image)

<table>
<thead>
<tr>
<th>Class</th>
<th>Colour</th>
<th>Area covered</th>
<th>Geological label</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Magenta</td>
<td>5%</td>
<td>rock</td>
</tr>
<tr>
<td>2</td>
<td>Brown</td>
<td>9%</td>
<td>sand</td>
</tr>
<tr>
<td>3</td>
<td>Blue</td>
<td>1%</td>
<td>unknown</td>
</tr>
<tr>
<td>4</td>
<td>Cyan</td>
<td>3%</td>
<td>muddy sand, gravelly sand</td>
</tr>
<tr>
<td>5</td>
<td>Sand</td>
<td>15%</td>
<td>sand</td>
</tr>
<tr>
<td>6</td>
<td>Green</td>
<td>67%</td>
<td>gravelly sand, gravelly muddy sand</td>
</tr>
</tbody>
</table>

Table 1: QTC MULTIVIEW™ classes for the Reson 8101 survey in the approaches to Portsmouth Harbor, coloured as shown in Fig. 2.
3. CLASSIFICATION METRICS

If the inter-class borders in feature space are defined by simple equations, by drawing
boxes, for example, classification of new records is straightforward. One simply locates a
new record in feature space and asks which box it is in. This type of technique, though, is
often too simple. Here we consider the general case of feature vectors that have been divided
into $k$ clusters. The number of dimensions in feature space equals the length of each feature
vector $x$. These vectors may have been rotated through a process such as PCA. The
Mahalanobis distance squared, $D^2_k(x)$, between $x$ and a cluster centre defined by mean $\bar{x}_k$,
and covariance $C_k$, is

$$D^2_k(x) = (x - \bar{x}_k)^T C_k^{-1} (x - \bar{x}_k)$$  \hspace{1cm} (1)

An optimal classifier is to assign each record to the class, $k$, for which this squared distance is
minimum.

Assuming the feature values are normally distributed along each feature axis, the Mahalan-
obis distance squared is chi-squared distributed with the number of degrees of freedom equal
to the dimensionality of feature space. This allows outliers to be identified in a statistically
sound way.

It is often observed that a few seabed types dominate a survey area. In these cases,
populations become part of an improved optimal classifier [4], which is to assign each point
to the class, $k$, for which $Q_k$ is minimum, where

$$Q_k(x) = (x - \bar{x}_k)^T C_k^{-1} (x - \bar{x}_k) - 2 \log p(\omega_k) + \log |C_k|$$ \hspace{1cm} (2)

Here log denotes natural logarithm and the superscript T denotes transpose, as it does in Eq.
1. The probability of cluster membership, $p(\omega_k)$ is the chance that a new point should be in
cluster $k$. In the absence of prior knowledge of cluster populations, assignments to any of the
$K$ clusters are equally likely and $p(\omega_k) = 1/K$. If cluster populations can be estimated, from a
training data set or from a previous iteration of the classification process, new assignments
can be based on the size of the clusters, in which case $p(\omega_k) = N_k / N$, where $N_k$ is the number
of members in the $k^{th}$ cluster and $N$ is the total number of observations. These options are
called equal and estimated priors, and the variable $Q$ is called a Bayesian metric.

4. SMALL FEATURE SETS FOR QUADRATIC CLASSIFICATION

Now we move to extending classification into a new area, using a training set. One of the
methods presented here uses a small feature set, just two or three features. This section
explains how those features are chosen. We start with the same 132 features that were used in
the unsupervised process. These are not the only possible choice. After manual classification
of a training area, for example, features could be calculated specifically for classifying new
areas. Why use so few features? A continuing criticism of statistical classification methods is
that it is not evident which characteristics of an image are important in classification. Classification with just a few features would allow interpretation.

To select a small feature set, we start by investigating how well just a single feature discriminates between pairs of classes from the training set. If histograms of occurrences, of which Fig. 3 is an example, show little overlap between classes, that feature is a candidate for a short feature set. A list of candidate features from all the 15 possible pairs of six classes was then pruned to those features that appeared most often. Starting with the best of these features, each record of the training set was classified with Eq. 1, and the overall agreement found between these new class assignments and those of the training set. Features were added until the overall agreement reached a plateau. This feature set was then taken as the small set to be used to classify new areas.

With one of the training sets, for example, it was found that many of MULTIVIEW’s 132 features discriminated well between pairs of classes when used individually. Between classes 2 and 3, for example, 59 of the features discriminated well enough that their feature histograms (Fig. 3 is an example) had less than 5% overlap. Between classes 1 and 6, on the other hand, only feature 2 discriminated that well. The list of top performing features started with 2, 3, and 61. Classifying records from all six classes, feature 2 alone gave 63% agreement with classes in the training set, adding feature 3 raised that to 91%, and adding feature 61 and others to those two gave no further improvement. Therefore classification of the entire data set was done with only features 2 and 3, which are the standard deviation and the skewness of the amplitudes in each sub-image. Supporting this choice is that their correlation was only 6%. It should be noted, though, that other pairs of features that include GLCM features classified almost as well.

5. CLASSIFICATION AGREEMENT

Six sets of classes were produced by discriminant analysis with quadratic classification to be compared with the initial unsupervised classification. The six results are from two methods for each of three training sets. Two of these were 10% and 25% of the records in each class with a minimum of 500 records per class. These were the first records in chronological order, which was a type of random selection because the survey sequence was not particularly systematic. The third training set was a better mimic of a real use of extended classification, in that the training set was from an east-west swath across the survey area. This swath was chosen to include areas in each of the six classes. All three training sets were classified individually using the MULTIVIEW process.

As is often the case in seabed classification, little can be said about accuracy. The MULTIVIEW map is completely accurate when its labelled classes are judged against the grab samples. All the other maps are visually quite similar. The differences between them are
more detailed than the spatial resolution of the data from the grab samples. Indeed, none of the neighbourhoods around any of the 22 grab samples was changed to a different class by any of these classification methods. The topic here, therefore, is the extent of agreement between class maps, not their accuracy.

Two methods based on quadratic classification were used to extend a training set to class assignments over the full survey area. In one method, a small set of features was selected using the class assignments of the training set (Fig. 1, set 1). The other method used the PCA eigenvectors of the training set (Fig. 1, set 2) to make combinations of the full set of features. In both, classes were assigned to all the records in the full data set with the quadratic classifier of Eq. 2. Table 2 shows that the two methods agree about equally well with the initial unsupervised classification and that agreement falls off if the training set is too small.

Selecting the small feature set (left side of Fig. 1) involves computing the squared Mahalanobis distance between each record and its cluster centre, in feature space. This variable has a chi-squared distribution with two degrees of freedom when only two features are being used. This provides two warnings if the classes in the training and full data sets do not coincide well. For example, 97.5% of records should lie within a squared distance less than

<table>
<thead>
<tr>
<th>Training set</th>
<th>with small feature sets</th>
<th>with combinations of the full feature set</th>
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</thead>
<tbody>
<tr>
<td>10%</td>
<td>83%</td>
<td>81%</td>
</tr>
<tr>
<td>25%</td>
<td>93%</td>
<td>93%</td>
</tr>
<tr>
<td>Swath</td>
<td>93%</td>
<td>91%</td>
</tr>
</tbody>
</table>

Table 2: Agreement between unsupervised MULTIVIEW classification of the full survey area and six classifications by discriminant analysis.

Fig. 4: Outliers in quadratic classification with a small feature set when trained with all six classes (middle) and with one class omitted (right). The class omitted from training was the rock class, class 1, and most of the outliers are atop rocks.
7.4. If classes coincide, about 2% of the records were found to be at longer distances. As shown in the middle panel of Fig. 4, most of them are located near class boundaries. The second warning is the mean and standard deviation, nominally $n$ and $\sqrt{2n}$, with $n$ degrees of freedom. With coinciding classes and $n = 2$, these were typically about 1.9. The purpose of finding outliers is to warn the user that one or more seabed types in the new area are missing from the training set, and that this need to be corrected. This was simulated by removing the rock class, class 1, from the 25% training set and re-classifying. This gave 8.3% outliers, mean 3.0, and standard deviation 5.2, clear indications of a mismatch. The right panel of Fig. 3 shows that most of these are over rock areas, where class 1, the missing class, used to be.

6. CONCLUSIONS

Classifying sonar images of the seabed first requires unsupervised or manual classification of a few images that encompass all seabed types in an area. These images form a training set. Two methods of using this training set to classify a wider area, both using quadratic classification, were evaluated in this paper.

The methods are very different in the number of features used. One uses only two or three features, selected for best discrimination within that training set. The other uses MULTIVIEW’s full set of 132 features, combined with weights from PCA. Assessed against an unsupervised classification of the full area, they perform almost equally well. An obvious advantage of the small feature set is calculation speed, which is fast enough for real-time classification. Perhaps as important is that with small features sets it is easy to understand which image characteristics are significant for classification. This last point addresses a common complaint about statistical acoustic classification, the difficulty of relating class maps to images. Methods that have been criticized in this way include principal components analysis, canonical variates analysis, and neural networks.

The fact that the squared Mahalanobis distance is chi-squared distributed, when the underlying data are multidimensional normal, provides a strong statistical basis for identifying outliers and thus indicating that a class may be missing from the training set.

This paper deals with agreement between class maps, while little can be said about accuracy. The true nature of the seabed is accessible only non-acoustically, from grab samples, for example. The various types of ground truth never have enough spatial resolution to rank maps like those in Fig. 2 for accuracy. Given this continuing limit on measuring the accuracy of acoustic classification, the aim of future research should be to develop quick convenient understandable methods that generate useful practical maps of seabed types. This paper describes steps in that direction.

REFERENCES

AN EFFICIENT METHOD FOR REDUCING THE SOUND SPEED INDUCED ERRORS IN MULTIBEAM ECHOSOUNDER BATHYMETRIC MEASUREMENTS

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Abstract: Nowadays extensive use is made of multibeam echosounders (MBES) for mapping the bathymetry of sea- and river-floors. The MBES is capable of covering large areas in limited time by emitting an acoustic pulse along a wide swathe perpendicular to the sailing direction. The angle and the corresponding two-way travel-time of the received signals are determined through beamsteering at reception. Water depths along the swathe can be derived from this angle and travel-time combination. In general, two sets of sound speed measurements are taken when conducting MBES measurements. The first set is used for the beamsteering and consists of the sound speeds at the MBES transducer. The second set is used for determining the propagation of the sound through the water column, needed for correctly converting the measured travel times to a depth. In general, this set of sound speed measurements consist of the complete sound speed profiles (SSPs). The quality of the sound speed measurements at the transducer position sometimes gets degraded, resulting in beam steering angles that differ from those aimed for. Also sometimes the SSPs used for converting the beam travel times to depths deviate from the true prevailing SSPs due to the, in general, limited amount of SSP measurements taken during a survey. Both above mentioned effects result in an erroneous bathymetry. Here, we present a method for eliminating these errors, without the need for additional sound speed information.

Keywords: Multibeam echosounder, sound speed profile, optimization
1. INTRODUCTION

Multibeam echosounder (MBES) systems are nowadays extensively used for mapping the bathymetry of sea- and river-floors. The MBES sends out an acoustic pulse along a wide swathe perpendicular to the sailing direction, thereby covering a large area of the seafloor at once. Beamsteering at reception allows for determining the (two-way) travel-time of the received signals as a function of angle. Water depths along the swathe can be derived from the combination of travel-time and angle, provided that the local sound speed profile in the water column is known [1]. However, in environments with large variations in the water column sound speeds (both temporally and spatially) this is not always the case, preventing a reliable conversion from the measured travel-times to bathymetry. Situations where information regarding the prevailing sound speeds is insufficient occur, for example, in estuaries where fresh river water mixes with seawater.

Also for the beamsteering, sound speeds need to be known accurately. Hereto, often a sound speed measurement device is placed close to the MBES transducer. However, due to e.g. algae growth on the sensor, the quality of these sound speed measurements can get degraded. As a result of this, the actual beamsteering angles deviate from the angles aimed for, and are not known.

Both effects, i.e., employing an erroneous sound speed profile for converting travel-times to water depths, and using an erroneous sound speed value for the beamsteering result in errors in the derived bathymetry. These errors can be such that surveys need to be repeated, which is very undesirable due to the high costs involved.

MBES surveys are normally carried out such that neighboring swathes partly overlap. This overlap allows for detecting the sound speed induced errors, since the water depths, as determined at the overlapping swathe points, will be different for each of the swathes. However, the overlap does not only allow for detection of the errors, it also allows for eliminating them. The method described here fully exploits the redundancy in the MBES measurements obtained from the overlap of adjacent swathes. Since temporal variations of the seafloor during the survey (several hours) are negligible, differences in bathymetry at overlapping parts of the swathes are the result of measurement errors. The method presented assumes errors due to erroneous sound speeds to be dominant. The prevailing water column sound speeds and thus the bathymetry are then estimated by inversion, requiring the derived bathymetry along overlapping regions to coincide, i.e., it searches for those water column sound speeds that result in a maximum agreement in the bathymetry along the overlapping swathes. The Gauss-Newton method is employed for the optimization. In [2] use is made of simulations to demonstrate the method’s applicability. The configuration considered in [2] represents a typical MBES survey geometry. The results demonstrate that the method allows for correctly estimating the true bathymetry and the true sound speeds. In [3] the method is applied to real MBES data and it is shown that it allows for efficiently eliminating sound speed induced errors in the bathymetry.

The redundancy in the measurements is due to the overlap in adjacent swathes. This redundancy increases with increasing overlap. However, increasing the overlap requires the adjacent tracks to be sailed closer together, decreasing the MBES survey efficiency. In this paper the effects of overlap on the method’s performance is assessed.

Section 2 provides a description of the approach. In Section 3 the results of applying the method for different overlaps are presented. The data considered are synthetic simulated data. The paper ends with the conclusions in Section 4.
2. DESCRIPTION OF THE APPROACH

Multibeam echosounders (MBES) emit acoustic pulses in an opening angle of 1 to 2 degrees in along-track direction, and in an opening angle of about 120–150 degrees in the across-track direction. Beamforming in across-track direction is applied to determine the corresponding two-way travel-times for a selected number of arrival angles. Water depths along the swathe spanned by the across-track opening angles are determined from the combinations of two-way travel-time and angle. Hereto, either propagation along straight sound rays is assumed for shallow water environments, or in case the curvature of the sound rays can not be neglected, ray-trace calculations are carried out.

Inaccurate knowledge about the water column sound speeds result in an erroneous bathymetry in two ways:

1. Errors in the beamsteering process. In case the actual sound deviates from the measured sound speed, the actual beamsteering angles differ from the beamsteering angles aimed for and are unknown.
2. Errors in the conversion from the angle and travel-time combinations to water depths along the swathe.

Fig. 1 shows the geometry of a typical MBES survey, consisting of a series of tracks sailed parallel to each other. Track distances are such that a certain overlap exists between adjacent swathes.

![Fig. 1: Schematic overview of MBES survey geometry. The arrows indicate the sailing direction. The rectangles indicate the area as measured per track.](image)

Fig. 2 shows an example of the typical bathymetric behavior along a cross-cut such as indicated in Fig. 1 in case erroneous sound speeds are used. The area in which these MBES were taken is located close to the entrance of the Rotterdam harbor, where mixing of fresh and salt water occurs. The number of parallel tracks amounted to 12. The bathymetry was estimated from the measured travel-times, employing all sound speed information available, i.e., sound speeds measured at the transducer head for the beamsteering and a single sound speed profile for calculating the sound propagation through the water column. The colors
indicate the bathymetry as estimated for each of the tracks. Differences in water depths along the overlapping parts of adjacent swaths range to almost 0.5 m.

Fig. 2: Example of ‘droopy’ effects. The MBES measurements were carried out near the entrance to the harbor of Rotterdam.

The method proposed for eliminating these effects fully exploits the redundancy of measurements in the overlap region between two adjacent swaths. Assuming that the seafloor does not change during a survey, the depth measurements in the overlap regions should be the same.

The measurements are the two-way travel-times. We aim to minimize the function:

$$E = \sum_{k=1}^{N} \sum_{j=1}^{S} (t_{k,j} - T_{k,j})^2$$

where $N$ and $S$ are the total numbers of MBES beams and swaths, respectively. The modeled two-way travel-times are denoted by $t_{k,j}$ and the measured two-way travel-times are denoted by $T_{k,j}$. The model that calculates $t_{k,j}$, accounts for both the effect of sound speed on the beamsteering and on the propagation through the water column. The unknowns are the sound speed profiles for each of the swaths and the bathymetry. These unknowns should be selected such that $E$ becomes minimal.

Hereto the seafloor is modeled with an interpolated grid function. The water depths at the grid positions are the unknowns that are obtained through the minimization of $E$. Assuming a shallow water situation, we approximate the sound speed profile by a constant sound speed. This results in one unknown sound speed for each of the tracks. The unknowns to be optimized are thus the depths at the grid positions and the sound speeds for each of the swaths.

For minimizing $E$ use is made of the Gauss-Newton method. The optimization procedure is as follows: For the seafloor, we use a fixed grid of horizontal positions, denoted $X_n$ in the across-track direction. At every position $X_n$, $Z_n$ denotes the corresponding water depth. Between the grid points the depth is interpolated linearly. For every beam $j$ at angle $\theta_{k,j}$, the point where the acoustic beam impinges on the model seafloor is denoted as $(x_{k,j}, z_{k,j})$. Fig. 3 shows a schematic overview of this model. The MBES is located at $(X_{k,MBES}, Z_{k,MBES})$. 
Mathematically, the function for $t_{k,j}$ can be derived by calculating the intersection between two lines: the sound ray and the line between the grid points in Fig. 3. The position at which these two lines intersect is given by:

$$x_{k,j} = \frac{X_{k,MBES} + Z_n - Z_{k,MBES} - X_n Z_{n+1} - Z_n}{\tan \theta_{k,j}}$$

and

$$z_{k,j} = \frac{x_{k,j} - X_{k,MBES}}{\tan \theta_{k,j}} + Z_{k,MBES}$$

Values for $t_{k,j}$ are calculated employing the water column sound speed $c_k$ as

$$t_{k,j} = \frac{2 \left(x_{k,j} - X_{k,MBES}\right)}{c_k \sin \theta_{k,j}}$$

where $c_k$ is the sound speed in swathe $k$.

We can write

$$y = A(x)$$

with $y$ the vector containing the measured travel-times, and $x$ the vector containing all unknowns.

Solving for $x$ in a least-squares sense requires the iterative Gauss-Newton approach. Hereto the expressions for the derivatives of $A$ to all unknowns have been determined.
3. RESULTS

The situation considered is that of 8 parallel tracks. The overlap between two adjacent swathes amounts to half the swathe. In practice, overlaps will in general be smaller. The grey area in the upper plot of Fig. 4 shows the bathymetry. True and “measured” sound speeds are given in Table 1. The bathymetry obtained when using the wrong “measured” sound speeds is also shown in the upper plot of Fig. 4. The resulting errors in the derived bathymetry are clearly visible.

![Figure 4. MBES measurement configuration consisting of 8 parallel tracks. The positions of the MBES are indicated by the vertical black dashed lines. The true bathymetry is indicated by the grey surface. The upper plot presents the estimates for the bathymetry along the swathes where use is made of erroneous sound speeds. The lower plot presents the optimized bathymetry.](image)

<table>
<thead>
<tr>
<th>Swathe</th>
<th>True sound speed (m/s)</th>
<th>Measured sound speed (m/s)</th>
<th>Optimized sound speed (m/s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1499.9</td>
<td>1505.6</td>
<td>1499.9</td>
</tr>
<tr>
<td>2</td>
<td>1504.3</td>
<td>1508.1</td>
<td>1504.3</td>
</tr>
<tr>
<td>3</td>
<td>1507.8</td>
<td>1496.7</td>
<td>1507.9</td>
</tr>
<tr>
<td>4</td>
<td>1504.0</td>
<td>1494.0</td>
<td>1504.1</td>
</tr>
<tr>
<td>5</td>
<td>1490.6</td>
<td>1504.9</td>
<td>1490.6</td>
</tr>
<tr>
<td>6</td>
<td>1500.0</td>
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<td>1499.8</td>
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<tr>
<td>7</td>
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<td>1502.2</td>
<td>1508.2</td>
</tr>
<tr>
<td>8</td>
<td>1502.4</td>
<td>1507.2</td>
<td>1502.4</td>
</tr>
</tbody>
</table>

Table 1: True, measured and optimized sound speeds for the 8 swathes.

The lower plot of Fig. 4 shows the optimized bathymetry. Clearly the optimized bathymetry almost coincides with the true bathymetry as indicated by the grey filled area. Furthermore, the optimized sound speeds almost coincide with the true sound speeds, as seen from Table 1. Fig. 5 shows the sound speed estimates obtained from 100 independent optimizations. From this plot it is clear that the method is capable of retrieving the true sound speeds. This off course also holds for the bathymetry.
In general, the overlap between adjacent swathes will be smaller. In addition, the measured travel-times will be subject to noise. To assess these effects, simulated data have been generated for measurement configurations with different overlaps and with noise added to the travel times. The left plot of Fig. 6 indicates the deviations of the optimized sound speeds from the true sound speeds as a function of the overlap. The right plot indicates the mean errors in the estimated bathymetry. The black lines indicate noiseless measurements. The red and cyan lines have been obtained by adding Gaussian noise to the measured travel times with standard deviations of 0.2 and 0.4 msec, respectively.

![Fig. 5. Sound speed estimates obtained from 100 independent optimizations. The situation considered is that of 8 parallel tracks. The black circles denote the sound speed estimates, plotted versus the true sound speeds. The grey circles indicate the measured sound speed values, also plotted versus the true sound speed.](image)

![Fig. 6. Mean error of the estimated sound speed as a function of the overlap (left plot) and the mean error in the estimated bathymetry (right plot). The black lines indicate noiseless measurements. The red and cyan lines have been obtained by adding Gaussian noise to the measured travel times with standard deviations of 0.2 and 0.4 msec.](image)

It can be concluded that both an increasing noise level and a decreasing overlap result in increased deviations of the estimations from the true bathymetry and sound speeds, but that still the deviation is limited. This is further illustrated in Fig. 7, showing the estimated bathymetry (lower plot) for the situation with noise superimposed on the measured travel times (standard deviation of 0.4 msec), and an overlap amounting to $1/8$th of the total swathe width only.
4. SUMMARY AND CONCLUSIONS

In this paper it is demonstrated that by employing the overlap between adjacent MBES swathes, errors in the bathymetry due to erroneous sound speed information, can be eliminated. In principle, this method allows for MBES surveys where no information regarding the prevailing sound speeds is acquired. The only requirement is that sufficient overlap exists between the neighboring swathes. Simulations such as presented in the current contribution allow for a quantification of the required overlap.

![Image 1](image1.png)

**Fig. 7.** Similar to Fig. 4. However, now noise is imposed on the travel time measurements and the overlap is limited.

5. ACKNOWLEDGEMENTS

We would like to thank Ben Dierikx and the Dutch Directorate-General for Public Works and Water Management for useful discussions. This work is financially supported by the Dutch Directorate-General for Public Works and Water Management, the Netherlands Geodetic Commission, and the Technical University of Delft.

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Fish Detection and accuracy of length measurement with DIDSON acoustic camera

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Abstract: Open water of lakes and rivers could be surveyed by DIDSON (dual-frequency identification sonar) beaming sideways from a moving survey boat. This approach could mitigate current difficulties with the interpretation of records of split-beam echosounders. Surface-oriented fish is then consequently recorded in all DIDSON beams. The detection is considered successful if fish is seen and its size can be read reliably. The standard acoustic experiment with a fish rotating carousel was conducted with DIDSON in the Rimov reservoir (Czech Republic). Cyprinid fish (bream Abramis brama, roach Rutilus rutilus and carp Cyprinus carpio) of known size and aspect angle were deployed in known positions of high-frequency DIDSON fan of beams. Fish were stunned and measured before being mounted in a rotatable frame of the carousel. Both fixed and simulated mobile records were obtained in two ranges (approx. 6 and 9.5 meters). Probability of recording equaled to 100% when observing the side aspect, but decreased in less reflective aspects. The decrease of the recording probability was more pronounced with small fish and with increasing range. Regular underestimation of observed fish sizes was found on the margins of beam fan and with oblique aspects over 45 degrees from side. These findings show that at the moment, the applicability of Didson in quantitative mobile surveys of smaller fish is questionable but not impossible. The model describing the detection probability and sizing accuracy as a function of Didson tilt, fish range, aspect and size is being developed.
SIMULTANEOUS ESTIMATION ACCURACY FOR ANGLES OF BEAM ARRIVAL FROM A SOURCE, BASED ON THE MODIFIED PRONI METHOD

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Abstract: Under water a signal from a source propagates along several beams. An accuracy of estimation for angles of beam incoming from a source depends on a signal-to-noise ratio, interbeam spacing and correlation factors of signals, propagating along different beams. Basing on the modified Proni method, an algorithm is suggested providing consistent, asymptotically normal and unbiased estimation of beam arrival angles. Error variances for beam arrival angle estimation are close to irreducible values determined from the Rao-Cramer inequality.

Keywords: Beam arrival angle, modified Proni method, error variance of angle estimation, Rao-Cramer inequality, correlated signals.

1. INTRODUCTION

A number of hydroacoustics applications require determination of arrival angles for a signal propagating along several beams from one source. This problem can be reduced to estimation of angular coordinates of several closely signal sources. There are many algorithms known, for example [1], which make it possible to solve this problem. However these algorithms are generally developed for determining angular coordinates of the sources of weakly correlated signals. A measurement accuracy of angular coordinates when using these algorithms gets drastically worse under increase of correlation between signals, and in the case of total signal correlation the estimates become inconsistent. Acoustic signals
propagating under water along different beams from one source are usually strongly correlated.

On the basis of the Proni approach an algorithm is developed for simultaneous estimation of angles of beam arrival from a source [2]. According to this algorithm, angular coordinates of signal sources are the roots of the family of polynomials, the degree of which is equal to the number of beams. As it is shown in the paper, the suggested algorithm gives consistent estimation of beam arrival angles, including that one under correlation of signals propagating along different beams. However, the estimate variance depends on the choice of a polynomial from the polynomial family, developed in [2]. The algorithm modification is suggested providing minimization of estimation errors variance. The modification consists in designing a polynomial being a linear combination of the polynomial family constructed in [2].

2. ALGORITHM FOR SIMULTANEOUS ESTIMATION OF BEAM ARRIVAL ANGLES

A source of narrow-band acoustic signal in water environment, parameters of which are isotropic in the horizontal, is considered in the paper. A signal from the source propagates along several beams in the vertical plane. The signals are received by the antenna array, in which a fan of directivity patterns (DP) is formed in the vertical plane. The number of propagating beams is considered to be known.

An signal under observation in the m-th DP at instant t can be represented by the following expression:

$$ u_{m,t} = \sum_{k=1}^{K} a_{k,t} g_m(x_{k,t}) + n_{m,t}; \ m = [1, M]; \ t = [1, T], $$

(1)

where $u_{m,t}$ – complex amplitude of signals being observed in the m–th DP at instant t; M, T – number of DP and time-split of observable signals, used to estimate beam arrival angles, respectively; $a_{k,t}$ – complex amplitude of a signal propagating along k–th beam at instant t; $x_{k,t} = \varphi(\varepsilon_k)$ – generalized coordinate of the k-th beam arrival angle in the vertical plane; $\varepsilon_k$ - the k-th beam arrival angle; $\varphi(\varepsilon)$ - some biunique function of beam arrival angle, introduced to simplify the estimation algorithms and determined by the geometry of the receiving antenna array; K – number of beams by which a signal from a source propagates; $g_m(x)$ – m-th DP, and $n_{m,t}$ – complex noise amplitudes in the m–th DP at instant t.

In expressions (1) and hereinafter: $[1, N]$ is index change from 1 to N.

B [2] basing on the Proni approach, an algorithm for simultaneous estimation of generalized coordinates of beam arrival angles. In accordance with this estimation algorithm, these coordinates are the roots of K-degree polynomial:

$$ p_0(x, B) = x^K + \sum_{k=0}^{K-1} d_k(B) \cdot x^k = x^K + \overline{x}^t \cdot \overline{d}(B) $$

(2)

In expression (2) and hereinafter: symbol "→" above a variable designates a vector; symbol "\(t\)" designates the operation of transposition.
Vector $\tilde{d}(B)$ of length $K$, composed of polynomial coefficients, is calculated by the formula:

$$
\tilde{d}(B) = -\left[ X_0 \cdot C(B) \cdot X_0^* \right]^{-1} \cdot \left[ X_0 \cdot C(B) \cdot x_0 \right]
$$

(3)

where $\|X_0\|_{k,m} = x_0^{k^{-1}}$; $k = [1,K]$; $m = [1,M]$ - matrix, size $K \times M$; $\|x_0\|_m = x_0^K$ - M-length vector; $x_0_m$ - generalized angular coordinate of direction maximum of the $m$-th DP; $C(B) = \left( \sum_{t=1}^{T} \left( V_t \cdot Y_0^t \cdot B \cdot Y_0 \cdot V_t^* \right) \right)$ - matrix, size $M \times M$; $\|Y_0\|_{r,m} = x_0^{r^{-1}}$; $r = [1,(R-K)]$; $m = [1,M]$ - matrix, size $(R-K) \times M$; $R$ – order of DP polynomial approximant (to be considered later); $\|V_t\|_{m,m} = \delta_{m,m} \cdot v_{m,t}$, $m = [1,M]$ - scalar matrix, size $M$; $\delta_{m,m}$ - Kronecker symbol; $B$ – arbitrary nondegenerate matrix, size $(R-K) \times (R-K)$.

In expression (3) and hereinafter: $\|X\|_{n,m}$ designates n,m entry of matrix X; symbol ‘*’ – Hermitian conjugation operation.

Scalar matrix $tV$ required to perform calculations by formula (3) is computed as follows. Every directional pattern is approximated by $R \geq M$ degree polynomial:

$$
g_m(x) = \sum_{r=0}^{R} f_{m,r} x^r.
$$

(4)

where $f_{m,r}$ - coefficients of a polynomial approximating the $m$-th DP.

Note that for some types of antenna arrays the directional pattern can be truly represented by a finite-degree polynomial. For example, DP of linear equispaced antenna array is represented by a polynomial, the degree of which is equal to the number of elements in the array minus 1.

Diagonal elements of matrix $V_t$ are components of vector $\tilde{v}_t$, which are calculated by the formula:

$$
\tilde{v}_t = G_{o^{-1}} \cdot \bar{u}_t.
$$

(5)

where $G_o = (F \cdot Z_0)$; $\|F\|_{m,r} = f_{m,r}$ - matrix, size $(R+1) \times (R+1)$, formed by coefficients of polynomials approximating DP; $\|Z_0\|_{r,m} = x_0^{-r}$; $r = [1,R+1]$; $m = [1,M]$ - matrix, size $(R+1) \times M$; $\|u_{m,t}\|_m = u_{m,t}$ - M-length vector formed by instantaneous values of signals under observation.

As it is seen from formulas (2) and (3), generalized coordinates of beam arrival angles are the roots of a polynomial family defined by nondegenerate matrix $B$.

Below we consider an accuracy of generalized angular coordinates (GAC) estimation by the given algorithm.
3. ACCURACY OF SIMULTANEOUS ESTIMATION OF GENERALIZED ANGULAR COORDINATES

Let us introduce a vector of GAC estimation errors: \( \Delta \hat{x}_r = \hat{x}_r - x_r \), where \( \hat{x}_r \) - GAC estimation vector; \( x_r \) - vector of true GAC. An accuracy of simultaneous GAC estimation will be characterized by a vector of mean values of GAC estimation error (characterizes an estimator bias): \( \Delta \hat{x}_r = M \{ \Delta \hat{x}_r \} \), and a correlation matrix of GAC estimation error (characterizes estimate efficiency): \( K_{xx} = M \{ \Delta \hat{x}_r \cdot \Delta \hat{x}_r^\top \} \). Hereinafter \( M \{ \} \) designates an expectation operator.

When considering GAC estimation errors we’ll assume that noises in different DP are random processes with zero expectation and variance \( \sigma_N^2 \); noises in different DP and in various time-split of signals are not correlated. These assumptions are not a matter of principle and are made to simplify the further consideration.

Having expanded polynomial (2) in the neighborhood of true GAC values and taking into account only the members of the 1st order of vanishing, after simple but lengthy calculations we can obtain an estimate of an error vector:

\[
\Delta \hat{x}_r (B) = -D_p \left[ \sum_{t=1}^{T} \left( A_t^* Y_r \cdot B \cdot Y_r \cdot A_t \right) \right]^{-1} \left[ \sum_{t=1}^{T} \left( A_t^* Y_r \cdot B \cdot Y_r \cdot A_t \right) \right] \tag{6}
\]

where \( D_p \) - scalar matrix formed of instantaneous beam amplitude values; \( Y_r \) - matrix formed of instantaneous noise amplitude values.

As it is seen from formula (6), the vector of GAC estimation errors is proportional to the sum of noise vectors. Taking into consideration this fact, as well as assumptions made about the noise properties, and the central limiting theorem, the following conclusion can be made: errors of GAC estimation are asymptotically distributed by the normal law with zero expectation, i.e. they are asymptotically normal and unbiased estimates.

Using formula (6) a correlation matrix of GAC estimation error can be calculated:

\[
K_{xx} (B) = \sigma_N^2 \cdot D_p \cdot C_0 (B)^{-1} \cdot D_0 (B) \cdot C_0 (B)^{-1} \cdot D_p \tag{7}
\]

Designations introduced in formula (7):

\[
C (B) = \text{Re} \left\{ \sum_{t=1}^{T} \left( A_t^* \cdot Y_r \cdot B \cdot Y_r^* \cdot A_t \right) \right\} ;
\]

\[
D (B) = \sum_{t=1}^{T} \left( A_t^* \cdot Y_r \cdot B \cdot Y_r \cdot A_t \right).
\]

In (7) and hereinafter \( \text{Re} \{ x \} \) designates the integral part of number \( x \).
As could be expected, the correlation matrix of GAC estimation error depends on matrix B used to evaluate vector $\mathbf{d}_B$. This results in a problem of determining the optimal matrix B, which will minimize a variance of GAC estimation error. To find an optimal matrix it is necessary to differentiate matrix $K_{xx}(B)$ with respect to matrix B, and set the result equal to zero matrix. After simple but lengthy calculations we can show that the optimal matrix is the following:

$$B_{opt} = \left( Y_0 \cdot P_0 \cdot G_0^2 \cdot P_0 \cdot Y_0 \right)^{-1}. \quad (8)$$

**Note.** Formula (8) includes scalar matrix $P_0$, the elements of which are the values of the desired polynomial (2) in supporting points. Since the polynomial is unknown a priori, matrix (8) cannot be formed. In order to solve this problem an interactive procedure may be applied.

At the first step matrix $B_0 = \left( Y_0 \cdot Y_0^* \right)^{-1}$ is calculated, and the polynomial coefficient vector is determined by formula (3). On the basis of this vector matrix $P_0$ is calculated. At the 2nd step matrix $B_{opt}$ is calculated by formula (8), and the polynomial coefficient vector is refined by formula (3). The model-based analysis shows that the proposed procedure ensures GAC estimation with the same accuracy as when applying matrix (8).

Taking account of (7) and (8), minimal correlation matrix of GAC estimation errors is equal to:

$$K_{xx, min} = \sigma_N^2 \cdot D_p \cdot C_0^{-1} \cdot D_p \quad (9)$$

where $\|C_0\|_{k,l} = \|\hat{R}_a\|_{k,l} \cdot \|D\|_{k,l}$, where $\hat{R}_a = \text{Re} \left\{ \sum_{t=1}^{T} a_t^* \cdot \hat{a}_t \right\}$ - the estimator for correlation matrix of amplitudes of signals propagating along different beams; $D = Y_r \cdot \left( Y_0 \cdot P_0 \cdot G_0^2 \cdot P_0 \cdot Y_0 \cdot Y_r \right)^{-1} \cdot Y_r$.

As it follows from formula (9), the error correlation matrix depends on the correlation matrix of signals incoming in different beams.

We’ll assume that the source radiates a narrow-band random signal with zero mean, and tic marks of the signal are not correlated. Consider two extreme cases of signal propagation.

In the first case signals propagating along different beams are not correlated. With sufficiently large sample size of the signal ($T \gg 1$) we may suppose that $\|\hat{R}_a\|_{k,l} \approx T \cdot \delta_{k,l} \cdot \sigma_k^2$, where $\sigma_k^2$ - power of a signal propagating along the k-th beam. The correlation matrix of GAC estimation errors becomes the scalar matrix. At that, errors estimation of GAC are not correlated and have the following variance:

$$\text{Var} \{ \Delta x_r_k \} = \frac{\sigma_N^2}{T \cdot \sigma_k^2 \cdot \|D\|_{k,k}} = \frac{1}{\text{SNR}_k \cdot \|D\|_{k,k}} \quad (10)$$

where $\text{SNR}_k = \frac{T \cdot \sigma_k^2}{\sigma_N^2}$ - signal-to-noise ratio for the k-th beam.
In (10) \( \text{Var}\{\cdot\} \) designates a variance operator.

In the second case signals propagating along different beams are completely correlated. With sufficiently large sample size of the signal \((T \geq 1)\) we may suppose that

\[
\text{Ra}_{k,l} = T \cdot \text{Re}\{a_k \cdot a_l^*\},
\]

where \(a_k\) - complex amplitude of a signal propagating along the \(k\)-th beam. The correlation matrix of GAC estimation errors is not scalar, and therefore errors estimation of GAC are correlated.

The paper size does not allow other properties of simultaneous estimation of beam arrival angles to be considered in details. Below the author presents the summarized analytical results obtained for the properties of simultaneous estimation of beam arrival angles on the basis of the modified Proni algorithm:

- the estimates are consistent, asymptotically normal and unbiased;
- error variances are close to minimum possible values determined from the Rao-Cramer inequality;
- if the signals propagating along different beams are not correlated, then: 1) estimate errors are uncorrelated; 2) error variances are inversely related to the squared interbeam spacing; 3) error variances are close to the error variance at single-beam propagation with the interbeam spacing more than the beamwidth;
- if the signals propagating along different beams are coherent, and phase difference between signals is divisible by \((2n+1)\pi/2\), the estimate has the same properties as in the case of uncorrelated signals;
- if the signals propagating along different beams are coherent and phase difference between signals is divisible by \(\pi\), then: 1) estimate errors are correlated; 2) error variances are inversely related to the biquadrate of the interbeam spacing; 3) error variances are close to the error variance at single-beam propagation with the interbeam spacing more than one beamwidth and a half.

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Structured Session 30

Polar Acoustics- Field and Remote Measurements

Organizers: Jaroslaw Tegowski & Alexander Gavrilov
Methods and results of in situ target strength measurements of fish in the Barents Sea during combined trawl-acoustic surveys

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Abstract: This paper presents methods for collecting acoustic and biological data, including in situ target strength (TS) estimates of fish, with results for Atlantic cod (Gadus morhua), haddock (Melanogrammus aeglefinus), polar cod (Boreogadus saida) and capelin (Mallotus villosus socialis) obtained from combined trawl-acoustic surveys. These include fish in small, average and maximum length classes. The investigations were carried out using SIMRAD EK500/EK60 (38 kHz) scientific echosounders with split-beam transducers and special post-processing software. These methods allow to receive enough great volume of in situ data TS estimates of fish to determine as varies TS during different seasons of year at various times day and on different depths. Based on the analysis of data collected in the Barents Sea over the years 2000-2008, was obtained at 38 kHz a relationship

\[ TS = 25.4 \log_{10}(LT) - 75.1 \] for Atlantic cod within total length \( LT = 5 - 136 \) cm; a relationship

\[ TS = 23.9 \log_{10}(LT) - 71.9 \] for haddock within \( LT = 9 - 77 \) cm; a relationship

\[ TS = 19.3 \log_{10}(LT) - 69.0 \] for polar cod within \( LT = 2.8 - 23.5 \) cm and a relationship

\[ TS = 20.6 \log_{10}(LT) - 74.7 \] for capelin within \( LT = 4 - 19 \) cm, with TS in dB and LT in centimeters.

Key words: length classes, target strength, trawl-acoustic surveys.
1. Introduction

Working equation to estimate density of fish concentrations \( \rho_y \) by hydroacoustic method (HA- method) is as follows [1]:

\[
\rho_y = P_y s_A \sum_{i=1}^{n} \sum_{j=1}^{m} P_j \sigma_{ij},
\]

(1)

where \( i \) – fish species index; \( j \) – fish size group index; \( P_j \) – portion of fish from j-size group of \( i \)-species in concentration; \( s_A \) - nautical area scattering coefficient (in m\(^2\)/mile\(^2\)); \( \sigma \) - backscattering cross-section for one fish (in m\(^2\)), which is connected with fish target strength \( TS \) (in dB) through the relationship \( TS = 10 \log(\sigma/4\pi) \); the dependence of target strength \( TS \) on fish length \( L_T \) (\( L_T \) - total fish length) is expressed as \( TS_i = B_i \log(L_T) + A_i \), where \( B_i \) and \( A_i \) - coefficients depending on fish species; \( s_A \) is measured by echosounder, parameters \( \sigma_{ij} \) and \( P_y \) are determined in the experiments and they are key in estimation of density of fish concentrations \( \rho_y \).

At present, in practice of using HA-method in the investigations of fish stocks, there are the two main problems: the first one is the determination of fish target strength in natural concentrations \( in situ \) during the acoustic surveys and the second one is correct estimation of fish size composition in concentrations. The paper considers the results of researches associated with solving the first problem, determination of \( TS \) and \( TS - L_T \) dependencies for fish \( in situ \).

Investigations of fish target strength have been conducted since 1962 as \( ex situ \) placing fish in special cages, as in natural concentrations of freely swimming fish \( in situ \) [2]. It was concluded that only direct measurements of \( TS \)\( in situ \) may provide with obtaining more correct and statistically grounded data [3].

Now a days, Russian and Norwegian specialists apply the following \( TS - L_T \) dependencies to estimate stocks of the Atlantic cod (\textit{Gadus morhua}), haddock (\textit{Melanogrammus aeglefinus}), Polar cod (\textit{Boreogadus saida}), capelin (\textit{Mallotus villosus socialis}), redfish (\textit{Sebastes marinus}) and blue whiting (\textit{Micromesistius poutassou}) in the Barents and Norwegian Seas [4,5]:

\[
\begin{align*}
TS(\text{cod, haddock, redfish}) &= 20 \log_{10}(L_T) - 68 \quad (2) \\
TS(\text{polarcod, bluewhiting}) &= 21.8 \log_{10}(L_T) - 72.7 \quad (3) \\
TS(\text{capelin}) &= 19.1 \log_{10}(L_T) - 74.0 \quad (4)
\end{align*}
\]

The relationships (2), (3) and (4) were derived 25-30 years ago and they need testing and correction. Appearance of new computer technologies of logging, storage and post-processing of \( TS \) data eases solving the task of estimating \( TS - L_T \) dependencies for fish \( in situ \).

Research on estimating fish \( TS \)\( in situ \) has been carried out by the Polar Research Institute of Marine Fisheries and Oceanography (PINRO) since 1998 [6,7,8,9,10]. These investigations were done using research vessels equipped a Simrad EK500/EK60-38 kHz
scientific echosounders with a split-beam transducers. The LTSD100 software developed at PINRO [9], or the FAMAS software developed at the Pacific Research Institute of Marine Fisheries and Oceanography (TINRO) [11] or the Simrad BI60 [12] post-processing software was used for post-processing the raw echosounder data during TS analysis. This paper presents the results of in situ TS measurements on Atlantic cod, haddock, polar cod and capelin done in the Barents Sea from 1998 to 2008.

2. Material and methods

The methodology of definition of TS of fishes in situ is presented on fig.1 and consists of three stages [7,8].

Stage 1: During the surveys, data on fish TS are collected over the entire depth range, using scientific echosounders with split-beam transducers and appropriate data-collection and processing software. Trawls are made while simultaneously measuring the TS distribution within the actual fishing zone. The length composition of the ensonified fish is determined from the trawl catches, using codends fitted with small-mesh liners to reduce their selectivity.

Stage 2: Acoustic and biological data collected are processed. Two methods have been used for this purpose; both have advantages and disadvantages, but they can complement each other to give better TS estimates for each length class. These methods are described below.

First, the TS-distribution, and the mean \( \overline{TS} \) are estimated within the fishing zone, using the post-processing software LTSD100 [9] or FAMAS [11]. TS-distribution from fishes of the given kind is allocated from multimodal TS-distribution from all objects. It is reached by a choice minimal (\( TS_{\text{minthr}} \)) and maximal (\( TS_{\text{maxthr}} \)) thresholds of TS registration, corresponding TS of fishes of the minimal and maximal length classes in catch. The mean length \( \overline{T_L} \) of the fish in the catch is estimated and the TS in the fishing zone is associated with the \( \overline{T_L} \) of the catch (fig.2). This method allows estimation of TS for fish in the small and medium length classes.

Second method assumes that the maximum observed TS is associated with the largest fish caught in proximity to the relevant echo tracks. Using the post-processing software Bi60 [12], a number of single-fish tracks with near-maximum TS are selected from the echogram of the active fishing zone. The mean greatest \( \overline{TS}_m \), from within an echo track, the standard deviation, \( SD_{TS_m} \), for each track and the mean standard deviation, \( \overline{SD}_{TS_m} \) over all the selected tracks are estimated (fig. 3). From the chosen tracks, values of \( \overline{TS}_m \) within \( SD_{TS_m} \) of the greatest \( \overline{TS}_m \) are selected. The \( \overline{TS}_m \) is the mean of these values. The corresponding fish in the catch are those from the maximum length, \( L_{T_m} \), down to a lower limit, as determined by \( SD_{TS_m} \). The mean length of these fish is denoted \( \overline{L_{T_m}} \). An analysis of echo tracks from cod, with TS ranging from 50 to 136 cm, at 50–500-m depths revealed that the TS variations within an echo track might range from 2 to 20 dB, whereas \( SD_{TS_m} \) varied from 0.7 to 3.0 dB, with a mean \( \overline{SD}_{TS_m} \) of ~ 2 dB. By inverting Equation (2) and considering length classes per 5 cm intervals, we found that the value \( \overline{SD}_{TS_m} = 2 \) dB corresponded to the following groups: 4–5 length classes (from 40 to 65 cm) for fish with \( L_T = 50 \) cm; 6–8 length classes (from 60 to 100 cm) for fish with \( L_T = 80 \) cm; and 8–9 length classes (from 80 to 125 cm) for fish with \( L_T = 100 \) cm.
If there is only one track in the fishing zone within $SD_{Tm}$, $TS_m$ is compared with $LT_m$. If there are several tracks meeting this criterion, $<\overline{TS}_m>$ is compared with $\overline{LT}_m$. This method allows us to estimate $TS$ of the maximal length classes of fish.

In the single-target measurements by the second method, there is significant uncertainty when comparing $\overline{TS}_m$ and $LT_m$ because of the difficulty in correctly pairing the data. However, when the number of paired data is large enough, their averaged statistics allow an acceptable determination of particular $TS - LT$ relationships. It must be emphasized that the second method has the advantage that it is more applicable when fishing commercially, that is, without a small-mesh liner in the codend. It can also be used in mixed-species conditions, when the catch is dominated by one size group, but there are a few larger targets. For instance, when haddock is more abundant than cod, individual cod are larger and therefore their tracks can be easily identified and analysed using the $\overline{TS}_m$ data.

**Methodology of definition of fishes target strength**

*Fig. 1: Methodology of definition of fishes target strength in situ.*
Stage 3: The TS - $L_T$ relationships are estimated for all length classes using the acoustic and biological data obtained by both above-mentioned methods, namely $\overline{TS}$ and $\overline{LT}$, $\overline{TS}_m$ and $\overline{LT}_m$, $<\overline{TS}_m>$ and $\overline{LT}_m$, in accordance with the methods described by Ricker [13] and Glantz [14]. The geometrical-mean TS-length linear regressions (GM regression) and the corresponding 95% confidence limits (95% c.l.) are estimated. Doubtful data outside the 95% c.l. are then removed. The variance $\hat{S}_y^2$ and 95% c.l. for each predictive value, $\hat{Y}$, in the formula $Y = A + BX$ (where $Y = TS$, and $X = \log L_T$), are calculated as follows [13]:

Fig. 2: Example of FAMAS post-processing echogram and TS-distribution on capelin in the fishing zone on the feeding aggregations by SIMRAD EK60 echo sounder (in trawl catch $L_T = 8.8$ cm).

Fig. 3: Examples of SIMRAD BI60 post-processing of single-cod echo trace in the fishing zone with maximal target strength values (in trawl catch $L_T = 126$ cm, $\overline{TS}_m = 19.9$ dB, $SD_{TS_m} = 1.8$ dB).
\[
S_{\hat{Y}} = \frac{\sum (Y - \bar{Y})^2 \cdot (1 - r^2)}{n-1} + B^2(1-r)^2(X - \bar{X})^2
\]  
(5)

\[
95\% \text{ c.l.} = \hat{Y} \pm t_{\text{St}(p=0.05)} S_{\hat{Y}}
\]  
(6)

where \( r \) is the correlation coefficient, \( t_{\text{St}(p=0.05)} \) is the Student’s \( t \)-test coefficient defining the confidence for \( P = 0.95 \) and \( n \) is the number of TS and \( L_T \) paired samples.

3. Results and conclusions

There were more than 1000 hauls made by bottom and pelagic trawls with simultaneous registration of TS - distribution in the fished zone. The following data were obtained.

**Cod.** Relationships TS - \( L_T \) were determined at the depths of 50-100 m, 100-200 m, 200-300 m, 300-400 m and 400-500 m for the following periods: February, May-June, August-September and October-December. Seasonal and daily variations of cod target strength were estimated comparing TS - \( L_T \) dependencies. Cod target strength is maximal at 50-100 m depth, in the day time, in August, and it is minimal at 300-500 m depths, in the nighttime in August and in the day and night time in October-December. The maximal range of target strength variations during a year is by the order of 3.0 dB. In the day time cod target strength is greater by 0.4-1.5 dB, than in the night time depending on season and depth. Based on the processing of data we have derived a key TS - \( L_T \) relationship for cod for the most of seasons and time periods (Fig. 2):

\[
\text{TS(Cod, } L_T = 5\text{-}136 \text{ cm}) = 25.4 \log (\text{TL}) - 75.1; r^2 = 0.95; \text{SE} = 1.2; n = 498
\]  
(7)

**Haddock, polar cod and capelin.** TS was determined at the depths of from 20 to 250 m for August-December (haddock and polar cod) and for all seasons (capelin). Considerable variations of target strength depending on day, night and depth were not observed. The following TS - \( L_T \) relationships for haddock, polar cod and capelin were derived (Fig. 4):

\[
\text{TS(Haddock, } L_T = 9\text{-}77 \text{ cm}) = 23.9 \log (\text{TL}) - 71.9; r^2 = 0.93; \text{SE} = 1.1; n = 76
\]  
(8)

\[
\text{TS(Polar cod, } L_T = 3\text{-}23 \text{ cm}) = 19.3 \log (\text{TL}) - 69.0; r^2 = 0.95; \text{SE} = 0.9; n = 88
\]  
(9)

\[
\text{TS(Capelin, } L_T = 4\text{-}19 \text{ cm}) = 20.6 \log (\text{TL}) - 74.7; r^2 = 0.94; \text{SE} = 0.8; n = 88
\]  
(10)

In our opinion, applying the relationships (7), (8), (9) and (10) will lead to increase in the accuracy of estimates of these fish species by hydroacoustic method.
Fig. 4: The $TS - L_T$ relationships for Atlantic cod, haddock, polar cod and capelin.

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5. Reference


ACOUSTIC STUDIES OF DIVING BIRDS IN THE ARCTIC

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Abstract: The paper presents the results of acoustical investigations of an Arctic seabird, the little auk (Alle alle), conducted as a part of the international ALBERT project (Arctic Life: Bridging Ecosystem function using Remote Technologies). The main objective of this project is to estimate the interactions between the physical environment of the Spitsbergen waters, zooplankton communities (Calanus) and zooplankton-eating birds (Alle alle), as governed by climate changes. The warming Arctic climate influences the marine trophic chain in this particular part of the world. While the Arctic species of zooplankton are replaced by the Atlantic ones, the condition of the little auk is a marker of a changing climate. Aiming to support ornithological observations, the acoustic part of the ALBERT project focuses on the investigation of the diving strategy and behaviour of little auks. This paper reports an 11-day series of measurements conducted by the Autonomous Hydroacoustic System (AHS) deployed in Isfjorden (Spitsbergen) close to the bird colony. The acoustic system is a bottom-anchored buoy consisted of the active section – an up-looking echo sounder working at the frequency of 130 kHz – and the passive (noise) section – two external omnidirectional hydrophones. The echo sounder can detect the entrainment depth of gas bubbles produced by birds diving for their prey. The hydrophones record the ambient sea noise in a broad frequency band from a few hundred Hz to 35 kHz. The AHS dataset, both the echograms and the noise records, were analysed with respect to detect diving birds. The echosounder records were used to determine the depth distribution of dives and the dive traces, whereas the spectral characteristics of the recorded noise signals were used to discern the bird noise from the background of the ambient noise and to eliminate the noise components produced by the ships, rain and wind.

Keywords: Autonomous Hydroacoustic System, ambient noise, climate change, Arctic, diving seabirds.
INTRODUCTION

The warming Arctic climate influences the trophic chain in this particular part of the world. The intense inflow of the Atlantic water to the Arctic (Walczowski and Piechura, 2006) causes changes in the zooplankton community and as a consequence, in their predator living conditions. An Arctic bird, little auk (Alle alle), is considered a keystone species in the subpolar ecosystems, with a total population estimated at 30 million pairs (Stempniewicz et al., 2007). Its role in carrying nutrients from sea to land is very important. It feeds on zooplankton that are caught during wing-propelled dives. While the Arctic species of zooplankton are replaced by the Atlantic ones due to changes in oceanographic processes, the condition of the little auk is a marker of a changing climate. The measures of chick growth and reproduction may be sensitive indicators of drastic changes in food availability. Little auks breeding in Spitsbergen feed mainly on the large species of zooplankton – copepods. The adults of Arctic Calanus glacialis and Calanus hyperboreus are considerably larger and contain, respectively, approximately 10 and 25 times more energy (lipids), than the Atlantic Calanus finmarchicus. This high lipid accumulated by calanoid copepods is an essential energy source exploited by large stocks of fish, sea birds and marine mammals, enabling them to sustain overwintering populations. Climate change could alter oceanic prey distribution, thereby forcing seabirds to fly longer distances to reach feeding areas where their preferred prey is most abundant causing increased energetic demands (Bech et al., 2007). Bimodal foraging flight patterns are found – long trips are likely to have evolved in order to replenish energy reserves for the adults, whereas the short trips are mainly for providing food for chicks (Steen et al., 2007).

This research has been conducted as a part of the international ALBERT project (Arctic Life: Bridging Ecosystem function using Remote Technologies). It has been invented as a synoptic campaign conducted in a chosen area and at the same time (summer 2007) by many international teams, experienced in polar investigation.

The main objective of the project is to estimate the impact of climate warming on Arctic zooplankton communities (Calanus) and Arctic birds, little auks (Alle alle), and their physical environment. The Polish acoustic part of the ALBERT project focuses on the investigation of the diving strategy of little auks. This paper reports an 11-day series of measurements conducted by the Autonomous Hydroacoustic System (AHS) deployed in Isfjorden (Spitsbergen) close to the bird colony. The acoustic system is a buoy consisted of the active section – an up-looking echosounder working at the frequency of 130 kHz – and the passive (noise) section – two external omnidirectional hydrophones. The system is originally oriented towards registration of wind and rain ambient sea noise components, so the bandwidth of the tract is set to the frequency range from 350 Hz to 35 kHz.

Both the echosounder and the hydrophones detect the gas bubbles generated by birds diving for their prey. AHS dataset is presented and analysed and the preliminary results are discussed.

EXPERIMENT

The experiment took place in Isfjorden in the vicinity of Longyearbyen, capital of Svalbard. This particular place (78°14′23″N 15°17′43″E) has been chosen, first of all, because of the neighbourhood of a large colony of breeding little auks, and also for logistic reasons, because of the easy access from the operational centre at the University of Svalbard. AHS was anchored at the depth of 60 m, about 1 mile from the shore. The operation started at
13:00 on 17 July and ended at 10:50 on 28 July 2007. The buoy was released and recovered on 8 August.

The Autonomous Hydroacoustic System (AHS) comprises the upward looking echosounder, two omnidirectional hydrophones and steering and recording electronics (Szczucka et al., 2002; Szczucka and Klusek, 2006). The whole system consists of a pressure container, a pressure sensor, a computer, hydrophones attached to the container and an upwards-pointing hydroacoustic transducer mounted on top of the container, with an acoustic release and anchor attached to the bottom of the container. The echosounder works at 130 kHz. Its receiver circuitry consists of an amplifier and an envelope detector. The time varied gain (TVG) of the system can be implemented as any function in a dynamic range of 80 dB. The preamplifier is built in the hydrophone output. It is equipped with high-pass filters to avoid saturation from low-frequency ship sound and to reduce the influence of the surface wave motion and cable vibrations. Also aliasing from high frequency noise portion is reduced due to low-pass filters. The whole system has a positive buoyancy due to 15 spherical floats attached symmetrically to the container.

The computer controls the external pressure, echosounder triggering, power supply relays, and data logging rates.

DATA ACQUISITION AND PROCESSING

The idea of experiment was to compare the active and passive acoustic records of diving birds. There was no technical possibility to conduct both observations and registrations at the same time, therefore, the successive series of echosounding and noise measurements were carried out by turns. The data were acquired in three modes: echosounding, two-hydrophones and one-hydrophone operation. The Autonomous Hydroacoustic System worked in repeated 10-minute cycles. Each full cycle included: (i) 2-minute period of echosounding; (ii) 6 minutes of the two-hydrophones registration; (iii) 2-minute period of one-hydrophone registration. In the echosounding mode, 128 pings were transmitted with the pulse duration of 0.3 ms and the pulse repetition time of 500 ms. The echo signal envelope was sampled by a 12-bit analog digital converter. The sampling rate of 11 kHz ensured a depth resolution of 0.068 m. The voltage data acquired were converted into the volume backscattering strength $S_v$ by concerning geometrical beam spreading and absorption in sea water as well as technical parameters and calibration constant giving the $S_v$ matrix (Szczucka et al., 2002):

$$S_v(z, t) = S_v(z_j, t_k)$$

where:

$$z_j = j \Delta z, \quad \Delta z = \frac{c}{2 f_{\text{sample}}} \cong 0.068 m$$

$$t_k = k \Delta t, \quad \Delta t = 1 \text{ samples}$$

This provided for the formation of vertical profiles of $S_v$ and, in consequence, for the creation of echograms in any timescale, from minutes to days.

The noise signals were sampled by a two-channel 16-bit ADC with the frequency of 32 kHz in the case of two hydrophones, or 85 kHz when only one hydrophone was used (Lisimenka, 2007). Afterwards, they were post processed using MATLAB procedures. At the first stage, the FFT was applied to sub-samples of signals, each of them consisting of 16384 (16x1024) points. The noise spectrum and spectral characteristics were obtained. They were
averaged in 512 frequency bands (over 32 samples). The characteristic frequencies, signal intensity, spectrum slope and central frequencies were determined.

The Noise Spectrum Level was received by the digital filters with central frequencies logarithmically spaced in the frequency range from 0.4 – 31.5 kHz, ensuring nonoverlapping coverage in the whole registered frequency band.

PRELIMINARY RESULTS AND DISCUSSION

Data processing started with the visual inspection of the echograms. The bird echoes have been selected from 1572 echograms and the adjacent noise records have been searched for the diving birds’ noise. Many cases of birds’ presence in the water were detected. Fig.1 is an echogram constructed of 8 blocks separated from each other by 10 minutes, each consisting of 128 pings. The red irregular trace in the upper part is the surface echo, its ruggedness depends on the instantaneous wind speed. The entire record lasts over 1 hour and several traces of upwardly moving objects can be seen. They are (probably) the echoes from gas bubbles conveyed by the diving birds to a given depth, where they are released. Usually their return to the surface lasts several minutes. It is interesting that while the birds move downwards, with folded wings, in the most streamlined position, they are invisible. Just after they reach the desired depth, the entrapped gas bubbles are released and start their way back to the sea surface in the usual chaotic way, producing strange sounds and being a distinct target for the echosounder.

Fig.2 illustrates the example of such process, non-accomplished one – bubbles do not reach the sea surface in the time of 80 seconds spanning 128 echosounder transmissions. The estimation of the upward velocity of the released bubbles is possible – from the slope of the line approximating the shape of bubble cloud trace (black line in the picture). This velocity is 8.2 cm s⁻¹ what agrees quite well with the theoretically calculated speed of bubbles ascending in water. For clean bubbles with radii about 300 µm this speed is 7.5 cm s⁻¹ (Thorpe, 1982). The total number of such events detected by AHS during 11 days of our experiment and qualified as bird traces was 51, the depth reached by the diving birds varied between 2.6 m and 34.5 m with a mean value of 9.6 m. The histogram of dive depth obtained in this way is presented in Fig.3. Over 50% of the recorded events were limited to 5 m dive depth, but deeper dives were also detected. There were some periods of observation when diving were quite frequent, several of them during 1 hour, but there were also very long periods, lasting even more than 20 hours, with no diving detected. The recorded diving images were very often fragmentary, incomplete, because of the limited time of individual registration (80 s),
the beginning and/or final stages of diving were cut. Another problem complicating a full observation of diving birds was the very narrow, $7^\circ$ wide beam of the echosounder, what resulted in the small area ensonified - with diameter of 4.3 m at the depth 25 m and 7.3 m just below the surface.

![Echogram with the trace of ascending gas bubbles approximated by linear regression.](image1)

**Fig.2.** Echogram with the trace of ascending gas bubbles approximated by linear regression.

![Histogram of the dive depth recorded by the echosounder.](image2)

**Fig.3.** Histogram of the dive depth recorded by the echosounder.

The next step in data processing was the noise examination. The disadvantage of the chosen location was a heavy ship traffic, which contaminated the natural noise background during the experiment. It had to be taken into account in our analysis. **Fig. 4** shows the

![Noise level in four exemplary 1/3 octave bands.](image3)

**Fig.4.** Noise level in four exemplary 1/3 octave bands. Central frequencies in kHz are shown in legend. 17 – 28 July (the whole series).
ambient noise levels filtered in the 1/3-octave frequency bands as a function of time. The whole 11-day experimental series is presented. For the clarity of the picture, only 4 exemplary bands with central frequencies 0.63, 1.6, 4 and 10 kHz have been chosen from the entire analysed frequency range [0.4 – 12.5 kHz]. Four curves represent the calculated noise level with central frequencies marked in the legend. There is about 6 dB difference in noise level between adjacent curves, which is connected with frequency-dependent sound absorption in the sea water. On the background of undisturbed noise there are plenty of sharp peaks in this diagram, caused mostly by the passing ships, but some of them result from bird presence in water and our task was to distinguish birds from other sources of sea noise.

Special algorithms have been applied to recognise the realisations with shipping noises dominating in the entire frequency band or in its part only, generally at lower frequencies. The methods of the noise classification were based on the spectrum slope and rates of the spectral levels in selected bands of frequencies. Spectral properties of the ambient sound enabled the classification of the noise sources into categories. The obtained noise spectra were classified into six groups: birds, silence, ships, wind, rain and artefacts. The samples recognised as dominated by the undesired sources were not included in the analysis leading to the detection of diving birds.

Some examples of the Noise Spectrum Level calculated in 16 frequency bands are presented in Fig.5. There are different events visible: a bird, a silence and a ship. The bird peak is narrow and intensive around several kHz, the ship peak is broad and high at low frequencies. Unfortunately, we do not have the simultaneous records of echo and noise because they were recorded successively not concurrently. We can only expect that the birds’ dive observed in the active mode will have its reflection in the following passive cycle of measurement. But it is very seldom the case.

![Fig.5. Noise level in sixteen 1/3 octave bands for 3 different situations: (a) diving bird; (b) silence; (c) ship. Central frequencies in kHz are shown in legend.](image-url)

There are characteristic differences in the spectrum shape depending on the type of noise. Fig. 6 illustrates the noise spectrum of the silent situation, the diving bird, the ship and the chirp sonar with the central frequency 7 kHz, recorded on 24 July, working in all probability on some ship (probably a distant one). The bird causes the general increase of the noise level and numerous peaks in the range of several kHz (resonance frequencies of released gas bubbles), the chirp sonar spectrum is characterised by the much higher total level and the multifingered maximum in the range 5 – 9 kHz. Ships produce high-level noises in many frequency bands, specially the lower ones. Rain drops generate rather high-frequency noise with a local maximum at 14 kHz – it changes the spectrum slope in that frequency range. The spectrum for the high frequency component of the undisturbed sea noise is characterised by the slope of -5 dB per octave.

In order to discern birds against the noise background, the analyses of the statistical moments as well as of the deviation from the long term running mean have been performed, but there is still a necessity for constructing some better and more efficient criterions for the automatic detection of the bird events in the noise file.

The presented preliminary analysis of the data collected in the year 2007 shows that the experimental setup has numerous drawbacks. Firstly, the gap between the echosounder mode and the hydrophone ones is too big for the records to be compared. The diving event observed on the echogram is generally too short to be detected on the succeeding noise record, so it is hard to state how many times the hydrophones detected a bird versus the active acoustics. For the next summer season in the Arctic it is decided to shorten the sequences and repeat the whole cycle more frequently. Secondly, the echosounder beam is too narrow – the existing transducer should be replaced by other one with broader beam, spanning much larger volume of water. Additionally, lack of the direct observations by bird watchers makes the correlation of the diving events with acoustical images very difficult. May be there is the possibility of anything besides a diving bird making such an echo trace like observed in our echograms.

Some fundamental questions have still to be answered. What can we expect of birds as a consequence of water mass changes and different Calanus proportion in water? Will they change the range of flights for food? Will they change the entire flight strategy? Will they change the frequency of diving, depth of diving? Some changes have been already found – long and short trips to feed adults and chicks, respectively – but this is not the end of the story.
CONCLUSION

This paper has presented the preliminary results of noise and echosounder research on the diving Arctic birds. The main conclusions are the following:

- diving birds are detected both by the echosounder and by the hydrophones: bird event can be visible on the echogram in a form of a bubble trace ascending to the surface and in the noise signal as a narrow intensive peak with a maximum at 1-4 kHz;
- the birds dive to the depth of up to 32 m;
- various noise sources (birds, ships, interfering chirp sonar, rain) are categorised on the basis of the spectral characteristics of the noise signal;
- birds can be discerned basing on the differences in spectrum shape.

Direct ornithological observations and verification of the acoustic measurements are needed. Further research is planned in scope of the Polish-Norwegian collaboration in project ALKEKONGE.

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REFERENCES

UNDERWATER MONITORING OF POLAR WEATHER: ARCTIC FIELD MEASUREMENTS AND TANK EXPERIMENTS

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Abstract: Measuring ocean weather patterns is important for weather forecasting and also for quantifying the local effects of climate evolution. Underwater acoustics is a tool of choice to access space and time scales not available by other means (e.g. ships or satellites), in particular in polar regions. In summer 2007, field measurements of ambient noise were carried out in an Arctic fjord, as part of a survey with the Polish Academy of Science. Ambient noise was measured underwater at several locations along Kongsfjord (Svalbard), at frequencies from 100 Hz to 48 kHz. Our first investigations confirmed the frequency separation of specific weather processes (wind, rain and ice-related) found by other workers and showed this applied to the Arctic environments as well. By using Principal-Component Analyses, we show that weather patterns can be distinguished more rigorously on the basis of 3 distinct frequency bands, centred on 5 kHz (wind), 15 kHz (rain) and 45 kHz (ice). Subtle variations in wind speed and rain types were detectable using these bands. Perturbations from shipping and marine mammals could also be identified. To complement these interpretations, tank experiments were carried out at the University of Bath. The first set, conducted in a small-size tank, involved rain of varying intensity and drop size, small wind and melting ice. These were successfully separated using frequency bands of 5 and 15 kHz, but sampling rate prevented access to frequencies larger than 22 kHz. We present here an extension of these experiments to the entire frequency range of 1 to 100 kHz (up to and including the thermal noise limit), in a larger tank. They confirm previous results, also showing the distinct acoustic expression of ice-related processes (e.g. melting, capsizing, scraping and colliding) and the role of high frequencies in their identification.

Keywords: ambient noise, Arctic, field measurements, tank experiments
1. INTRODUCTION

Global climate models and local weather forecasts rely on information about weather across oceans. These are usually derived mostly from satellite measurements, but are limited by the large field of view of the satellite and finite time scales. This is particularly true in the Arctic and Antarctic regions, because of satellite orbits and cloud cover (only transparent to microwaves). Acoustic measurements, taken underwater, are complementary in that they can monitor areas and timescales not accessible by ship or satellite measurements. The work presented here aims to investigate more rigorously the potential uses of underwater acoustics to monitor ocean weather patterns, extending across wider frequency ranges and investigating processes not covered by previous studies.

Ambient noise underwater has been studied for several decades [1-3], and recent work [2-4] has shown that average powers at specific frequencies can be used as acoustic discriminants. These frequency ranges can be attributed to specific weather processes and therefore used to separate one process from another. However, authors have disagreed over the most appropriate intervals – Nystuen [5] used bands of 4-10 kHz and 10 – 30 kHz to separate different types of rain (Figure 1, left), after Black et al. [4], whereas Quartly et al. [3] separated weather events using 5 and 25 kHz with varying success (Figure 1, right). Very few studies have looked at the extension and applicability of these techniques to fjord environments with melting ice, in particular at higher frequencies.

Field measurements were carried out in Summer 2007 in Kongsfjord (Svalbard). A broadband hydrophone (effective bandwidth 100 Hz – 48 kHz) was deployed 10 m deep at regular intervals in deep water from the mouth of the fjord to the glaciers at its end. Ten long recordings were obtained at six different locations, including ambient noise from wind and small waves, light rain, marine mammals, growlers and bergy bits. Section 2 shows how the acoustic signatures of different weather events can be separated using Principal-Components Analysis, with clear frequency bands centred on 5, 15 and 45 kHz respectively. To clarify the contributions from individual processes, and in particular those related to icebergs and growlers, several laboratory experiments were conducted (Section 3). The first used a relatively small tank. They simulated different types of rain, different wind speeds and
mimics of small growlers, whose acoustic noise was recorded with the same hydrophone used in the Arctic (0.1-48 kHz). They were expanded using a larger tank and a hydrophone with flat frequency response (0.1 Hz-100 kHz), looking in more detail at the acoustic expression of ice-related processes, such as melting rates and collision.

2. ARCTIC FIELD MEASUREMENTS

2.1. Acoustic Recordings

The measurements were taken along the fjord of Kongsfjorden, 20-km long and 4-10 km wide. Mapping [6] shows its bedrock has relict sub-glacial, ice-scoured topography overlaid by a thin (<10 m) sediment cover. Depths vary from 360 m to 60 m, with some deeper depressions (up to 400 m) in the middle of the fjord. It shares a common mouth with neighbouring Krossfjorden, and where the two connect, a submarine glacial trough is formed and acts as a deep-water connection across the shelf [7]. All of these features lead to complex oceanography (close to recording station A, Table 1). Five tidewater glaciers act as a source of freshwater, and the West Spitsbergen Current carries relatively warm and saline water along the coast, and this influence is combined with Arctic and Atlantic water masses which causes temperature and salinity changes, meaning that Arctic and boreal ecosystems are mixed [8]. Kongsfjorden is also an important feeding ground for marine mammals and seabirds, many of whom use the icebergs as vantage points.

<table>
<thead>
<tr>
<th>Station A</th>
<th>78°59'45.7&quot;N ; 11°34'47.4&quot;E – Duration 5'00&quot;. Light rain. Small waves. No ice. Depth ~ 300 m – Closest shore ~ 3 km away. Whale less than 50 m away + distant ship (~ 5 km)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Station B</td>
<td>79°00'53.6&quot;N ; 11°30’00.4”E – Durations: B₁ (1’20”), B₂ (2’01”), B₃ (3’01”). Light rain. Very small waves. No ice. Depth ~ 300 m – Closest shore ~ 4 km away. Whale visible at the surface + ship at horizon (&gt; 5 km)</td>
</tr>
<tr>
<td>Station C</td>
<td>78°59’30.9”N ; 11°40’41.8”E – Durations: C₁ (0’46”), C₂ (3’01”), C₃ (3’01”). Light rain, increasing. No waves. No ice. Depth ~ 300 m – Closest shore ~ 3 km away.</td>
</tr>
<tr>
<td>Station D</td>
<td>78°58’47.4”N ; 11°46’53.4”E – Durations: D₁ (3’00”), D₂ (1’31”). No rain. No waves. No ice. Depth ~ 300 m – Closest shore ~ 2 km away.</td>
</tr>
<tr>
<td>Station E</td>
<td>78°58’01.0”N ; 11°52’26.9”E – Duration: 2’01”. Very light rain. Very small swell. Small icebergs (growlers and bergy bits) ~20 m away. Depth ~ 250 m – Closest shore ~ 1.9 km</td>
</tr>
<tr>
<td>Station F</td>
<td>78°57’09.4”N ; 11°57’42.2”E – Duration: 2’00”. No rain. No waves. No ice. Depth ~ 200 m – Closest shore ~ 1.9 km away. In front of Ny-Ålesund harbour, with cruise ship at berth.</td>
</tr>
</tbody>
</table>

Table 1: Measurements details at each recording station. Several recordings were taken at stations B, C and D to measure ambient noise stationarity.

Ambient noise was measured with a single hydrophone SQ26-07 (manufactured by Cetacean Research Technology). It is omnidirectional, as verified in earlier tank tests. Its
operational frequency band ranges from 10 Hz to 50 kHz. Its sensitivity is nearly constant at -168 dB re. 1V/µPa from 100 Hz to 30 kHz; it decreases linearly to -174 dB re. 1V/µPa at 35 kHz and increases again to -168 dB re. 1V/µPa above 40 kHz. The preamplifier gain was fixed at 25 dB and the deployment depth was ~ 9.5 m each time. Data from the hydrophone was input to a digital recorder M-Audio Microtrack 24/96 (sampling at a frequency of 96 kHz). Measurements covered an effective broadband frequency range of 10 Hz to 48 kHz. They were acquired in water always deeper than 200 m and in several distinct environments – Table 1 gives details of the 6 positions and the recordings taken at each. The wind speed was measured as 11 km/h (Sea State 2) at a nearby meteorological station; it is a lower estimate of that in the fjord and matches visual observations. Wind chill temperatures always remained sub-freezing. The different acoustic measurements are recorded in 24-bit .WAV format, ensuring full access to the normalised waveform.

2.2. Frequency Analyses

The measurements were split into 8,192-point segments (of duration ~ 85 ms comparable to that used in other studies, e.g. [4]), with a 10% overlap between segments to minimise the risk of missing events. The power spectra were averaged over frequency ranges of 1 kHz, from 1 to 48 kHz. Plots by individual frequency bands produced valuable results [9] but to address the variations between studies, and to follow the recommendations of Black et al [4], we used Principal-Components Analysis to identify the most significant frequency bands, by maximising the variance between combinations of separate frequencies. The results of this analysis are shown in Figure 2, highlighting three bands centred on 5, 15 and 45 kHz respectively. These bands explain 91.6% of the overall frequency variance.

![Figure 2: Distinctive frequency bands were identified by Principal-Component Analysis.](image)

They match the experimental and theoretical studies of present physical processes [4, 5, 10]: wind-related processes can be identified with band $X_1$ (~ 5 kHz) and rain-related processes with band $X_2$ (~ 15 kHz). Band $X_3$ (mostly ~ 45 kHz) is more difficult to explain, but after plotting the results using the three discriminants (Figure 3), $X_3$ is seen as a combination of ice-related processes with contaminations from shipping and marine mammals (a whale was visible at the surface near measuring station A, and although it later dived, its vocalisations can be clearly heard in the rest of the recording).
3. TANK EXPERIMENTS

3.1. Small-Tank Experiments

The analyses of these Arctic measurements confirm the conclusions of previous studies by other workers (e.g. [3-5]) for wind- and rain-related processes. To better interpret our measurements and how the combination of individual processes (such as wind and rain) contributes to the ambient noise in a field of melting icebergs, laboratory experiments were conducted in 2008 using a small tank (1.8 m long × 1.2 m wide) filled with 1.2 m of water [9]. The hydrophone used in the field survey was placed at mid-depth and several weather-related processes were simulated. Background noise was recorded before and after each experiment to put these results in perspective. A large fan with variable settings simulated wind; although the noise from the fan itself did not contribute to the background noise, the wind thus produced was very small and should not be seen as representative of field conditions. A hose simulated different types of rain: on its “cloud” setting, it produced a fine mist of small droplets, and on its “fan” setting, it produced larger-size droplets. Rain rates could be varied and were measured each time (using a simple bucket). To simulate freshwater icebergs calving from glaciers, ice blocks of 0.05 m$^3$ were produced using the method of Barker and Timco [11]: frozen blocks of fresh water are ground into small pieces, sieved and refrozen after addition of icy water, as in the actual formation processes. Melting was recorded continuously and episodes of breaking/capsizing of sub-blocks could be identified acoustically and visually.
Acoustic signals from the hydrophone were amplified by 25 dB (as in the Arctic field measurements) and recorded with the computer sound card. The maximum sampling rate was 44.1 kHz, meaning that frequencies could only be measured up to 22 kHz. This prevented the use of the acoustic discriminants identified on the field measurements (more particularly X₃, centred on 45 kHz). Using frequencies of 5 and 15 kHz (centre frequencies of bands X₁ and X₂, respectively), we showed in [9] that the different processes can be distinguished (Figure 4). They correspond to separate clusters of points. Background noise corresponds to a very tight cluster of points. Melting ice corresponds to another tight cluster and as the ice melts, its acoustic signature logically merges with that of the background noise. There are two clusters of points for rain, attributed to variations in rain rate and raindrop size, in orthogonal directions.

![Fig.4: Small-tank measurements of simulated weather processes, showing relative levels at frequencies of 5 kHz and 15 kHz respectively. Note the clear separation of individual clusters and how they can be attributed to specific physical processes.](image)

### 3.2. Medium-Tank Experiments

The first series of experiments showed the interest of laboratory experiments to study one physical process at a time, or a combination of specific processes. There were concerns however about the small size of the tank, with the possibility of reflections from the side walls, and the general noise level of this laboratory. Two important drawbacks of these first experiments were the lower sampling rate and the varying sensitivity of the hydrophone. As a result, the next series of experiments used another laboratory, in a quieter environment and with a much larger tank (5.1 m long by 1.5 m wide, filled with 1.8 m deep water). A B&K-8103 omnidirectional hydrophone was used, with a constant sensitivity of -211 dB re. 1V/µPa between 0.1 Hz and 100 kHz (as calibrated by the manufacturer). Its signal was amplified by 30 dB and band-pass filtered between 10 Hz and 100 kHz using an Ortec-Brookdeal 9452 amplifier, before sampling at 500 kHz with a LeCroy LT-264 digital storage oscilloscope controlled with LabView. Weather processes were simulated in the same way as in the small-
tank experiments, and measurements spanned 3-minute periods, sampling in 0.2-second bursts every 10 seconds. The melting of different iceberg types was recorded for periods of 6 hours each time. The collision and scraping of individual blocks of similar sizes were also measured over 3-minute periods.

Background noise was generally lower in this setting, and the overall results show the same separation of individual processes as in Figure 5. One question was the role played by the formation of the ice: frozen freshwater blocks were compared with blocks prepared following the method of [11]. Both types of ice proved to have relatively similar acoustic signatures, but the frozen/sieved/refrozen ice blocks of [11] were several dB louder at low frequencies (<3 kHz) and very high frequencies (85-88 kHz and 97 kHz). One tentative explanation is the role of air inclusions in the refrozen ice, implying that the mode of formation of the icebergs, or the amount of heterogeneities, would influence its noise when melting. Its importance still needs to be assessed for a field of melting icebergs of different sizes, subject to wind and wave action, and interaction between ice blocks. These tank experiments show that, at frequencies below 1 kHz, the acoustic signature of colliding ice blocks is distinct from the melting ice (louder by 1-2 dB). It is indistinguishable from melting ice at all other frequencies, except 8-10 kHz (with distinct episodes 2-3 dB louder) and at specific frequencies of 73 and 97 kHz. Acoustic recordings of ice blocks scraping against each other show distinct, loud events at frequencies of 5 kHz and below, with a large variation in amplitudes (up to 7-db louder). Between 8 and 35 kHz, the number of distinct events decreases to only a few in the 3-minute recording period, louder on average than all other measurements by 3-7 dB. Interestingly, frequencies of 89-91 kHz show louder (by 2-3 dB) events, where the scraping of ice blocks against each other is distinct from melting ice, colliding ice (to a degree) and all other weather-type measurements (i.e. different types of rain and wind). As well as confirming previous results in a better controlled and calibrated setting, these new experiments show the acoustic variation brought by the formation and behaviour of individual ice blocks. Using higher frequencies does not give better separation of individual weather processes, as measurements get more clustered, but ice-related processes have distinct acoustic signatures around 35 kHz and between 70 and 100 kHz.

4. CONCLUSIONS

The study of underwater acoustic noise generated by weather processes benefits from several decades of research and many applications [1, 3-5, 10-13] although work in Arctic fjords has been limited to Alaska [14, 15]. The measurements presented here correspond to one day of experiments, from a surface vessel, therefore covering a small range of processes: small waves, light wind, light rain, melting icebergs, marine mammals and passing ships. This range was however sufficient to justify the use of Principal-Component Analyses. The resulting frequency bands confirm the current consensus about the main frequency characteristics of wind, rain and other processes. Tank experiments were used to investigate the acoustic contributions of the different physical processes, particularly the melting of small ice blocks (similarly sized to the growlers encountered in Kongsfjord). Using two acquisition set-ups in two tanks, these experiments concurred in showing the importance of rain rate and drop size (as mentioned previously [3, 5]) but they mostly showed the variations of ice-related processes and the necessity to record at high frequencies, close to the thermal noise limit (100 kHz). Ice melting research had looked at frequencies from 0.1 to 10 kHz (e.g. [16]), but higher frequencies (> 30 kHz) have an influence, particularly when ice blocks collide or scrape. The next experiments will involve larger blocks and look at interactions with rain and waves. They will aim to identify acoustically and physically the processes
leading to these high-frequency events, and potential uses in ambient noise recording systems, for example to monitor glacial melting.

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Acoustic technologies for observing the Arctic Ocean

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Abstract: Operational ocean monitoring and forecasting systems, such as MERSEA-IP, (2005-2008, http://www.mersea.eu.org/) and MyOcean (2008-2011), combine observations from different satellite remote sensing techniques and in-situ open ocean measurements (mainly Argo floats and moorings) with ocean circulation models through advanced assimilation techniques. In the Arctic Ocean, however, the internal water-masses, circulation and seafloor is largely unknown due to lack of observations. Development of an Arctic Ocean Observing System (AOOS) is still hampered due to significant lacks in technology and investment. The objective of the two European projects DAMOCLES-IP and ACOBAR is to develop and test new technologies to measure ocean and sea ice to obtain new knowledge of the Arctic Ocean environment, and to contribute to the development of a future AOOS. This includes the implementation of an ocean acoustic system for tomography and navigation of gliders and floats in the Fram Strait, prepare gliders and floats for under ice operations, deployment of sophisticated Acoustic Ice Tethered Platforms for internal ocean observations, development of communication between underwater platforms and approaches to achieve real time capability from sub-surface units. In this presentation we will focus the technological achievements within DAMOCLES and how these will be followed up within the ACOBAR project with an emphasis how acoustic technologies will contribute to the development of an Arctic Ocean Observation System.
Correlation between ocean noise and changes in the environmental conditions in Antarctica

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Abstract: Long-term variations in the intensity and occurrence frequency of transient noise signals spreading to the Southern and Indian Oceans from the Eastern Antarctic continental shelf were analysed using the sea noise data recorded over 6 years at a hydroacoustic station deployed off Cape Leeuwin in Western Australia as part of the International Monitoring System of the Comprehensive Nuclear-Test-Ban Treaty. These signals were found to be produced primarily by ice breaking events on the Antarctic ice shelves and icebergs. Strong seasonal components can be observed in both mean intensity and occurrence of the Antarctic ice events detected at Cape Leeuwin. However, the seasonal cycle in the occurrence frequency is significantly delayed, by approximately 3 months, relative to the seasonal variation of the mean signal intensity. The correlation of variations in the intensity and occurrence frequency of signals from ice events with changes in different environmental characteristics, such as air and water temperatures, wind speed and surface wave height, over the Eastern Antarctic coastal zone are analysed in this study.
ESTIMATION OF MACROPHYTES USING SINGLE AND MULTIBEAM ECHO SOUNDERS AND SIDESCAN SONAR IN ARCTIC FJORDS (HORNSUND AND KONGSFJORD, WEST SVALBARD)

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Abstract: The characterisation of sea vegetation over large areas and at all depths needs very efficient and cost-effective tools such as acoustic methods. We present here an acoustic method for the evaluation of benthic habitats in Arctic periglacial environment. The great intensity and rapid changes of morphodynamic processes in the biotic environment of Arctic fjords make them prime areas to research climate change impacts on ecosystems. This paper presents comparative results of study on the spatial distribution and biomass of macrophytobentos in two Svalbard fjords, Hornsund and Kongsfjord. Acoustical data in Hornsund fjord was collected by systematic surveys with co-registered single-beam echosounder and side-scan sonar data. For classification of acoustical imageries of bottom covered by algae, we developed algorithms using multidimensional wavelet decomposition of echo signals and fuzzy-logic clustering. Classification results are supported by video recordings and biological samplings by divers. Another, slightly different data collection approach was used in Kongsfjorden, with simultaneous single-beam and multibeam echosounder recordings, verified by biological samplings and observations. Both methods provide maps of phytobenthos distribution and biomass estimation in the fjords investigated. Comparison of single-beam and multibeam echosounder results proved a great opportunity for understanding the visualisation of macrophytes by multibeam systems and potential improvements to associated macrophytobentos recognition techniques.

Keywords: Seafloor classification, Arctic environment, habitat mapping, algae, image classification
1. INTRODUCTION

Periglacial ecosystems, with intense morphodynamic processes and rapidly progressing changes in the biotic environment, are a difficult place to conduct research despite their key role in climate changes investigations. Underwater acoustics can provide efficient tools for bottom type recognition and estimation of spatial distribution and biomass of macrophytobentos. All this data supports present analyses of fjord ecosystem functioning and allows modelling its further behaviour with global climate change.

The main aim of this study was to analyse the use of acoustic devices from small boats for habitat mapping procedures and design tools for macrophyte recognition and evaluation. All surveys were done in two glacial fjords: Hornsund (2005) and Kongsfjord (2007), both part of West Spitsbergen (Svalbard) (Fig.1). They are open fjords, without a sill at the entrance, so every exchange at the shelf-fjord boundary has a significant impact on the entire fjord’s ecosystem [1]. Large-scale investigations can map, follow and accurately quantify these changes from year to year.

![Fig.1 Svalbard Archipelago and west coast of Spitsbergen with outlined research areas: 1-Kongsfjord and 2-Hornsund](image)

2. METHODOLOGY

The primary tool employed in this study is a single-beam, down-looking echosounder, as it is a widespread and well-understood sensor. The shape of backscattered echoes carries a lot of information about bottom hardness and roughness; it also allows to estimate kelp heights (Fig.2).

Macroalgae grow in the littoral and sub-littoral zones of the fjord, mainly from 0 to 20 m on hard bottom. They are rather large underwater plants [1]. Depending on the exact species, their blades can reach up to 2-3m high. Of particular importance for acoustic studies, their body reflects acoustic signals well at typical echosounder frequencies.

During the Hornsund experiment, the single-beam echosounder Simrad EK500 was used. Its operating frequency is 120 kHz and it has a beamwidth of 10°. It was used with a pulse length of 0.3ms. As shown on a typical echogram (Fig.2, left), macrophytes are clearly visible on the bottom as lighter clouds of separate reflectors above the distinct red (dark) bottom line, but there is also a lot of particle suspension in the water column (green clouds).

In the Kongsfjord experiment, we used instead a Biosonics DTX single-beam echosounder. Its operating frequency is much higher (420 kHz) and its 3-dB beamwidth is significantly narrow (5.2). At a range of 0.5 -30 m and with a pulse length of 0.1ms, it
performs with a spatial resolution of ~0.3–0.9 m (according to depth) × 0.08 m, making it an efficient device for algae investigation (Fig. 2, right).

Fig. 2 Comparison of typical echograms from two different echosounders. Left: Simrad EK-500 (120 kHz). The green area (lighter) on the red (dark) line marks the presence of algae, reducing the overall backscattering strength from the stony bottom. Right: Biosonics DTX (420 kHz). In this case, algae are in light blue tones.

During post-processing echoes were classified as a function of bottom type and algae presence. Each signal envelope was treated as a separate source of information about the bottom (Fig. 3.). During parametrical analysis and segmentation algorithms [2,3] places with macrophytes were identified. Using backscattering strength differences between water, kelps and bottom, borders between their respective areas were found, algae presence were established and their heights were calculated.

Fig. 3 Comparison of two ping echo envelopes, both from places with algae. Red marks show the part of the signal with information about the algae, just above the bottom.

3. KONGSFJORD RESULTS AND MULTIBEAM SYSTEM

The Kongsfjord experiment took place in Summer 2007, with logistical support from the Ny-Ålesund AWIPEV research station. The Imagenex 837 Delta-T multibeam (MBES) sonar is used mostly for bathymetry measurements but has shown a strong potential for habitat mapping work. It is operating at a frequency of 260 kHz, transmitting 120 beams per swath. Beamwidths are 1° across-track and 20° along-track. This MBES principally acquires bathymetry measurements for each beam, but it is possible to acquire backscattering strengths over the entire water column (i.e. not only for the bottom). They can then be normalised by normalising to common levels (derived for example from modelling for specific bottom types and morphologies). It means that, from the centre beam(s), it is possible to create an
echosounder-like profile. Fig. 4 shows a typical MBES frame (using all 120 beams), at the acquisition stage and with some basic interpretation. The nominal vertical resolution of the MBES at this range is 1 cm, high enough to resolve the broad, flat leaves typical of the Laminaria macrophytes visually observed below the boat and confirmed by echosounder Biosonics data and direct observations. Similar MBES systems have already been used with success [4,6] for habitat mapping but not in the Arctic environment and not for this type of macrophytes.

![MBES acquisition frame with Laminaria macrophytes and gravelly bottom. The depth is 5 m and the vertical resolution 1 cm. The red (darker) sub-horizontal stripe in the middle corresponds to the fjord’s bottom and the yellow-green (lighter) clouds to actual macrophytes.](image)

**Fig. 4** MBES acquisition frame with Laminaria macrophytes and gravelly bottom. The depth is 5 m and the vertical resolution 1 cm. The red (darker) sub-horizontal stripe in the middle corresponds to the fjord’s bottom and the yellow-green (lighter) clouds to actual macrophytes.

![Derived macrophyte distribution with height scale and depth contour in South Kongsfjord (mid-fjord); b1 and b2 are places where biological samples were taken.](image)

**Fig. 5** Derived macrophyte distribution with height scale and depth contour in South Kongsfjord (mid-fjord); b1 and b2 are places where biological samples were taken.

Bathymetry from MBES was used to delimit the kelp range distribution in the euphotic zone. After Biosonics data classification, it was found that some echoes were wrongly interpreted as algae instead of mud, with a similar “cloud” over the bottom but much deeper than macrophytes. GIS combination with detailed bathymetry allows discarding these results and creating more accurate maps (Fig. 5). The top-left corner would suggest additional presence of algae, doubtful as the depth exceeds 40 m. General validation of the procedure shows a good accuracy overall. For example the dashed line in the middle, bottom part of Fig. 5 shows the Biosonics transect of Fig. 2 (right), with the same alga heights.
4. HORNSUND RESULTS AND SIDE-SCAN SONAR SYSTEM

This expedition took place in August-September 2005, with logistic support from the Polish Polar Station in Hornsund Fjord. Acoustic data were collected with the Simrad EK500 echosounder and EdgeTech DF1000 side-scan sonar, operating at 100 kHz and 390 kHz (but due to its better resolution, only the highest was used). Data were collected around the entire fjord in the euphotic zone, with concurrent video ground-truthing of the classification results. A typical portion of the research area is shown in Fig. 6, combining sonar data, algae height distribution and depth contours.

![Fig.6 Inner part of Hornsund Fjord (North). Maps of algae distribution and heights are combined with bottom morphology, derived from side-scan data and bathymetry.](image)

Similarly, single-beam recordings were classified and the spatial and height distribution of macrophytes were estimated. Side-scan sonar data was corrected using bespoke Matlab software. Its main role is to make slant range correction and TVG adjustments [5] and prepare the sidescan images for further classification. After this pre-processing, segmentation was done using two methods.

![Fig.7.Hornsund sidescan sonar classification results. Algae area-green(middle grey) is about 5,680 m². These results were validated by divers (between marked points).](image)

Promising results were obtained by using spectral, wavelet and fractal features of echoes computed inside the sliding signal window as the input to factor analysis and principal component analysis [2,3]. Although effective, this method was unfortunately very time-
consuming. The second classification method allowed faster data processing; it was based on
discrete wavelet analysis of echoes connected with texture analysis parameters: entropy and
standard deviation of image. All this data was clustered using Fuzzy C-Means (FCM). These
results are promising and allow accurate algae area estimation (Fig.7). The classified sonar
image (part of the entire dataset) covers around 6,630 m². The green (medium grey) area
shows algae presence and is about 5,680 m².

5. CONCLUSIONS

Results of research of distribution range and biomass estimation of macrophytobentos in
two Arctic fjords demonstrated that the proposed method of side-scan sonar analyzing in
combining with single beam echosounder registration, video recording, multibeam
bathymetry and biological probing provides proper and accurate recognition of areas covered
with vegetation. Further work should be done for MBES backscattering signal processing, in
particular to provide better algorithms for fast and efficient habitat mapping.

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Target Scattering

Organizer: Alessandra Tesei
DETECTION AND CLASSIFICATION OF A BURIED CYLINDRICAL SHELL FILLED WITH DRY MATERIAL

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Abstract: In previous paper (9\textsuperscript{th} ECUA Paris Acoustics'08 pp. 435-440) the detection and the classification of an air-filled cylindrical shell buried in saturated sand were presented. The Method of Isolation and Identification of Resonances (MIIR) was used to obtain resonance spectra characteristic of the target. It allowed us to minimise the influence of the reverberation signal, especially the reverberation signal due to the water-saturated sand interface. In this paper, the influence of the filling of the cylindrical shell is particularly studied. The experimental conditions are identical to the one of the previous study. The cylindrical shell is buried at various depths in a saturated sand. It is excited by a broadband impulse with a central frequency 500 kHz. In a first part the backscattering spectrum and the resonance spectrum of the air-filled cylindrical shell immersed in free water are plotted to serve of references. In second part the cylindrical shell, immersed in free water, is filled with dry sand, the resonance spectrum is plotted and is compared to the one obtained when the target is air-filled. In a last part, the resonance spectrum of this cylindrical shell filled with dry sand and buried in saturated sand at various depths is experimentally plotted. The resonance spectrum is constituted by the resonances of the $S_0$ circumferential wave. From this resonance spectrum, it is possible to detect and to classify a man made object.

Keywords: Underwater acoustics, Target scattering, Buried target, Resonance spectrum.
1. INTRODUCTION

Since some years several authors have studied the possibility to detect and to classify objects buried in silt at the bottom of sea [1-3]. In laboratory, others researchers have developed methods to separate the acoustic signal in relation to the buried objects and the acoustic signal reverberated by the silt [4,5]. In this paper, the Method of Isolation and Identification of Resonances is used to detect and to classify buried objects in very thin sand and partially filled with thin dry sand [6].

The Method of Isolation and Identification of Resonances (MIIR) that experimentally verifies the Resonance Scattering Theory (RST) developed in numerous papers [7,8] has shown that it is possible to characterize cylindrical and spherical shells from acoustic resonance spectra. The authors of the RST have shown that the backscattering acoustic spectrum of a cylindrical shell, insonified by a plane wave perpendicularly to its axis, is constituted by a background due to the reflection of the incident wave on the object, and by resonances in relation to the propagation of circumferential waves which circumnavigate around the target in the shell or at the interface between the target and the water. The characteristics of the circumferential waves are strongly influenced by the material and the radius ratio $b/a$ ($b$: inner radius; $a$: outer radius) of the shell. In the frequency domain studied in this paper, only the resonance modes of two types of circumferential waves are observed: the $A_0^*$ wave labelled also $A$ wave [9] and the $S_0$ wave. These waves, during their propagation, are coupled with the fluid surrounding the cylindrical shell and reradiate progressively their energy in this fluid. The coupling of the $S_0$ wave is weak and this wave can propagate during several circumferences before to vanish, whereas the $A$ wave is attenuated after few number of circumferences. In this work, the influence of the sand on the propagation of circumferential waves in and around the buried tube is specially analyzed. The resonance spectra are plotted to characterise this buried tube partially filled with dry sand, thanks to the acoustic method MIIR.

2. EXPERIMENTAL SETUP

The experimental arrangement is described on figure 1. The studied object is an aluminum cylindrical shell. Its geometrical characteristics are: length $L=25$ cm, outer radius $a=1$ cm and the radius ratio $b/a=0.95$. The physical characteristics are: the velocity of longitudinal and shear waves respectively $C_L=6380$ m.s$^{-1}$ and $C_T=3100$ m.s$^{-1}$; the aluminum density is $\rho_A=2900$ kg.m$^{-3}$. This target is immersed in water of density $\rho_w=1000$ kg.m$^{-3}$ with an acoustic velocity $C_w=1470$ m.s$^{-1}$, or buried in water-sand mixture of density $\rho_s=1250$ kg.m$^{-3}$ with an acoustic velocity $C_S=1650$ m.s$^{-1}$.

The tube can be immersed in water or buried in sand. It is partially filled with dry sand (Fig. 1(B)). The sand grains in the tube cavity are identical to the sand grains in which the tube is buried. The broadband transducer, with central frequency 500 kHz, perpendicularly insonifies the tube and the interface of the water/water-sand mixture. The tube is excited by a short impulse with large band pass. The backscattered signal is recorded by the digital oscilloscope and transferred to a micro computer to be treated by a Fourier Transform.
3. EXPERIMENTAL RESULTS

To understand the experimental results, it is necessary to know the theoretical results of the acoustic backscattering from the used aluminum tube \((b/a=0.95)\) immersed in free water. These results are described in the paper of reference 5. The calculated resonance spectrum shows us resonances related respectively to the \(A\) wave and to the \(S_0\) wave.

3.1. Empty tube immersed in free water

Figure 2 presents the results of the experimental acoustic backscattering of the previously described aluminum tube immersed in free water and filled only with air. The tube axis is horizontal. The time signal is displayed on Fig.2(A). The reflected echo, the \(A\) wave echoes and the \(S_0\) wave echoes are indicated by arrows. The \(A\) wave echoes are constituted by several periods of sinusoid and have a narrow bandwidth contrary to the \(S_0\) wave echoes which have a large bandwidth. Four echoes of the \(A\) wave are observed as well as several echoes of the \(S_0\) wave. These last ones have a small amplitude which slowly decreases between two consecutive echoes. This small decreasing is due to the weak coupling between this wave and water. The \(S_0\) wave makes numerous rounds in the tube shell before to vanish.

The backscattering spectrum (Fig. 2(B)) confirms these remarks. The \(A\) wave resonances are observed in a reduced frequency window \((10 < k_wa = 2\pi aF/C_w < 35)\), not in the whole of the explored frequency band. The first resonances, at left of the figure, are thin and their thickness increases with the frequency. The resonances of the \(S_0\) wave are wide in low frequency and their width decreases with the frequency. The influence of the \(A\) wave is important in the frequency window \((300 \text{ kHz} < F < 700 \text{ kHz})\) and hides the most resonant effects of the \(S_0\) wave in this frequency band. The resonance spectrum on figure 2(C) is obtained applying a Fourier transform to the impulse time signal of figure 2(A) after having replaced the specular echo by zeroes. The resonances appear as peaks with a good resolution. The background signal is suppressed. \(A\) wave resonances are indicated by black points and \(S_0\) wave resonances by arrows. Between the frequencies 400 and 700 kHz, the \(A\) wave
resonances are wide and create a secondary background. This spectrum can be used to classify an underwater target.

Fig. 2: Experimental results obtained with the empty tube immersed in free water. (A): time signal, (B): backscattered spectrum, (C): resonance spectrum.

3.2. Half sand-filled tube immersed in free water

Before to study the acoustic backscattering by a tube buried in sand, the backscattering by a half sand-filled tube (Fig. 1(B)) immersed in water is measured. The sand in the tube cavity is dry. The tube is horizontally immersed, in the experimental tank, under the same conditions that the one of § 3.1. Figure 3 gives the experimental results.
Fig. 3: Experimental results obtained with the half sand-filled tube immersed in free water.  
(A): time signal, (B) and (C): two resonance spectra.

Figure 3(A) presents the time signal. The observed echoes are: the reflected echo \( \Theta \), one echo of the \( A \) wave \( \Omega \) and a series of echoes of the \( S_0 \) wave \( \Theta \). The dry sand in the cavity modifies the propagation of the \( A \) wave. Figures 3(B) and 3(C) show us two examples of resonance spectra. In the first case (Fig. 3(B)) only the reflected echo \( \Theta \) is replaced by zeroes. A background due to the \( A \) wave echo is observed in the frequency window identical to the one of figure 2(C), the resonances of \( A \) wave are not detected. In the second case the reflected echo and the \( A \) wave echo are replaced by zeroes, only the \( S_0 \) wave resonances are detected. The dry sand in the cavity deadens the \( A \) wave propagation.
3.3. Half sand-filled tube buried in sand/water mixture

In this paragraph, the half sand-filled tube is buried in sand water mixture at two centimeters of the water - sand/water mixture interface. Figure 4 gives the experimental results.

Fig. 4: Experimental results obtained with the half sand-filled tube buried in sand/water mixture. (A): time signal, (B) and (C): two backscattering spectra, (D) resonance spectrum. Figure 4(A) presents the time signal, ① is due to the reflection of the incident wave on the water – sand/water mixture interface, it is the main reverberation echo, ② is the reflected
The possibility to detect and to classify a cylindrical shell sand-filled and buried in sand/water mixture with a resonance spectrum is shown in this paper. Spectra have shown the influence of the $S_0$ wave in the formation of resonances. However this classification is less accurate because the $A$ wave resonances, which allow to determine the radius ratio with the frequency window, are not observed when the tube is buried in sand/water mixture.

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Modelling and Measurement of Backscattering from Partially Water-filled Cylindrical Shells

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Abstract: The backscattering from partially water-filled cylindrical shells has been studied using finite element (FE) analysis and experimental measurements. Low to medium frequencies, corresponding to $2 \leq k a \leq 30$ (where $a$ is the shell outer radius), were used in the experiments. These were performed in both an indoor laboratory tank and in a reservoir, using appropriate shell dimensions and wide band signals (Ricker pulses and frequency chirps) generated by a parametric array. The form functions were calculated for a horizontally incident wave as a function of the water-level in the shell and also measured experimentally. The measured form functions are in very good agreement with the FE modelling. The backscattering from a partially-filled cylindrical shell, filled to three quarters of the inner diameter, has been investigated as a function of the elevation angle of the incident wave, and in particular as wave direction changes from horizontal to vertical (with the wave from above). Comparisons with fully air-filled and fully water-filled shells indicate that the cylinder resembles the former when the wave is incident from above and the latter when the wave is incident horizontally. The short pulse duration of the wide-band Ricker pulses made it possible to observe a number of waves after the specular return from a partially-filled shell with a high filling fraction. These included the first few contributions due to $S_0$ waves generated by the wave incident on the front of the shell, the reflection from the back wall and a contribution due to an $S_0$ wave generated by the incident wave at the back wall of the shell. The inversion of the FE model data enables the expected waveforms to be predicted and compared with the measured results.
Wideband Measurement of Acoustic Scattering from a Cylindrical Shell using a Parametric Array

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Abstract: An experiment to measure the backscattering from a cylindrical shell using a parametric array has been performed at the National Physical Laboratory (NPL) facility on Wraysbury reservoir. The purpose of the experiment was to obtain information on the frequency responses of the target for different angles of incidence, over a wide frequency range, for comparison with theoretical predictions. A parametric array system developed by Loughborough University was used for the experiment. The transmitting transducer consists of 27×27 elements arranged in a square planar array with a centre frequency of 75 kHz. The array can be beam-steered in one plane within a sector of ±20 degrees. A number of secondary waveforms were used, including tone bursts, Ricker pulses and frequency chirps, with various centre frequencies and bandwidths. A vertical line array of six elements was used to receive the backscattered signals. Overall the response was measured at secondary frequencies from 2 to 24 kHz and primary frequencies from 63 to 87 kHz. A mild steel tank was used in the experiment; this was 1.4 m long with hemispherical ends, and had a radius of 0.28 m and a wall thickness of 4 mm. The backscattering from the tank when partially and fully filled with water was measured. The target was suspended from a rotation mechanism so the responses as a function of angle could be measured. The response was measured under free field conditions and with the target on the reservoir bottom. Some measured results in terms of the time and frequency responses as a function of incident angle are presented here and comparisons are made between the measured results and theoretical and numerical simulations.
Exact and approximate techniques for scattering from targets in a waveguide

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Abstract: Accurate solution of scattering from a target in a waveguide requires that the scattering and propagation problems be solved as a single boundary value problem. This can be accomplished by solving the wave equation in an environment that contains both the target and the waveguide and satisfying boundary conditions on the surface of the target as well as the boundaries of the waveguide. The virtual source technique provides the means to do this by replacing the target with a collection of sources whose amplitudes are determined by applying the boundary conditions on the surface of the target. This method converts the problem of scattering from a target in a waveguide to a multi-source propagation problem. In this paper we compare the virtual source solution for scattering from a target in a waveguide to other less accurate, but numerically efficient solutions. We will also examine various ways to speed up computations within the virtual source technique. These include the use of various propagation models to propagate the field produced by the virtual sources to the receiver. We will compare various models and discuss the advantages and disadvantages of each model.
Panel transmission measurements: the influence of diffraction

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Abstract: In underwater applications, materials are often characterised by the acoustic transmission and reflection properties of an ideal, infinite planar sample, however laboratory measurements are often performed on samples of finite size. This paper explores the influence of the diffracted wave on the resulting measurements, particularly its effect on the measurement dynamic range for highly reflective materials. Boundary Element predictions are presented for perfectly hard, perfectly soft and steel panels, and for mixed boundary conditions the effect of panel orientation on the achieved dynamic range is explored. The efficacy of different measuring schemes including single hydrophone, pseudo random array and intensity probe is examined and results compared to those obtained on real panels. The results show the advantage of array based processing on reducing the influence of diffraction on transmission measurements.
TIME REVERSAL ITERATION FOR ELASTIC OBJECT DETECTION, CLASSIFICATION AND ECHO ENHANCEMENT

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\textbf{Abstract:} In this paper, a method is introduced for elastic object detection, classification and echo enhancement with a single transceiver. Iterative time reversal will gradually cause echo waves to converge to a single frequency signal, which corresponds to a resonance mode of the elastic object, and this intrinsic frequency can be used for target classification, and the return level is enhanced compared to the echo of the first iteration. Tank experiment is performed with an aluminium ball as the target, and the results illustrate the feasibility of the method.

\textbf{Keywords:} Elastic scattering, iterative time reversal
1. INTRODUCTION

Since wave equation in non-dispersive and motionless media holds time reversal invariance and spatial reciprocity, the iterative time reversal will lead echoes gradually to converge to a dominant narrowband resonance mode of the elastic object, thus excites body eigenvibration and enhances return level in noisy and reverberant environment. The center frequency of the converged-upon signal corresponds to a resonance frequency of the elastic object, and this important acoustic signature could be used for classification and identification purposes.

Previous work [1] demonstrates the efficiency of this method with numerical studies to solutions of a sphere target buried in the two layered media, [2] tests the potential of the echo enhancement ability by tank experiments with a spherical shell, a small stone, and a complex steel object as targets. Similar work [3][4] carry out some scaled laboratory investigations in buried resonant target detection with single channel model. This work just derives some intrinsic mechanisms of the methodology, and carries out the tank experiment to demonstrate the feasibility of elastic object detection, classification and echo enhancement.

The process schedule of the method is as follows:

1) Illuminate the target with a broadband interrogation pulse.
2) Measure and window backscattered wave and apply a band pass filter.
3) Reverse the time series and re-transmit it.
4) Repeat above procedures iteratively.

Tank experiment has been conducted with an aluminium ball hung in free space as the target. The results illustrate this technique can find out the positions of the object resonance peaks by changing the center frequency of the interrogation pulse and adjusting the corresponding filter, and the return level is enhanced compared to the first iteration. Only the stationary situations are considered here, and the common time factor $\exp(-i\omega t)$ of all formulas will be suppressed just for convenience.

2. THEORETICAL FORMULATIONS

We take the homogeneous and isotropic aluminium sphere with radius $a$ immersed in fluid as the target, which is centered at $r_s$, and the transceiver, which is a transmitter and receiver couple, is situated at $r_c$ with point-like model. The probe signal is first sent out by the transmitter, then scattered by the elastic sphere, and at last received by the receiver, the whole process can be looked as through a linear time invariant system which can be expressed in terms of the backscattering form function [5]:

$$h(\omega) := \alpha_e(\omega)\alpha_r(\omega)\frac{a}{2R^2}\exp(2ik_0R)f(\pi;\omega).$$

(1)

Here $\alpha_e$ and $\alpha_r$ are the transmission and reception acousto-electric transformation coefficients, respectively, and $k_0$ is the wave number in the fluid, and $R = \|r_s - r_c\|_2$ is the distance between the target and the transceiver. The form function of the elastic solid sphere $f(\theta)$, which denotes the steady state response of the target scattering, is derived by
expanding both the incident and secondary scattered pressure field in the spherical polar coordinate and determining the coefficients from the appropriate boundary conditions, and as the problem is investigated in a far field monostatic deployment, that is $\theta = \pi$.

Assume $e_0$ is initial broadband interrogation pulse, then the received signal is $e_0 h$. As time reversal operation conducted in the time domain is equivalent to phase conjugation in the frequency domain, we use superscript $*$ to denote the complex conjugate operator, then the time reversed signal of the first iteration is $e^*_0 h^*$, and of the second iteration is $e_0 h h^*$, and so on. So the renewed input signal for next emission after the $n^{th}$ iteration is

$$e_n(\omega) = \begin{cases} (e_0 \vert h \vert^{n-1} h^*) (\omega), & \text{if } n \text{ is odd}, \\ (e_0 \vert h \vert^n) (\omega), & \text{if } n \text{ is even}. \end{cases} \tag{2}$$

and the signal spectrum amplitude is $\vert e_n \vert = \vert e_0 \vert \vert h \vert^n$ for both cases, then the normalized version, which is divided by the maximum, will converge to

$$\frac{\vert e_n(\omega) \vert}{\max \{\vert e_n(\omega) \vert\}} \to 1_{\{\omega; \omega = \omega_0\}}(\omega) \text{ as } n \to \infty. \tag{3}$$

The indicator function of the set $A$ is $1_A(\omega) = 1$ if $\omega \in A$, and $1_A(\omega) = 0$ if $\omega \notin A$, and $\omega_0$ is the frequency corresponding to the maximum of the amplitude of the system transfer function,

$$\omega_0 := \arg \max_{\omega \in [\omega_1, \omega_2]} \vert h(\omega) \vert, \tag{4}$$

and the selected frequency interval $[\omega_1, \omega_2]$ is limited by the band pass filter. We always choose this band in the relative flat domain of the transceiver frequency response amplitude $\vert \alpha_r \alpha_t \vert$, which may also be calibrated, otherwise the signal may converge to the transceiver response peak, then the converged frequency point is

$$\omega_0 \approx \arg \max_{\omega \in [\omega_1, \omega_2]} \vert f(\pi; \omega) \vert. \tag{5}$$

Eq.(3) denotes the returned echoes will gradually converge to a harmonic wave, and its frequency corresponds to a resonance mode according to eq.(5), hence cause eigenvibration and enhance the return level. The form function is the intrinsic property of the target and can be looked as its fingerprint, iterative time reversal can find out the position of the maximums by changing the center frequency of the interrogation pulse and adjusting the corresponding interval of the band pass filter, and this signature can be used for classification and identification.

**TANK EXPERIMENTAL RESULTS**
As depicted in fig.1, the transceiver and the target are hung at the same depth in the tank with a free field monostatic deployment. The wideband pulse synthesized by the programmable signal generator is emitted by the transceiver in transmission mode, and the echo scattered by the aluminium ball is received by the same transceiver in reception mode and the signal is recorded with an oscilloscope. Both the signal generator and the oscilloscope are connected to a PC machine through USB connector and twisted-pair cable, respectively. All pre-processing procedures such as echo cutting, band pass filtering and result displaying, etc., are dealt with Matlab® software.

Fig.2 is a picture of the target, which is a 10 cm radius homogeneous and isotropic aluminium sphere.

Fig.3 displays the change in the backscattering form function amplitude both of numerical and experimental results. It is demonstrated the strong fluctuation behaviour even for regular shape of sphere, which is caused by a superposition of narrow resonance in individual partial waves. The form function is experimental measured by a reference hydrophone placed between the transceiver and the aluminium ball, the ratio of the root mean square pressure values of the incident and reflected is proportional to the amplitude of the form function, and the results show good agreement with the theoretical calculation.
The form function is the object’s instinct characteristic that is determined by the complicated structure and the material properties. The elastic body resonance can be excited with single channel iterative time reversal, and the echo will converge to a single frequency signal corresponding to a resonance peak. By applying different center frequency interrogation pulse and adjusting the filter, all resonance peak positions can be find out. Fig.4 illustrates two, one converged frequency is 12.5 kHz, corresponding to a resonance peak at $ka=2.55$, another is 56.5 kHz, and approximate the maximum at $ka=12$. The bias is mainly caused by the inclination of the transceiver frequency response and the ripples in the pass band of the filter, and the Gibbs oscillation of the spectrum is introduced by the rectangular window.

Fig.4: Received Echoes and their respective spectral amplitudes with different center frequency CW as initial interrogation signals. (a) is of 10 kHz center frequency and the converged frequency is 12.5 kHz; (b) is of 55 kHz center frequency, and the converged frequency is shifted to 56.5 kHz.

Fig.5 plots total signals recorded by the transceiver, and the first envelops are the normalized sending signals, and the second are the reflected echoes. We can see the amplitude of converged echo is enhanced compared the echo of first iteration, thus can improve the signal-to-noise ratio in noisy and reverberant environment. This is mainly because the iterative time reversal will make the echoes converge to a response peak, and cause body eigenvibration.
Fig 5: Signals recorded by the receiver with the 2 cycles 10 kHz center frequency CW signals as interrogation pulse. (a) is of the 1st iteration, and (b) is of the 8th iteration. The first envelopes are the sending signals directly picked up by the transceiver, which are normalized, and the second envelopes are the echoes scattered from the aluminium.

3. SUMMARY

Time reversal iteration can be used for elastic object detection, classification and echo enhancement, and it is fast, reliable and can be realized with a single channel. The converged eigenfrequency can provide information about the size and constitution of the target. This technique can also be used for buried and near surface target detection, and the at-sea experimental results of bottom target detection will be presented somewhere else due to the page number restriction here.

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Structured Session 32

Underwater Observation Technologies

Organizers: Henrik Schmidt, Chris Malzone & John Potter
Autonomy Architecture for Planning and Prosecution of Multiple Mobile Assets in an Ocean Observatory

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Abstract: Underwater observation is experiencing a paradigm shift from platform-centric to distributed networks of gliders and autonomous underwater vehicles (AUVs). In order to achieve the performance goals in a wider area with smaller apertures carried by single platforms, new concepts of operations need to be developed, which make use of the mobility to adapt to the changing environmental conditions. Collaboration between mobile assets becomes the key when sampling the ocean with multiple assets. This paper presents an autonomy architecture developed at MIT, for multiple mobile assets in an observatory connected via a low-bandwidth acoustic communication network to carry out adaptive ocean sampling.
CHARACTERISATION OF A DIGITAL THIN LINE TOWED ARRAY – EXPERIMENTAL ASSESSMENT OF VIBRATION LEVELS AND TOW SHAPE

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Abstract: Development of light-weight and small diameter arrays has gained importance in recent years primarily because of their ability to operate from small and autonomous assets such as AUVs and USVs. Like conventional arrays these arrays are also subjected to platform vibrations, even though the levels could be small, which may tend to limit its acoustic performance. Similarly, knowledge of the hydrodynamic behaviour of the array, such as snaking of the array, is also important because such effects would result in undesirable performances during array beamforming and bearing estimation. Hence it is important to study these effects so that necessary compensations can be applied. For example, to suppress the vibration, suitable vibration isolation module can be built and if the array snakes, then it could be instrumented with suitable sensors for estimating the array shape. In this paper we describe an experimental approach to study the vibration levels experienced by a thin array under tow while simultaneously videoing the array for its hydrodynamic behaviour in a tow tank facility. The array was instrumented with three 3-axis accelerometers, one each at the array ends and one at the middle. The platform vibrations were monitored using another 3-axis accelerometer and the information was used to correlate the signals from the array accelerometers. Processing of data from the array accelerometer provided the information on how the disturbances were propagating along the array. The array was also equipped with acoustic ‘super-elements’ to measure the noise couple to the array at different tow speeds. The analysis of results showed that there is a low frequency component propagating along the array which increases with tow speed and decreases in amplitude from the tow end to the tail end for any given speed. It was also
observed that different low frequency components at certain tow speeds were propagating as if in a dispersive medium. The video recording showed that the array was being towed with minimal snaking and the measured values of drag showed much lower values compared to the computed ones.

**Keywords:** Digital thin line array, towed array, vibration isolation, tow shape

1. **INTRODUCTION**

Conventional towed arrays, which are hundreds of meters long and a few tens of mm in diameter, have been built to operate from ships and submarines for underwater sound detection and localisation of suspected objects. Autonomous Underwater Vehicles (AUVs) and Un-manned Surface Vehicles (USVs) are now being extensively used as underwater sensing platforms. There is a strong interest among the underwater research community to build scaled down versions of the conventional towed arrays so that they could be operated from autonomous assets such as AUVs and USVs [1-4]. The Acoustic Research Laboratory (ARL) under Tropical Marine Science Institute (TMSI) of National University of Singapore (NUS) has been engaged in building thin line towed arrays for the past 7 years and built the first proof of concept array in 2002 [5]. In 2007, ARL built the world’s smallest Digital Thin Line Array (DTLA) and was tested in the field using an AUV similar to REMUS100 class. The array was 10.5mm in (external) diameter and 12m long with 20m of tow cable. The DTLA with its associated software programmes was able to detect and estimate the bearing of acoustic transmitters located a few hundreds of meters away from it while being towed by the AUV [6].

Even though the DTLA concept has been tested out and the array performed its basic functions, such as detection and estimation of bearing of a suspected underwater object, there was no information available on how the array performance was being impacted by the platform vibration. For example, there was no special Vibration Isolation Module (VIM) used in the above DTLA and whether VIM is necessary at all was a question that remained to be answered. Through a comparison of the heading sensor outputs in the array and the AUV, it was estimated that the array remained horizontal during the tow. However, the heading sensor in the array was located very near the tow end and hence the results could not verify whether the array was subjected to snaking towards the tail end.

In the following sections we describe the approach, instrumentation and experimental procedures employed to study the vibrational levels experienced by the DTLA when towed at different speeds. The results and conclusions based on the analysis of data from the experiments have also been given. A video recording of the array, while under tow, was also performed to see whether there is any appreciable snaking on the array.

2. **DESCRIPTION OF DTLA INSTRUMENTATION**

The DTLA employed for this experiment was very similar to the one that was used in our 2007 experiments. However to measure the vibration levels along the array, three 3-axis
Accelerometers were added. The accelerometers were placed with one each at the tow-end, tail-end and the third one at the centre of the array.

The objective was to measure how an induced vibration at the array tow point was propagating through the array as it was being towed at different speeds. The array measured about 12m in length and 10.5mm in external diameter. There were 11 acoustic sensors, but only three were acquiring the data and were mounted next to the accelerometers. This arrangement provided a direct measurement of the level of coupling of the vibrations to the acoustic sensors through a comparison of their signal output with those of the accelerometers next to it. The nine other acoustic sensors, even though were dummy, were still retained in the array so as to duplicate as faithfully as possible the original array and to ensure similar weight distribution. The array configuration used is illustrated in figure 1.

The accelerometers used in the array were ADXL330, from Analog Devices. With no external capacitors connected to their output, these accelerometers have a frequency response of 1200 Hz along the X & Y axis and 600Hz along the Z-axis. The accelerometer can sense typically a full scale range of ± 3g accelerations and has a sensitivity of 300 mV/g. The unit can be used both for static (tilt) and dynamic applications. The accelerometers were aligned inside the array in such a way that their Y-axes were along the length of the array. The acoustic sensor used in the array was named as a ‘super-element’ and was formed by connecting six EDO micro-linear sensing elements in series. Each sensing element had a sensitivity of ~ -217 dB re 1V/µPa. The six elements connected in series provided ~ 15dB array gain and thus the overall sensitivity of the super-element was -202 dB re 1V/µPa. A preamplifier with ~60 dB gain provided a final sensitivity of -142 dB 1V/µPa per super-element.

3. EXPERIMENTAL SET UP AND PROCEDURES

The experimental set up employed is shown in figure 2. The experiment was performed in a high speed tow-tank facility which was about 500 m long, 8 m wide and 8 m deep. The...
array was attached to a tow post which was then towed using a tow-carriage. A 3-axis reference accelerometer, with a sensitivity of 100 mV/g, was attached to the tow post near the cable termination to measure the vibration levels induced at the cable end of the array. Figure 3 shows a snapshot of the array with carriage about to be towed. A Nexus preamplifier was used to amplify the signals from the reference tri-axial accelerometer and the data were recorded on to a PC using National Instruments NI6115 four channel data acquisition system. The receiver system for the DTLA was custom built using a STR-912 micro-controller. The SPI interface over 20 m of tow cable between the array and the receiver limited the clock frequency to a maximum of 2 MHz. Each array channel was sampled at 5 kHz rate and in this configuration there were 12 channels (9 accelerometer channels and 3 acoustic channels). Each channel was represented by 16 bits with the first four bits representing the channel ID followed by 12 bits of data. The data packets were sent over the Ethernet interface on to a laptop, where they were later converted to a text file to be processed using Matlab® software.

An underwater video-camera was positioned in such a way that it can capture the array dynamics during the tow. Later the video was played back to see whether the array was streamlined or was experiencing any snaking.

Experiments were conducted for various tow speeds from 2 to 10 knots in steps of 1 knot, even though most AUVs operate in the 4 to 5 knots speed regime. The drag on the array was also measured using a load cell attached to the tow cable near the termination point. Measurements were made after the platform and hence the array had attained a uniform velocity. Recordings were done only for the period when the array was in uniform motion.
4. DATA ANALYSIS AND RESULTS

The data from both the array accelerometers and the acoustic sensors were analysed and also compared with the data from the reference accelerometer for various tow speeds. The analysis was carried out to find out what components of vibrations were prominent and how do they change along the array and at different speeds.

The PSD of the accelerometer signals gave a measure of the frequency components of vibration experienced by the array and their relative amplitudes as they propagate along the array.

Preliminary observations showed that the PSD of the Y component of the accelerometer varied more than the X and Z components for different tow speeds. To get a better understanding, the variation in the PSD of Y components alone of the accelerometers for different speeds were plotted and is shown in figure 3. The energy increase in the accelerometer Y-component with increasing tow speeds is evident from the above plots. The plots also shows that the energy levels decreased from the tow end to the tail end of the array indicating that the disturbances originated from a point closer to the tow end and it was damped as it moved down the array.

The cross correlation of the output of the array accelerometers indicated that the array has some resonance at a tow speed of 4&6 knots. The results of cross correlation of the y-component of the accelerometers for selected tow speeds are given in figure 4. The plot at 10 knots indicates that there are multiple frequencies propagating along the array.

Figure 3 Variation in the PSD of the Y components of the array accelerometers for different tow speeds

Preliminary observations showed that the PSD of the Y component of the accelerometer varied more than the X and Z components for different tow speeds. To get a better understanding, the variation in the PSD of Y components alone of the accelerometers for different speeds were plotted and is shown in figure 3. The energy increase in the accelerometer Y-component with increasing tow speeds is evident from the above plots. The plots also shows that the energy levels decreased from the tow end to the tail end of the array indicating that the disturbances originated from a point closer to the tow end and it was damped as it moved down the array.

The cross correlation of the output of the array accelerometers indicated that the array has some resonance at a tow speed of 4&6 knots. The results of cross correlation of the y-component of the accelerometers for selected tow speeds are given in figure 4. The plot at 10 knots indicates that there are multiple frequencies propagating along the array.
To get a better understanding of the frequency components and the velocity with which they were propagating down the array, wave-number analysis was applied. The y-components of the three accelerometer outputs were summed after applying appropriate delays corresponding many different velocities of propagation. The PSD of the summed output was then plotted against the speed of propagation and for different tow speeds, and the results are shown in figure 5. The results showed that there are indeed some low frequency components propagating down the array with their speed increasing with increasing frequencies. The higher frequencies were found being attenuated at a faster rate compared to the low frequencies. The figure also shows that there is an increase in the velocity as the tow speed is increased this is consistent with our understanding that an increased tension would results in a higher velocity.

Efforts were made to see whether there is any correlation between the accelerometer reference data and the array accelerometer data. Even though a PSD plot of the signals

![Figure 4 Cross correlation output of array accelerometer Y component at various tow speeds](image)

*Figure 4 Cross correlation output of array accelerometer Y component at various tow speeds*
from the y-component of the reference accelerometer showed some increase in energy with tow speed around the same frequency band as those of the array accelerometers, a cross correlation of the array accelerometer and reference accelerometer components did not show much correlation. This is believed to be may be because of the non-linear or dispersive nature of the transmissions of the waves propagating along the array as brought out earlier.

The outputs of the ‘super-elements’ were analysed to see how do their acoustic outputs change for different tow speeds. Figure 6 shows the PSD plots for various tow speeds. An ambient noise recording, when the array was stationary, is also included in the plot. It is evident that the noise floor is going up when the array is being towed and increasing further with the tow speed. Both the array accelerometers and acoustic sensors were also picking up 50Hz noise and its odd harmonics as can be seen from the figures.

5. CONCLUSIONS

In this paper we have described an experimental set up to measure the vibrations and other acoustic disturbances that may arise when a thin line towed array is being towed at various speeds. Even though this may not be a substitute for the actual scenario where an AUV will be towing the array, this experiment has provided insight into the complex wave propagations along the array at various speeds.
1. There appears to be preferential excitation of certain frequency components depending
up on the speed as indicated by the cross correlation plots. For example at 10knots we
see that there are at least two main frequency components propagating along the array
2. The array output exhibits resonance near tow speeds of 4 and 6 knots as evidenced
from both the autocorrelation and cross correlation plots.
3. For any given speed the frequency of excitation is the same at all accelerometers, but
varies in amplitude. In general for a given tow speed the amplitude decreases as the
wave travels from the tow end to the tail except for speeds 4 and 6 knots where the
second accelerometer has a higher energy content probably due to the resonance and
snaking of the array
4. At 10 knots the velocity of propagation seems to be changing as a function of
frequency suggesting a non-linear or dispersive behaviour in the wave propagation.
5. The prominent frequencies of the propagating waves were found to be of very low
frequencies, with higher frequency components attenuated very fast. This suggest that
a VIM is probably required only if one is interested in measuring very low frequencies
and the design has to be catered to absorb the low frequencies.
6. The recorded video showed that the array was not subjected to large snaking during
the tow. A snap shot of the video recorded is given in figure 7.
7. The measured drag on the array was found to be much lower than the computed
values. For example the computed tension on the 12 m array being towed at 1&2 knots

Figure 7 Snapshots of array when under tow. Tow speed was 4
knots

are respectively 20 and 42 Newton. However the measured values during the
experiment were respectively 6.5&8.7 Newton. The tension measured at the max tow

speed of 10 knots was only 33 Newton and the array survived multiple runs at this tow speed.

A field trial employing an AUV platform is planned to be conducted in June 2009. This test would provide a more realistic scenario.

6. ACKNOWLEDGEMENTS

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REFERENCES


ACOUSTIC OBSERVATORIES OF THE AUSTRALIAN MARINE OBSERVING SYSTEM

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Abstract: Three underwater acoustic observatories will be deployed on Australia’s continental shelf in the Indian, Southern and Pacific Oceans in 2009 as part the Integrated Marine Observing System (IMOS) of the National Collaborative Research Infrastructure Strategy program. The observatories are a new national research facility, which will provide long-term passive underwater acoustic observations for studying abundance and migration of marine mammals in Australian waters and for investigating other natural sources of noise in the ocean, such as earthquakes, underwater volcanoes and ice breaking processes in Antarctica. Each observatory consists of four autonomous sea noise loggers deployed on the seafloor to form a triangular array of about 5-km sides with the fourth logger installed in the centre. The observatories can autonomously collect sea noise data over 8 to 12 months at useable frequencies up to 3 kHz, depending on the regime programmed for recording. After each cycle of autonomous recording, the observatories will be recovered for data retrieval and then redeployed for another recording cycle. It is planned to keep almost continuous data collection from the three observatories until at least 2011. Sea noise data from the IMOS acoustic observatories will be available for the scientific community worldwide.

Keywords: Underwater acoustic observatories, sea noise logger, marine mammals
1. INTRODUCTION

A network of three underwater acoustic observatories is planned to be deployed on Australia’s continental shelf in the Indian, Southern and Pacific Oceans as part the Integrated Marine Observing System (IMOS) [1] of the National Collaborative Research Infrastructure Strategy (NCRIS) program. One of the IMOS observatories has already been installed in the Perth Canyon area in Western Australia. The other two observatories will be deployed by mid 2009 on the continental shelf in Victoria and New South Wales. The proposed locations of the IMOS acoustic observatories are shown in Fig. 1.

The observatories are a new national research facility, which will provide long-term passive underwater acoustic observations for studying abundance and migration of marine mammals in Australia’s waters and for investigating other natural sources of noise in the ocean, such as earthquakes, underwater volcanoes and ice breaking processes in Antarctica. This new means for underwater acoustic measurements will support in particular long-term studies of sea ambient noise and marine mammals that have been conducted by the Centre for Marine Science and Technology (CMST) at Curtin University of Technology in Australia since the early 1990s. The design of the IMOS observatories is based on multi-year experience acquired by CMST in manufacturing and deploying autonomous sea noise loggers and analysing sea noise data. The CMST acoustic loggers have been collecting sea noise data from different areas all around Australia, which is used in particular for studying population and migration of marine mammals, for monitoring fish stocks and assessing their links with local productivity [2, 3]. The new observatories will continue such measurements on a regular basis. Figure 2 demonstrates an example of acoustic observation of pygmy blue whales around the Perth Canyon using sea noise loggers deployed over periods of several months in the recent years.
2. DESIGN AND DEPLOYMENT SCHEME OF ACOUSTIC OBSERVATORIES

Each observatory consists of four autonomous sea noise loggers deployed on the seafloor to form a triangular array of about 5-km sides with the fourth logger installed in the centre as shown in Fig. 3a. The mooring scheme of each logger is shown in Fig. 3b. Such a mooring design provides decoupling of the hydrophone with the cable and floats of the mooring system and, hence, reduces significantly noise from cable vibrations.
pinger of an acoustic release installed in the central mooring. The pinger signals are used for correcting different time drift rates in the quartz clocks of individual loggers. One of the IMOS sea noise loggers and its housing are shown in the photograph in Fig.4.

The accurate position of the hydrophones on the seafloor is measured after array deployment using the acoustic travel times from an impulsive source which is deployed near the sea surface and transmits signals from different locations above the hydrophone array measured by GPS.

The Perth Canyon IMOS acoustic observatory was deployed on 24 January 2009 at about 31°53′S and 115°00′E. It will be recovered for data retrieval after 2 October 2009 when the logger is programmed to stop recordings. The other two observatories will also be deployed for a period of about 8 months. After each cycle of autonomous recording, each observatory will be redeployed for another recording cycle. It is planned to keep nearly uninterrupted sea noise recordings at the three observatories until at least the end of 2011.

![IMOS sea noise logger with the hydrophone, its battery pack (alkaline batteries) and housing.](image)

3. MANAGEMENT AND COMMUNICATION OF IMOS ACOUSTIC DATA

According to the NCRIS and, in particular, IMOS rules, all data collected with the new research facilities funded within the IMOS project should be freely available for the Australian and international scientific community. All sea noise recordings (raw data) will be archived at the CMST. Because of the very large size of data files, the raw data will not be available online. A set of metadata will be created for each cycle of sea noise recordings at the three IMOS acoustic observatories. These metadata will contain information on the hydrophones’ position, start time and duration of all recordings, sampling rate, calibrations, etc. The metadata will be available from the eMarine Information Infrastructure (eMII) webpage [4] of the IMOS website and from the CMST website [5]. Raw acoustic data will be
provided upon request from potential users for certain time periods and location (observatory) selected by the users according to the metadata. Any additional processing of the raw sea noise data can be carried out at the CMST within joint projects or contracts with the interested institutions.

4. ACKNOWLEDGEMENTS

Manufacturing and deployment of the IMOS acoustic observatories, as well as collection and management of sea noise data obtained from the IMOS observatories are supported within the National Collaborative Research Infrastructure Strategy program conducted by the Australian Government.

REFERENCES

RECEIVING AND TRANSMITTING ACOUSTIC SYSTEMS FOR AUV/GLIDERS

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Abstract: NURC in the past years have focused its research on the use of AUV and more recently on gliders and is now developing littoral autonomous sensing networks based on those sensors to perform missions such as mine warfare, anti-submarine warfare, marine mammals risk mitigation and Rapid Environmental Assessment. For the different missions, new acoustic sensors, much smaller in size and having lower cost and lower power consumption are needed to be developed. The purpose of this paper is to present the work NURC started to do in that field since 2007. For example, in August 2007, NURC began to design and build a new thin High-frequency (up to 20 kHz) nested towed array (31 mm diameter) for ASW purposes. Array data analysis of at sea experiments will be shown to demonstrate that both AUV self noise as well as flow noise were not an issue whatever the tow speed and did not alter with the performance of the array. Based on those results, NURC decided in 2008 to investigate the use of thinner diameter arrays, very thin line arrays, which could be used with small size AUVs and/or gliders as with reduced weight and drag. A very thin line 12 mm diameter array prototype has been built in order to assess flow noise performance and also for exploring Linear Noise Power Averaging as a technique for suppressing flow noise on small diameter towed arrays. Then, the paper will describe the current development of a very high frequency (up to 160 kHz) tetrahedral array for a glider that will allow detection and accurate localization of marine mammals. Eventually, the paper will describe the current development and the first results of a towed transmitter that will be towed by a NURC AUV to perform both bottom characterization and ASW missions.

Keywords: AUV, glider, Acoustics, Transmitter, Receiver, towed array, tetrahedral array
1. INTRODUCTION

NURC in the past years has focused its research on the use of AUV and more recently on gliders and is now developing littoral autonomous sensing networks based on those sensors to perform missions such as mine warfare, anti-submarine warfare, marine mammals risk mitigation and Rapid Environmental Assessment. For the different missions, new acoustic sensors, much smaller in size and having lower cost and lower consumption are needed. The purpose of this paper is to present the work NURC started in that field in 2007. Both receiving and transmitting systems are described.

2. ACOUSTIC RECEIVING SYSTEMS FOR AUV/GLIDER

This section will present different systems already developed or under development at NURC for acoustic measurements to be performed from both AUV and glider. The first paragraph will described two different diameters arrays that have been developed for ASW applications while the third paragraph of this section will present a very small acoustic system under development for the detection and more importantly localization of marine mammals.

2.1. 31MM DIAMETER THIN LINE ARRAY (SLITA)

In August 2007, NURC began to design and build a new thin diameter (31 mm) High-frequency (up to 20 kHz) nested towed array for ASW purposes. An Engineering at-sea trial of the array towed by the 21” OEX-C Explorer AUV was performed beginning of November 07. The flow noise level of the array while towed and the potential influence of the AUV self noise on the acoustic array were also measured. Since that, a new 4 octaves 31mm array (called BENS) has been recently (beginning 2009) built at the Centre. This array includes as well 3 compass units.

2.1.1. Specifications

The SLIm Towed Array SLITA (see Fig.1) is a towed array derived from the SLIVA Vertical Line Array developed at NURC.

![Figure 1- SLITA Array deployment and OEX-C](image-url)
It features:

- 48 hydrophones in total
- 2 x 32 hydrophones
  - octaves spacing 0.211 and 0.422 m
  - array apertures are named Octave 2 (3550 Hz), and Octave 3 (1780 Hz)
  - cylindrical hydrophones (sensitivity -201 dB ref. to 1 Volt per µPa)
  - total gain 33.8 dB

The analog to digital acquisition board is a General Standards PCI-24DSI32 with 24 bits resolution. The used Benthos AQ-4 hydrophones (Sh=-201 dBV re 1 µPa ±1dB) are designed to compensate for noise generated from array movement, commonly known as acceleration noise. Noise is substantially reduced by symmetrically supporting the active element inside the mounting structure.

The mechanical design is shown below on Fig. 2.

![Figure 2 - SLITA array mechanical drawings](Image)

### 2.1.2. Performance evaluation at sea

Two trials of the SLITA array have been carried out during November 2007 and February 2008 in an area in front of the Palmaria Island, in around 30 m water depth.

**Mechanical noise considerations**

In order to differentiate between acoustic and non-acoustic noise, e.g., mechanical noise, one tool is provided by the wave-number-frequency diagrams (also called k-ω transform). These diagrams allow the identification of the speed of the pressure waves measured by the array.

Wave-number-frequency diagrams for real data do not show significant artifacts which could be produced by mechanical vibrations or array distortions. Figure 3 shows the k-f diagrams obtained during AUV navigation. The V-shape is well characterized, and the array gain is around 30 dB as expected. The residual energy outside the V-area is due to sidelobes effects (above roughly 3500 Hz and 1750 Hz for MF and LF aperture respectively).
**AUV propeller noise**

During the experiment carried on in February 2008, a tone generated by the AUV propeller can be seen when the AUV was on the surface on the SLITA array. The same noise however is not measurable while the AUV is underwater.

**Array shape**

An analysis of the array shape during a 90 degrees turn of the AUV was performed using acoustic means. It was observed that the array is coming straight behind the AUV after less than 80 seconds which is operationally very interesting.
2.2. 12MM DIAMETER VERY THIN LINE ARRAY (Micro-SLITA)

Based on the previous very promising results, NURC decided in 2008 to investigate the use of thinner diameter arrays, micro thin line arrays, which could be used with small size AUVs and/or gliders as with reduced weight and drag. A very thin line 12 mm diameter experimental array has been built in order to assess flow noise performance and also for exploring Linear Noise Power Averaging as a technique for suppressing flow noise on small diameter towed arrays.

2.2.1. Specifications

The prototype built is made of 20 hydrophones and has one depth sensor and one tilt sensor. The array total length is 14 meters. The array has an interface to the SLITA data acquisition system. The micro-SLITA has been designed to be towed from the OEX-C explorer.

![Diagram of Micro-SLITA Array](image)

*Figure 5 – Very thin 12mm diameter Line Array (Micro-SLITA)*

2.2.2. First at-sea results

The trials consisted of towing from the R/V Leonardo the micro-SLITA at various speeds from 3-7 knots along straight paths approximately parallel to the Italian coast adjacent to Tellaro, just SE of NURC. By design unfortunately, the towing cable length was not long enough to get rid of the R/V Leonardo noise. This was a limitation that is shown later in the data analysis results but this trial served as a first initial test for this array in water.

There were a number of issues that limited the quality of the acoustic data from the micro-SLITA. Briefly, these were:

- As the tows progressed, there were more drop-outs, spikes and other signal breakdowns in the hydrophone channels, most likely due to mechanical stressing of the array under tow. This was particularly noticeable on higher-speed tows.
• The R/V Leonardo generates substantial mechanical noise below 6 kHz, making analysis of acoustic data for other sources in this frequency band impossible.

• At 14 kHz and above there are substantial quasi-tonals, apparently generated electronically by the R/V Leonardo. The lack of phase lag along the array suggests that these are injected into the hydrophone data stream via electromagnetic pickup rather than by an acoustic propagation path.

Fig. 6 shows the characteristic R/V Leonardo mechanical acoustic noise prevalent to 6 kHz, the 9 kHz pinger event (about 1/3 of the way through this record) and the 14 kHz R/V Leonardo electromagnetic tonal. Towards the latter part of the spectrogram one of the episodic vortex shedding events can be seen, interpreted as R/V Leonardo hull wake turbulence.

![Figure 6. Broadside beamformed acoustic data from run 4.](image)

The acoustic data were bandpass filtered with a Butterworth IIR filter, passing 6 kHz - 13.5 kHz with a 1 dB passband ripple and -20 dB stop band ripple, the filter being passed twice through the data (once forward and once in reverse) to obtain a zero time-shifted output. Time-frequency transients were then removed by a bi-orthogonal Debauchies QMF wavelet decomposition, thresholded filtering and re-synthesis to suppress the impact of noisy transients on the subsequent processing. Having finally obtained an acoustic dataset cleaned of R/V Leonardo mechanical and electromagnetic noise and time-frequency transients, and excluding faulty channels, it is possible to examine the cross-correlation between channels to explore if there are turbulence signals that are advected along the array at or near the tow speed. For each separation from the minimum available (5 mm) to the maximum (70 mm) all possible pairs of hydrophones were cross-correlated and the average normalised amplitude plotted as a function of temporal and spatial separation. An example output is shown in Fig. 7.
The high cross-correlation coefficients at near zero time lag are due to the propagating acoustic energy received by the hydrophones. That there are negligible cross-correlation amplitudes at lags outside the narrow acoustic propagation angles in space-time indicates that there is no discernible passive turbulence signal in this acoustic data.

As a summary of the data analysis, the following things were observed:

- While the artefacts due to the mechanical and electromagnetic noises from R/V Leonardo are significant, they are not the only sources of noise in the acoustic data.
- The tilt and pressure sensors worked well, and the data shows much of interest that can be understood in terms of the towing environment. Most of the acoustic energy observed in these data is in some way generated by the R/V Leonardo.
- No apparent turbulence energy is found in the frequency range 6-13.5 kHz. Below this band the acoustic energy is dominated by R/V Leonardo mechanical noise. Above this band the hydrophone data is dominated by R/V Leonardo electromagnetic noise.

An other trial with the micro-SLITA towed from the OEX-C Explorer will be conducted in June 2009.
2.3. TETRAHEDRAL ARRAY FOR MARINE MAMMALS “TAMM” DETECTION AND LOCALIZATION

2.3.1. Background

It is now widely accepted that sound-generating anthropogenic activities may have a negative impact on marine life. This impact will not only depend on the type of sound, but will also depend on the species and their behavioural activity. Expected hazards range from temporal behavioural disruption, over permanent displacement to potential fatal stranding.

To mitigate the risk of anthropogenic sound, it is critical not only to uncover the causal effects of the sound and to establish criteria for the onset of negative effects, but also to develop risk mitigation methods aimed to minimize sound exposure. An important approach for the development of mitigation measures is to develop the capability to monitor for the presence of marine mammals in an area prior to, and during anthropogenic activity.

One approach for reducing the risk of exposing cetaceans to unacceptable sound levels is to avoid areas with their presence. This strategy would require build up of confidence that, with a given probability, no cetacean is within the area that is impacted by a certain sound pressure level from the sound source. For this it is important to be able to quantify the probability of cetacean presence in a given area. Acoustic detection of cetacean sounds is a viable way to detect their presence also at depth and may be an efficient complement to visual sightings of surfacing animals. When integrated in autonomous systems like buoys/gliders passive acoustic is an efficient alternative to expensive ship based systems. The range over which whale sound can be heard is usually greater than the range they can be detected during surfacing, and acoustic detection remain effective at night and, especially using autonomous systems, in poor weather.

2.3.2. Specifications

The TAMM (Tetrahedral Array for Marine Mammals) is a self-contained underwater sound recording and detecting device capable of acquiring wide bandwidth sound continuously, processing the sound, and then storing extracts of sound to non-volatile memory. The device can be used as a stand-alone recorder or with an external GPS and radio telemetry as part of a monitoring installation.

The data collected by the TAMM can be offloaded, after the device is recovered, via USB to a personal computer. The internal battery in the TAMM is recharged at the same time (either from the USB connector at a low rate or from an external power supply at the full rate). The amount of time required to offload and recharge the device will depend on use but should be less than 5 hours (charging with an external power supply).

The system under design and development will have 4 hydrophone inputs sampled at
500 kHz 18 bit and will use home-made low noise pre-amplifiers. Digital compasses, pressure sensors and temperature sensors will be also included in the TAMM system. A TAMM comprises 2 circuit boards. The main board contains a digital signal processor (DSP), memory, power supply, and interface circuits. The sensor board contains audio acquisition circuits and depth and (possibly) orientation sensors.

The design will leverage on the recent development of the NURC CPAM (Compact Passive Acoustic Monitor) as shown on the figure below:

![Figure 8 – Compact Passive Acoustic Monitor (CPAM) tow body and acoustic channel electronic noise](image)

This system is equipped with four hydrophones positioned in tetrahedral mode and with low pre-amplifier ($< 2 \text{nV/\sqrt{Hz}}$ in the 30-60 KHz bandwidth) and can be deployed at a depth up to 750 meters and towed up to 5 knots speed. New A/D technology will be used within the TAMM to decrease the noise floor in the upper band of the frequency band of interest.

3. ACOUSTIC TRANSMITTING SYSTEM FOR AUV

In addition to the previously described acoustic receiving systems, NURC has decided in late 2008 to develop as well a transmitting sound source to be towed by AUV for both bottom characterization and ASW missions.

For the bottom characterization mission, the objective is to characterize the geoacoustics properties (sound speed, density, attenuation and stratification) of the clutter features as well as the seabed surrounding the clutter features. Both characterizations are needed for sonar performance modelling. The broadband spherical reflection coefficient contains significant information about the seabed geoacoustic properties at spatial scales relevant to the clutter problem. The method uses an AUV with a towed source and horizontal array, and a moored vertical array as a receiver along previously conducted tracks using a boomer source. The boomer source runs are considered as “ground truth” to the AUV runs. The method requires CTD and XBT measurements to properly characterize the water column. The system described below will be used at-sea in May 2009.
3.1. DESIGN SPECIFICATIONS

The TOwed Sound Source for Auv (TOSSA) is described on Fig. 9. The transmitter bandwidth covers from 800 Hz to 3400 Hz and is using Ultra MPS transducers. The required bandwidth is achieved in two separate bands using both the MPS 6-35 and the MPS 2-100. The achieved Source levels are given on the figure below.

![Figure 9 – Acoustic transducers characteristics](image)

Greater Sound Pressure Level (SPL) (5 to 10 dB more) will be easily achieved soon in changing the amplifier modules.

The power amplifier container is described below:

![Figure 10 – Acoustic transducers amplifier](image)
The mechanical design is shown on Fig. 11.

![Figure 11 – Towed Sound Source AUV (TOSSA)](image)

The following figure shows the tow body that has been specifically designed for implementing the transducers and providing the best towing capability:

![Figure 12 – OEX-C AUV+ TOSSA](image)

First at-sea experiments of this system were conducted beginning of April. It was shown that the OEX-C explorer was able to tow the Source at around 2 knots. The at-sea behaviour of the AUV was not affected at all by the towed transmitter. During the experiment, one of the 31mm towed array (BENS) described above were towed also from the AUV while attached to the transmitter.
Fully monostatic sonar operation on an AUV was fully demonstrated as shown by the beams figure below. An echo repeater was used to simulate a submarine and a clear detection was seen on the processing display.

![Image of beams figure](image.png)

**Figure 13 – OEX-C AUV+ Sound source + echo repeater: beam display from towed array**

4. CONCLUSION

This paper has described very promising new acoustic equipments developed or under development at NURC for both Autonomous underwater vehicles and gliders. The results obtained so far are very good and open new ways for performing bottom characterization anti-submarine warfare and marine mammals risk mitigation.

5. ACKNOWLEDGEMENTS

The authors would like to thank the Leonardo crew for their dedication to the successful test of all the described equipments. A special thanks to Marco Mazzi and Stefano Biagini, without whom, any test with the OEX-C explorer could not have been made with such a great success.

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AUTONOMOUS INITIAL CAPTURE SYSTEM FOR AUV RECOVERY

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Abstract:
As Autonomous Underwater Vehicle (AUV) technology matures and applications become more widespread, it is becoming clear that AUVs will be deployed in fully autonomous scenarios (that is, without direct human intervention at any point in the mission, including launch and recovery) and in collaborative ‘swarms’. With multiple AUVs in the water at the same time, or for fully-autonomous operation, the efficient, reliable autonomous launch and recovery of AUVs becomes an as-yet unsolved challenge. There are several areas in which current launch and recovery techniques need to be advanced to meet this challenge, including all-weather operation, recovery in reduced visibility (e.g. at night) and automation of the recovery.

The next generation of automated recovery systems will need to provide a reliable method to achieve first contact between the support platform and the AUV to support the subsequent recovery, recharging, data exchange, etc. This is perhaps the hardest step. We propose a generic capture device that would provide this initial contact for a wide range of AUVs, including those without homing or Ultra-Short Base Line (USBL) capabilities.

One of the key features of the design is to minimise assumptions and requirements on the AUV so that the system has wide application to a range of AUVs, including simple AUVs that might be used in ‘swarms’ and recovery by surface ships, Unmanned Surface Vessels (USV), bottom-mounted data/power hubs, etc. It is assumed only that the AUV can be programmed to proceed from a predetermined ‘Start’ waypoint to an ‘End’ waypoint along a ‘recovery track’ at a nominally-constant depth and speed, ignoring any internal collision avoidance warnings.

This leads into another novel feature; the homing is performed by a mobile Tethered Initial Contact Hoop (TICH) rather than the AUV. This design choice in turn demands that there should be a mobile and a ‘fixed’ part of the system; in our case connected by an umbilical tether cable. The Tethered Initial Contact System (TICS) then consists of a wet-end mobile TIC Hoop (TICH) coupled via an umbilical to a dry-end (though this could be sub-surface) signal processing TIC Intelligence Unit (TICIU). The TICS additionally requires two small self-contained transponders to be attached to the AUV, perhaps housed on a jacket, collar or internal to a flooded AUV fairing.
The TICH is equipped with an interrogating transducer and three acoustic receivers (forming an USBL for the AUV-mounted transponders), depth, heading and IMS sensors. Data from these sensors are fed from the TICH to the TICIU over an ethernet backbone via the umbilical. Control data are, in return, supplied by the TICIU, housed on the support platform. The TICH has thrusters so that it can move autonomously within the range of its umbilical tether, following TICIU commands. TICH power may be on-board (using high energy-density batteries) or supplied via the umbilical.

While the TICH is concerned only with its position relative to the AUV, an optional transponder on the TICIU would permit the position of the TICH relative to the TICIU to be estimated, which may prove useful in limiting attempted motion at the limits of the umbilical reach.

The TICH will initially be deployed close to the anticipated rendezvous point (towards the end of the recovery track), manoeuvring itself to the planned intercept point while seeking reliable acoustic contact with the AUV transponders. Once the AUV is detected and its bearing estimate has stabilised, the TICH will switch to a predictive pursuit mode of homing, positioning itself in an anticipated capture pose given the known nominal recovery track and estimated AUV bearing, allowing for AUV navigational errors and water currents. As the AUV closes and the signal to noise ratio improves, near-field USBL range estimation will become available for final fine-tuning of the homing motion. The TICH will be fitted with a mechanical guidance cone of flexible staves, entering into a tube, in which a grappling mechanism will physically engage the AUV for capture.

**Keywords:** Autonomous Underwater Vehicle Recovery Homing Navigation Acoustic Transponder USBL AUV LARS

1. INTRODUCTION

With interest in exploiting marine resources moving into ever deeper water and distributed sensing networks coming to the fore, the Autonomous Underwater Vehicle (AUV) with diverse and modular sensing payloads will play an increasing role in the exploration, detection, classification and tracking of marine phenomena. In addition, the prospect of using many AUVs simultaneously in a ‘swarm’ is gaining momentum. As AUVs become commonplace and with the expectations of exploiting collaborative behaviour, the technical demands of efficient handling of AUVs is brought to the fore, both in terms of the launch and recovery and also in operations like data download, mission upload, and power recharging to extend the autonomous endurance of the AUV. In addition, there is considerable interest in being able to operate AUVs from unmanned surface vessels (USVs), in high sea states and at night.

AUVs, with limited endurance, need to rely on a support platform to provide power and data exchange between missions. Given the desire to be able to recover AUVs in all weathers, day and night, the natural way forward is to recover the AUV sub-surface. For an USV to provide this support, there is, in addition, a need for the entire operation to be automated.
2. CURRENT RECOVERY STRATEGIES FOR ‘TORPEDO-TYPE’ AUVS

Generally, a ‘torpedo-type’ AUV (defined here as having small frontal area compared to its length whose control requires external surfaces deflecting the water flow around the vehicle while underway) is recovered by either coming to a standstill at the surface and/or by ejecting a capture line from the nose. In the former case (applicable to smaller AUVs) the AUV is generally recovered by personnelle operating from a small boat close to the waterline (such as from a Rigid Hull Inflatable) and in the latter the AUV may be recovered by snagging the recovery line with a grappling hook and drawing the AUV onto some type of recovery sled lowered from the support platform to the water surface. Cocoons are often used for AUVs with payloads that cannot directly handle ramp contact, intended to sheath and protect the AUV during the recovery process [3].

In all these cases, human intervention is required for the initial contact and the operation is unsafe and/or unreliable during high sea state and in limited visibility, such as at night, during rainstorms, etc.

Some AUVs like the Remus 100 series, and more recently a 21” Bluefin, have demonstrated the capability to navigate and home into a fixed bottom mounted docking station using an onboard USBL system homing into a static hoop [1,2]. This requires that the AUV has an USBL receiver array, sufficiently responsive motion control to home effectively and software to implement homing signal processing and real-time ‘externally-determined’ control. While some AUVs can home into dedicated stationary docking stations, the majority do not posses all of the required hardware and software, or perhaps even the manoeuvrability, required. This technique is also not directly applicable to recovery by surface ships. In addition, with heterogeneous ‘swarms’ of co-operating autonomous assets, it is likely that more than one type of AUV will be deployed at once.

For a generic baseline AUV we permit ourselves only to expect that the AUV can be programmed to proceed at a specified depth and speed from a ‘Start’ waypoint along a ‘recovery track’ towards an ‘End’ waypoint and then to stop. If the AUV has a collision avoidance system, this would obviously need to be disabled during the recovery phase. This project addresses the need to autonomously handle AUVs of this nature in a generic fashion.

3. TETHERED INITIAL CONTACT SYSTEM (TICS)

The first step in defining the design constraints for our problem is in recognising that the homing and manoeuvrability requirement to have a generic AUV intercept a support-platform must be transferred from the AUV to the recovery system. This means that the recovery system must not only implement the homing (USBL array and signal processing) but also have at least some movable element, to react to homing estimates and close the control loop. Since the support platform is assumed essentially fixed (or at least unlikely to be able to move with sufficient agility to adapt to capturing the AUV in surge and currents), this implies that the recovery system must consist of at least two parts, one mobile and agile, the other ‘fixed’ to the support platform. We choose to connect these two components by an umbilical cable that supports an ethernet backbone data communication link.

The proposed approach automates the first physical contact with a returning AUV. A Tethered Initial Contact (TIC) capture device that we call a ‘Hoop’ (TICH) provides the
agile mobile element of the recovery system. The TIC Intelligence Unit (TICIU) takes the sensor data from the TICH, computes the thruster activation for the TICH to home onto the approaching AUV and, optionally, provides power to the TICH. The TIC System (TICS) then consists of the mobile TICH and fixed TICIU, connected by an umbilical cable, with transponders mounted on the AUV and possibly also the support platform.

The separation of the system into two main parts, necessitated by the homing and agile mobility requirements, has a collateral advantage. For recovery from surface platforms, the TICH can be allowed to descend typically 2-20 metres below the sea surface, away from surface wave action, greatly reducing the impact of wind and wave on the motion of the AUV and initial contact component of the recovery system. This greatly reduces the all-weather problem experienced by surface vessels.

1. Functionality

The basis of the homing control is an acoustic estimation of the AUV position and orientation by means of an Ultra Short Base Line (USBL) array, formed by three receiving hydrophones arranged round the outer edge of the receiving funnel of the TICH. This requires that two self-contained transponders, operating at slightly different frequencies, be attached to the AUV. The transponders may be housed in an external jacket or collar, attached to the outer hull or incorporated internally in a flooded design.

The geometry and positions of the transponders is not critical, providing they generate as large an aperture as possible and that their positions with respect to the leading point of the AUV are input to the TICIU software as parameters. The transponder aperture is used in the closing stages of the interception, when the AUV is in the near field of the USBL, to estimate the heading orientation of the AUV independently of its track to improve estimates of drift due to water currents. Fig. 1 shows a cartoon of possible AUV transponder arrangements that might be used. Note that the ‘glider-like’ AUV on the right would have to be a hybrid to satisfy our requirement of proceeding between waypoints at constant depth.

In addition, an optional transponder may be attached to the support platform to provide platform-centric navigation information on the TICH position. This is useful both for navigating the TICH to the nominal intercept position and for managing manoeuvrability at the limits of the umbilical tether.

![Fig. 1. Schematic of possible AUV transponder arrangements](image-url)
The AUV mission will end with instructions to move first towards a specific ‘Start’ waypoint and from there to proceed on a constant heading, thrust and depth to a designated ‘End’ waypoint some 500 m distant, then coming to a halt (at which point it would normally float to the surface if not captured). Depending on the sophistication of the AUV navigation system, it may or may not be instructed to adjust its heading to remain on the nominal track between Start and End waypoints. To improve navigational accuracy and establish pre-recovery contact, the AUV may surface at the Start waypoint prior to starting its recovery run. The Start and End waypoints would normally be chosen at the beginning of the mission (before the AUV is launched if no contact is possible afterwards), aligned with anticipated water currents at the planned time of recovery. The depth of the recovery track would be chosen to insulate the AUV and TICH from surface wind and wave action and to remain clear of the support platform’s own draft. If the support platform were bottom-mounted, the arrangement would be inverted, with the TICH now above the TICIU at sufficient altitude above the bottom and TICIU to minimise collision risk.

The TICH would be launched from the support platform adjacent to the recovery track, near the ‘End’ waypoint. The TICH would be controlled by the TICIU on board the support platform in the following four stages:

1. Descend/ascend to nominal intercept depth and proceed to the intended recovery intercept point on the recovery track.
2. Maintain position while orienting the USBL perpendicular to the anticipated arrival track of the AUV and interrogating the transponders to establish acoustic contact.
3. Once reliable acoustic contact has been made and the AUV range and bearing estimates are steady, estimate the AUV actual track deviation from the intended recovery track (by the temporal derivative of the azimuthal and elevation bearings) and proceed to the updated estimated intercept point. Repeat as necessary.
4. Once the AUV is in the near field of the USBL, use the independent range and bearing information from the two AUV transponders to improve AUV heading versus track estimates and refine homing motion to intercept.

Fig. 2: TICS Components
The TICH would feed the TICIU control software with USBL acoustic data, TICH heading and depth. The TICIU then estimates the state space parameters and computes the control input for the thrusters to position the TICH. By keeping the TICH sensors simple and the computational load on the TICIU, the size and inertia of the TICH can be kept to a minimum, improving manoeuvrability.

2. TICS Components

The Tethered Initial Capture System comprises the Tethered Initial Capture Hoop (which is immersed and mobile), the Tethered Initial Capture Intelligence Unit (which may be above the surface) the ethernet umbilical connecting these two modules and the COTS transponders fitted to the AUV. These components are depicted in Fig. 2.

5. TICS ARCHITECTURE

1. Tethered Initial Contact Hoop (TICH)

The TICH is the physical hoop that will navigate to intercept the AUV; an umbilical cable connects the TICH to the TICIU. The TICH will have a transponder interrogator to trigger the AUV transponders. The rim of the capture guidance hoop will have a set of three receiving hydrophones spaced at 120 degrees. Fig. 3 shows a schematic of the hydrophone geometry on the rim of the TICH and the interrogation of transponder 2 by the TICH. Here \( \delta t_i \) is the time delay to propagate from the transponder on the AUV to receiver hydrophone \( i \).

The hydrophone data will be pre-processed and sent to the TICIU on the Ethernet backplane via the umbilical tether.

The TICH's navigation unit will have a solid-state compass, depth sensor (pressure) and possibly an inexpensive IMU. The outputs from the navigation unit will also be sent over ethernet to the TICIU. The TICIU will then estimate the bearing (and at closer range the range) of the AUV and generate control outputs for the TICH thrusters. The TICH's physical dimensions will be designed to accommodate the Nose and a section of the hull of the AUV as it approaches the TICH. Obviously, larger guidance hoops and docking tubes will be required for larger AUVs, but in principle these could be interchangeable.
TICH modules, selected for the mission underway. A sketch of the TICH layout is shown in Fig. 4.

2. **Tethered Initial Contact Intelligence Unit (TICIU)**

The TICIU accepts inputs from the TICH sensors and controls the thrusters on the TICH. If a support platform transponder is deployed the TICIU can also calculate the distance and bearing of the TICH from the support vessel. The TICIU can then use this information to manage the range of operation of the TICH based on the available length of the umbilical tether. The TICH thrusters provide for heave, sway, surge and yaw degrees of freedom. The block diagram in Fig. 5 outlines the main steps taken by the TICIU to navigate the TICH to intercept the AUV.

![Fig 5: TICIU Control Flow](image)

The TICIU will execute the interception process in two main phases, the ‘Positioning’ phase and the ‘Homing’ phase. In the ‘Positioning’ phase, the TICIU will navigate the TICH towards the nominal intercept position. This position is on the line connecting the Start and End waypoints at the proscribed depth. The co-ordinates of the TICH at launch are known to the TICIU from GPS data on the support platform. At this point the control system of the TICIU operates on the support platform transponder data (if available) combined with the data from the compass and IMU in the navigation unit on the TICH.
After reaching the nominal intercept position the TICIU will shift to the second ‘Homing’ phase. In this phase the TICIU controls the TICH in response to the positioning data from the interrogation of the transponders on the AUV by the TICH. This homing phase is itself composed of an initial long-range traditional pure pursuit algorithm, followed by an anticipatory intercept homing sequence as the signal to noise ratio improves and the AUV moves into the near field of the TICH. Pure pursuit is a special case (unity gain) of the well-known proportional navigation algorithm [4]. In the final stages of approach, higher gains may be appropriate to have the TICH lead the AUV estimated track to account more effectively for cross-drift. The TICIU will also need to keep track of the acoustic channel conditions to dynamically adapt the interrogation interval and track error estimates on the bearing and range. The state-space estimator uses these error estimates, the position returned from the signal detector and the data from the navigation unit of the TICH. This data is used to orient the TICH towards the approaching AUV.

3. AUV transponders

The AUV will be fitted with two COTS transponders which may be jacketed to provide a hydrodynamic profile to the AUV. As the transponders will only operate when the transducer on the TICH interrogates them they are not expected to interfere with the normal operation of the AUV during its mission.

6. CONCLUSION

This project outlines a design solution to the generic problem of autonomously capturing a returning AUV by a support platform, assuming only minimal capability on behalf of the AUV. The subsequent docking and recovery is not addressed, there being a number of established solutions to this part of the recovery problem. The conceptual solution can be applied to both surface and bottom-mounted support platforms, including both manned and unmanned surface vehicles. The solution is valid for zero visibility and high sea state conditions, since it relies on acoustic rather than visual homing and the initial contact can be arranged to occur well below the sea surface. The next planned phase of this project is to develop a TICS system for trials with small torpedo-type AUVs such as the REMUS 100 series.

7. REFERENCES

Numerical Modelling of sound velocity profile in different layers in the Persian Gulf

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Abstract: The three-dimensional variability of sound speed in the Persian Gulf is investigated. In this study, a three-dimensional hydrodynamic model (COHERENS) is employed in a fully prognostic mode to derive sound speed profiles in the Persian Gulf, an evaporation-driven inverse estuary that is governed by the import of surface water from the adjacent ocean and the export of saline bottom gulf water through the Strait of Hormuz. During spring and summer, a cyclonic overturning circulation establishes along the full length of the Gulf. During autumn and winter, this circulation breaks up into mesoscale eddies, laterally stirring most of the Gulf’s surface waters. Results of the model show that sound speed in the Persian Gulf depends mainly on the temperature in the surface layer whereas the bottom layer as well as the southern part of the Gulf depends on temperature and salinity.

Maximum sound speed occurs during the summer in the Persian Gulf which decreases gradually moving from the Strait of Hormuz to the north western part of the Gulf. A gradual decrease in sound speed profiles with depth was commonly observed in almost all parts of the Gulf. However, an exception occurred in the Strait of Hormuz during the winter. The results of the model are in very good agreement with previous observations.

Keywords: Persian gulf, Numerical modelling, Sound velocity, Strait of Hormuz.
1. Introduction

Because of its natural resources, the Persian Gulf is an important military, economic and political region and is one of the world’s busiest waterways. Countries bordering the Persian Gulf are Iraq, Kuwait, Saudi Arabia, Qatar Bahrain, the United Arab Emirates and Oman, on one side and Iran on the other side (Fig. 1). The Persian Gulf is ~990 km long and has a maximum width of 370 km. The average depth of the Gulf is 36 m. The Persian Gulf occupies a surface area of ~239,000 km² (Emery 1956). Extensive shallow regions, with depths below 20 meters, are found along the coast of the United Arab Emirates, around Bahrain, and at the head of the Gulf. Deeper portions, with depths over 40 meters, are found along the Iranian coast continuing into the Strait of Hormuz, which has a width of ~56 km and connects the Persian Gulf to the northern part of the Indian Ocean via the Gulf of Oman. The narrow Strait of Hormuz restricts water exchange between the Persian Gulf and the northern part of the Indian Ocean.

The Persian Gulf is a semi-enclosed, marginal sea that is exposed to an arid, sub-tropical climate. It is located between latitudes 24°-30°N, and is surrounded by most of the Earth’s deserts.

![Fig.1: Bathymetry of the Persian Gulf.](image-url)

The Gulf experiences evaporation rates of ~2 m/y (Privett 1959, Hastenrath and Lamb 1979, Meshal and Hassan 1986, Johns and Olson 2003) that exceed by far the net freshwater input by precipitation (~0.15 m/yr) and river discharge (Johns and Olson 2003). The major river source of the Persian Gulf is the Shatt-Al-Arab (called Arvand Roud by some countries), being located at the head of the Gulf and being fed by the Euphrates, Tigris and Karun rivers. The salinity distribution in the Persian Gulf experiences significant seasonal variations. The inflow of Indian Ocean seawater strengthens in late spring and summer and moves further up the Iranian coastline and closer to the coast of the United Arab Emirates (Reynolds 1993). As a result of this, surface Gulf waters are saltier in the winter than in the summer. Theoretical and experimental studies on sound transmission have been carried out for different sea and ocean conditions. So far different equations with some limitations have been
given by scientists. These equations slightly differ from each other due to some limitations. The sound speed in the ocean is a function of depth, salinity and temperature.

Sound waves propagate much better in the ocean than in the air. The average speed of sound in the air with a temperature of 20 degrees centigrade is 334 meters per second but the average sound speed in the sea is about 1500 meters per second.

Usually, in the sea sound speed is increases with arise in temperature, depth and salinity. Due to large seasonal variation in the temperature and salinity in the Persian Gulf, the study of sound speed profiles seems to be of great interest to oceanographers. Brewer and Dyrssen (1985) and Emery (1956) observed winter and summer hydrographic conditions respectively in the Persian Gulf. This data was scattered over certain areas, whereas the observations of RV Mt. Mitchell (Reynolds 1993), provided an intensive and comprehensive data set for the entire Gulf. As far as sound speed in the Persian Gulf is concerned little has been published with regard to its acoustic properties. Previous attempts have utilized temperature data only and neglected the effects of salinity. These results have been used to investigate seasonal variations in the acoustic properties of the Persian Gulf. However, the data was scattered, giving neither acoustic information for the northern part of the Gulf nor the vertical variations across and along its entirety.

This study focuses on the distribution of sound speed profiles in the Persian Gulf by employing a three dimensional hydrodynamic model, COHERENS (Luyten et. al 1999) which has not been addressed before. Characteristics of seasonal changes of sound speed have been investigated using the combined effect of temperature and salinity, in order to compare their relative importance.

1. Materials and Methods

We employed the hydrodynamic part of COHERENS [COupled Hydrodynamical Ecological model for REgioNal Shelf seas] (Luyten et. al 1999), which is based on a vertical sigma coordinate. The model is run in a fully prognostic mode with Cartesian lateral coordinates on the f-plane, using geographical latitude of 27°N. The model is based on hydrostatic versions of the Navier-Stokes equations that embrace conservation equations for momentum, volume, heat and salt. We employ 10 sigma levels and Cartesian lateral grid spacings of $\Delta x = 7.4$ km (east-west direction) and $\Delta y = 6.6$ km (north-south direction). Bathymetry and coastline locations are based on ETOPO-2 data that has been interpolated and slightly smoothed onto a 4-minute grid (see Fig. 1). Minimum water depth is chosen at 5 m and maximum water depth is restricted to 150 m. The latter applies only to the Gulf of Oman and has no significant impact on the results. Note that restriction to 10 sigma levels is a compromise to maximize model efficiency with relatively fine lateral resolution and long simulation times (~20 years).

The model is initialized in winter when vertical stratification is weak throughout the Gulf using uniform temperature and salinity fields with values of 20°C and 38PSU, respectively, which are reasonably close to observational evidence (see, Alessi et al., 1999). The model is forced by climatologic monthly mean atmospheric forces (wind speed, air temperature, humidity, cloud cover and precipitation) derived from 54 years (1952-2006) of NOAA data. River discharge is implemented in the model by means of inflow of a low-salinity (10 psu) surface layer of 1.5m in thickness and 700m in width. Riverine inflow is assumed to vary in a sinusoidal curve with minimum values of 350m$^3$/s in October and a maximum of 650m$^3$/s in April. This gives an annual-mean river discharge of 500m$^3$/s (15.8km$^3$/yr), which we deem a realistic estimate of current discharge rates. River temperatures are assumed to vary between minimum value of 16°C in December and a maximum value of 32°C in July.
At the open-ocean boundary we prescribe 2-layer profiles of temperature and salinity, derived from previous numerical modelling in the Persian Gulf (Kaempf and Sadrinasab, 2005) on a monthly basis. Temperature and salinity do only vary significantly in the upper 60m of the water column. The water column underneath does not experience significant temporal variations and is kept at a temperature of 22°C and a salinity of 36.5 psu throughout the simulations. Amplitudes and phases of the four major tidal constituents, $M_2$, $S_2$, $O_1$, and $K_1$, are prescribed as constant values along the eastern open-ocean boundary in the model. In order to simulate sound speed, Lorans equation is employed and added to the model to calculate sound speed in the domain as a function of temperature, salinity, depth and latitude, that is:

$$c(x, s, t) = 1449.05 + 45.7t - 5.21t^2 + 0.23t^3$$
$$+ (1.333 - 0.126t + 0.009t^2) \times (s - 35) + \Delta (x)$$
$$\Delta (x) = 16.3x + 0.1$$

As the Lorans equation is defined for latitude 45 degrees, we replace $x$ with $x(1 - 0.0026 \cos \varphi)$ in the mentioned equation.

In this equation $s$, $t$, $\varphi$ and $x$ denote salinity in PSU, temperature in Celsius degree, latitude in degree and depth in kilometres.

2. Results and Discussion

In order to simulate seawater properties as well as circulation of the Gulf, the model was run for 15 years and has reached its steady-state. Our findings, which are in good agreement with observational evidence, suggest that the Persian Gulf experiences a distinct seasonal cycle in which a gulf-wide cyclonic overturning circulation establishes in summer, but this disintegrates into mesoscale eddies in winter. In other words, water from the Indian Ocean enters from the surface into the Persian Gulf via the Strait of Hormuz during summer time and reaches to the head of the Gulf, but during the winter the Indian Ocean water reaches only near the middle of the Gulf. Moreover, top to bottom temperature gradients exceed 11°C during the summer and diminish to 1 to 2 degrees in the winter. As a result of this a seasonal thermocline establishes during the summer which is in a very close agreement with Emery (1956). The model also predicts the vertical mixing of the water in winter in the entire Gulf. It is due to the passing of cold air over the surface of the Gulf, resulting in an increase of density in the surface layer of the water and creating a vertical movement from top to bottom. Figure 2 shows last 4 years of the simulated Gulf-averaged salinity, temperature and sound speed which attain a robust, steady seasonal cycle. Gulf-averaged temperature follows the seasonal cycle of incident solar radiation with a time lag of 1-2 months. Gulf-averaged salinity, on the other hand, attains minimum values during March-May each year. Effects of precipitation and river run-off on salinity changes are negligible on a gulf-wide scale. Decreases in salinity can be fully attributed to inflow of Indian Ocean surface water which is in agreement with observational evidence, peaks in spring. Maximum salinities occur during October - December when the evaporative surface salinity flux dominates over injection of low-salinity water through the Strait of Hormuz.
Also analyses of sound speed in different seasons suggest that, sound speed in the Persian Gulf mainly depends on temperature variations which are clearly illustrated in figure 2. As mentioned earlier maximum water temperature as well as the maximum difference between surface and bottom temperatures occurs during the summer. Hence maximum sound speed occurs during summer in the Persian Gulf.
From the model results we have constructed sound speed profiles in surface and bottom layers of the whole domain in winter and summer. During the summer due to lack of vertical mixing maximum stratification of temperature occurs in this region. Figure 3 shows sound speed in the surface and bottom of the Persian Gulf during summer. As can be seen from this figure, sound speed is maximum during the summer (~1560 m/s) in the southern part of the Persian Gulf. Also sound speed in the northern part of the Gulf varies from 1524 m/s to 1528 m/s in the bottom and from 1547 m/s to 1552 m/s in the surface. These show that sound speed has a 20 m/s difference from surface to bottom during the summer.

![Sound speed profile](image1)

Fig.3: Sound speed (from 1500 m/s) in surface and bottom layers during summer in the Persian Gulf.

In the southern part of the Gulf except for the region surrounding Bahrain, sound speed varies from 1548 m/s to 1552 m/s in the bottom layer and from 1553 m/s to 1557 m/s in the surface layer. This location is shallow and the difference between surface and bottom temperature is negligible and sound speed is a function of both temperature and salinity. Findings of the model also suggested that around Bahrain, sound speed varies from 1539 m/s to 1544 m/s in the bottom and from 1554 m/s to 1558 m/s in the surface. In this part of the Gulf, sound speed varies more than in the other parts because this region is shallow and salinity has more effects on sound speed than temperature. Figure 4a and 4b respectively show vertical profiles of sound speed constructed from the model output and field observation in the Strait of Hormuz during the summer by Abdelraman (1998).
Figure 4a and 4b show that the sound speed profile in the Strait of Hormuz decreases from surface to bottom i.e. varies from 1546 m/s to 1528 m/s during the summer. It is seen that the correspondence between model and observations is very good in the deeper layers. The model produces slightly higher sound speed at the surface and does not place the strongest gradients in the uppermost layer, but the general structure is reproduced quite well.

Figure 5a and 5b respectively show vertical profiles of sound speed constructed from the model output and field observation in the Strait of Hormuz during the winter by Abdelraman (1998).

As can be seen from the figures variation of sound speed is very small during winter at the Strait of Hormuz.
Moreover sound speed in the Strait of Hormuz has negligible changes during the winter. These figures also show that sound speed in the Strait of Hormuz increases from surface to bottom in the winter, which is the only exception in the Persian Gulf. Comparing figures 4a with 4b and also 5a with 5b shows that the results of the model are in very good agreement with the observations.

![Fig. 5.a: Vertical profile of sound speed (from 1500 m/s) from the model results in the Strait of Hormuz during winter.](image)

![Fig. 5.b: Vertical profile of sound speed (from 1500 m/s) from field measurement in the Strait of Hormuz during winter 1996 (Abdelraman 1998).](image)

### 3. Conclusion

1. Sound speed is a function of temperature, salinity and depth in the Persian Gulf but the effect of salinity and temperature is more than the effect of depth.
2. Because the Persian Gulf is shallow, depth can not affect sound speed and is negligible.
3. The temperature is the main factor that affects sound speed, so we can conclude that sound speed is a strong function of temperature in the Persian Gulf.
4. Sound speed in the Persian Gulf has higher values during the summer than the winter because of higher water temperature.
5. From the results of this model we can simulate the sound wave propagation in the Persian Gulf. This is of great importance in underwater communication applications.
6. Changes in the seasonal pattern of sound-speed, particularly in the Strait of Hormuz are evident.

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5. References


PRODUCTION AND LOW FREQUENCY CALIBRATION OF A SMALL SPHERICAL HYDROPHONE

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Abstract: In this article, production and low frequency calibration of a small spherical hydrophone was described. The manufactured hydrophone is omnidirectional in both vertical and horizontal directions for all frequencies up to its specified limits which were shown by directivity patterns.

In order to measure the sensitivity of the hydrophone, the vibrating column of fluid method in low frequency (frequency range 50-110 Hz) was deployed. The designed instruments are described and experimental results are analyzed. To measure the accuracy of the method, the B&K hydrophone type 8105 was considered as a reference and the sensitivity of the B&K hydrophone type 8104 was measured with the accuracy of 0.6 dB. Considering the complexity of low frequency hydrophone calibration, the method has satisfactory accuracy compared to other low frequency calibration methods.

Using this method, the sensitivity of the manufactured hydrophone was measured to be -201 dB re 1V/µpa.

Keyword: spherical hydrophone, calibration, hydrophone sensitivity, piezoelectric
1- INTRODUCTION
Electroacoustic Transducer is a device that converts electrical energy to acoustic energy and vice versa. A hydrophone is a transducer which is used as a receiver and is capable of converting underwater sound waves into electrical signals.
To achieve high sensitivity and wide frequency band, it is important to choose the material characteristics and physical dimensions wisely. The electrical amplifier of the hydrophone should also have a low noise level. Typically, hydrophones have a small size, because:
1-An omni-directional hydrophone in a specific frequency range should have smaller dimensions compared to the acoustic wavelength in water in the highest frequency of the mentioned frequency range. 2-To achieve a flat frequency response in a frequency range, the first natural frequency of the hydrophone should be higher than the frequency range.
Piezoceramic sensors of different material, shapes and dimensions are used in hydrophone production [1].
Hydrophone sensitivity measurement has an important role in analyzing the output data in underwater acoustic applications such as environmental noise measurements or seismic exploration. Therefore, calibration of hydrophones in different frequency bands is one of the most important and substantial goals of an underwater acoustic laboratory. Different methods have been introduced for measuring hydrophone sensitivity in a water tank in frequencies between 2 to 20 kHz in which the tank dimensions are proportional to the acoustic wavelength of the transmitter. As a result, these methods are not applicable in frequency bands bellow 1 kHz. Considering the nature of the wave propagation in small, laboratory environments, creating an environment which ensures the repeatability of the tests is a substantial acoustic problem. The problem becomes more complex in low frequencies because of the formation of standing waves. In fact, the first aim of calibration is to create a uniform pressure field in which the measurement results are independent of the hydrophone location [2].
In this study PTZ spherical piezoceramic sensors were used to produce a spherical hydrophone. Then, the sensitivity of the produced hydrophone was measured in the frequency range of 50-110 Hz using vibrating column of liquid method.

2-PRODUCTION METHOD
In order to produce a spherical hydrophone [3], a spherical piezoceramic sensor was used (Fig 1a). Table 1 shows the specifications of the applied piezoceramic sensor, where d is spout diameter of sphere, t: thickness, φ: outer diameter of sphere, f: frequency, D: dielectric loss, C, capacitance and Z is electrical Impedance.

<table>
<thead>
<tr>
<th>Material</th>
<th>Dimensions</th>
<th>f=1(kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PZT-5A</td>
<td>D (mm)</td>
<td>T (mm)</td>
</tr>
<tr>
<td></td>
<td>6.0</td>
<td>1.5</td>
</tr>
</tbody>
</table>

Table1: piezoceramic sensor's specification

2-1-COATING OF THE SPHERICAL HYDROPHONE
PU was used to coat the sensors after the assembly of hydrophones. The applied PU is a bicomponent polymer which solidifies in room temperature and has acceptable flexibility in low temperatures down to -60°C. The polymer has proper adherence to a verity of material
such as metals, ceramics, cement, rubber and wood. Due to low sensitivity to humidity and suitable flexibility, this polymer is typically used for thin layer coatings. Coating was performed by casting in this project as the coating material was selected to be PU. However, considering the complexity of the shape of sensors and the assembled element and the difficulty of manufacturing the suitable mold for casting, a creative method was developed to coat the assembled element [4]. Since the coating is in direct contact with the sensor, formation of the air bubbles should be prevented. Air bubble formation can cause inaccuracies in the sensor's performance. Using the new coating method, air bubble formation can be prevented to a great extent. In the mentioned method coating was performed layer by layer. During the coating in each layer (especially in the first layer) air bubbles were omitted. The coating was applied in different layers until the coating thickness reached 2.5 mm. The assembled element is shown after coating in figure 1b.

![Fig.1a: Employed piezoceramic sensor](image1a)

![Fig.1b: The assembled element](image1b)

Finally the coated sensor was successfully tested in 20 bars pressure to make sure that the produced hydrophone has enough mechanical strength in working condition.

### 3- LOW FREQUENCY CALIBRATION METHOD

The method consists of immersing a hydrophone in a column of liquid which is excited externally by a sinusoidal vibration, while the hydrophone is held fixed and vertically suspended near the central axis of the column. Assuming that the entire body of liquid participates uniformly in the motion and every particle departs by the same amount \( x \) from its mean position, the instantaneous departure \( p \) of the pressure from its mean at depth \( h \) is given by

\[
p = \rho g x + \rho h \ddot{x},
\]

where \( \ddot{x} \) is the acceleration of the liquid and \( g \) is the acceleration of gravity. [5] If the oscillation is sinusoidal, at angular frequency \( \omega \), then the pressure amplitude will be

\[
p_0 = \rho x_0 (g - \omega^2 h).
\]

The first term is associated with the oscillating static head and the second with the alternating acceleration.

When a hydrophone is placed in the liquid, the uniformity of the motion is disturbed. The pressure is perturbed from the value that would exist in the absence of the hydrophone by the same amount the instantaneous perturbation \( p' \) of the pressure at a point on the surface of the hydrophone can be written as

\[
p' = \rho c_1 \dddot{x} + \rho c_2 \ddot{x},
\]

and the total pressure would be

\[
p = p_0 + p',
\]

where \( c_1 \) and \( c_2 \) are pressure coefficients for steady flow and for accelerated flow, respectively. The value of these coefficients varies with position on the hydrophone surface and also depends on the shape. [6] Values of these coefficients are available for simple
shapes, but these values are not well known for the specific shape of most hydrophones. However, for a sphere with radius $a$, $p'$ would be negligible if $h \gg x \gg a$ \hspace{1cm} (5)

The measurements were accomplished using a B&K vibration exciter type 4809, a PVC cylinder with the 15cm internal diameter and 50 cm length. The deployed fluid was Silicon oil 50 with the density of 959kg/m$^3$. The height of the fluid column was 42.3cm and the hydrophone was fixed in the depth of 37.3 cm.

First the sensitivity of a standard 8105 hydrophone with the diameter of 22mm was measured, then the sensitivity of a standard 8104 hydrophone with the diameter of 21mm was measured using the standard 8105 as a reference and later the test was repeated for the produced spherical hydrophone (which has the diameter of 24mm) [3]. The whole system was vibrated with the harmonic acceleration of 0.1g which was measured by a standard B&K piezoelectric accelerometer type 4931. The hydrophone output signal was measured using a DS-6121A Iwatsu Digital Storage-scope.

4-DISCUSSION AND CONCLUSION

4-1- ELECTRONIC TESTS

To measure the electronic characteristics of the produced hydrophone, an Impedance Analyzer was deployed. The electrical impedance, capacitance, dielectric loss, and the first resonance frequency of the produced hydrophone were measured in 1 kHz. Table 3 shows some electronic characteristics of the produced hydrophone with 10 meters B&K cable.

<table>
<thead>
<tr>
<th>$Z_0$($\Omega$)</th>
<th>$f_0$(kHz)</th>
<th>f=1kHz</th>
</tr>
</thead>
<tbody>
<tr>
<td>55.6</td>
<td>81.0</td>
<td>1.7</td>
</tr>
</tbody>
</table>

Table3: Electronic characteristics of the produced hydrophone

4-2-SENSITIVITY MEASUREMENT AND DIRECTIVITY OF THE PRODUCED HYDROPHONE IN HIGH FREQUENCY

The produced hydrophone was tested in an acoustic tank using B&K standard equipment to measure the sensitivity and other acoustic characteristics of the prototype. Frequency response of the hydrophone was measured in the frequency range of 2.1 to 32.1 kHz. A transducer was used as the transmitter and the frequency response of the produced hydrophone was measured using a standard B&K hydrophone type 8105 as a reference. figure 4 shows the frequency response of the produced hydrophone compared to a standard B&K hydrophone type 8105 in the before mentioned frequency.

Fig.4: Frequency response of B&K hydrophone type 8105 and the produced hydrophone
To study the receiving sensitivity of the produced hydrophone in all angles in the X-Y and X-Z planes in different frequencies, the hydrophone was tested in an acoustic tank using B&K equipment. A transducer was used as the transmitter and the produced hydrophone was tested as the receiver.

Figure 5 shows the polar pattern of the produced hydrophone in the vertical (X-Z) plane in different frequencies.

Figure 5a, 5b shows that by increasing the frequency, the polar pattern loses its omnidirectionality in some areas. This phenomenon is caused by the inconsistencies in the structure of the hydrophone. These variations in the vertical polar pattern are completely similar to those of the standard B&K hydrophone type 8105 which has a similar structure to the produced hydrophone [3].

Figure 5c shows the polar pattern of the produced hydrophone in the horizontal (X-Y) plane in different frequencies. As can be observed in the figure 5c, the receiving sensitivity of the produced hydrophone in X-Y plane is omnidirectional in all angles even in relatively high frequencies.

### 4.3-SENSITIVITY MEASUREMENT IN LOW FREQUENCY

The dimensions of the tested hydrophones satisfy the equation (5), therefore, the perturbation can be neglected and the pressure amplitude on the hydrophone surface can be calculated using equation (2). The measured sensitivities of the hydrophones and the accuracy of the sensitivity measurements are given in table 4.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>Hydrophone Sensitivity (dB re 1V/μPa)</th>
<th>The absolute accuracy of the sensitivity measurements</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>8105 (Reference Hydrophone)</td>
<td>8104</td>
</tr>
<tr>
<td>---------------</td>
<td>-------------------------------</td>
<td>------</td>
</tr>
<tr>
<td>50</td>
<td>-202.0465</td>
<td>-202.80</td>
</tr>
<tr>
<td>60</td>
<td>-201.5199</td>
<td>-202.75</td>
</tr>
<tr>
<td>70</td>
<td>-201.3512</td>
<td>-202.60</td>
</tr>
<tr>
<td>80</td>
<td>-200.5538</td>
<td>-201.95</td>
</tr>
<tr>
<td>90</td>
<td>-200.5538</td>
<td>-201.95</td>
</tr>
<tr>
<td>100</td>
<td>-200.1083</td>
<td>-201.6</td>
</tr>
<tr>
<td>110</td>
<td>-199.8938</td>
<td>-201.43</td>
</tr>
</tbody>
</table>

*Table 4: The measured sensitivity of the 8105, 8104 and the produced hydrophone*
Figure 7 shows the sensitivities of the 8104, 8105 and the produced hydrophone in the frequency range of 50 to 110 Hz.

![Hydrophone Sensitivity Graph](image)

Fig 7: The measured sensitivities of the 8104, 8105 and the produced hydrophone

### 4-4 CONCLUSION

The employed piezoceramic sensor is Lead Zirconate Titanate spherical in shape. PU was deployed to coat the piezoceramic sensor. The element was coated layer by layer. The resulted coat can endure pressures up to 20 bars. Receiving sensitivity of the produced hydrophone is uniform in X-Y plane in all angles. By increasing the frequency, the vertical polar pattern loses its omnidirectionality.

The sensitivity of the produced hydrophone was measured in low frequencies using vibrating column of liquid.

The B&K hydrophone type 8105 was considered as a reference and the sensitivity of the B&K hydrophone type 8104 and the produced hydrophone was measured with the accuracy of 0.5 dB. Using this method, the sensitivity of the small hydrophones which are deployed in different underwater acoustic applications can be measured. Considering the complexity of low frequency hydrophone calibration, the method has satisfactory accuracy compared to other low frequency calibration methods.

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Underwater communication, navigation and positioning

Organizer: Paul Van Walree
EXPERIMENTAL RESULTS OF THE EFFECT OF THE NUMBER OF RECEIVING CHANNELS IN WIDEBAND ACOUSTIC COMMUNICATIONS

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Abstract: With using developed wideband transducer, the sea experiment of high-speed underwater acoustic communication was carried out at near the bottom of 1,000 m in depth. The modulation method is quadrature phase shift keying (QPSK). As the results, it was proved that the communication was possible at the distance of 600 m and the data rate of 80 kbit/s. Furthermore, the data were recorded in every channel of four elements. So, the modulator performance was estimated with using different channel number. Then, in case of single channel, the receiver performance has deterioration of 3 dB comparing with symbol error rate of additive white Gaussian noise (AWGN) channel. When used 2 channels for demodulation, its performance was improved 3 dB and SER has become same as AWGN channel case. Then, when the using channels for demodulation to 4 channels, its performance was increased another 3 dB. In that case, error free communication was carried out with average input signal-to-noise ratio of 9 dB. And also, error free communication was carried out up to 600 m of slant range, when the sound pressure level was 172 dB and 4-channel demodulation.

Keywords: wideband, acoustic communications, QPSK
1. INTRODUCTION

There are various researches on the underwater acoustic communication.\textsuperscript{1-3} Authors have also been researching the underwater acoustic communications, especially for short range wideband acoustic communication at deep sea.\textsuperscript{4,5} In this paper, “wideband” is defined as the bandwidth of over one third of its carrier frequency. There are several papers which report the wideband acoustic communication for short range\textsuperscript{6-8}.

The authors have been investigating high speed acoustic communication for image transmission at deep sea. In order to control remotely operated vehicle (ROV) by acoustic signal, high update rate for image transmission is required. The goal is to develop a colour image transmission system, which can transmit more than 1 colour image per second. For this objective, the research is in progress based on sea experiment. As the sound speed in the sea water is approximately 1,500 m/s, the travel time of sound for the round trip of 500 m is approximately 2/3 s. It is assumed that the operator will accept the transmission delay of less than 1 second in round trip. Because of this operation response, the transmission range should be limited to approximately 500 m. In consideration of this limitation and absorption loss\textsuperscript{9}, the carrier frequency was set to 80 kHz. The wideband transducer with tilted toroidal beam was developed in 2006.\textsuperscript{10} It has 40 kHz bandwidth, which is 50 % of 80 kHz carrier frequency.

2. OUTLINE OF THE EXPERIMENT

The experiment was carried out at Suruga Bay on the 2nd of November, 2007. Fig. 1 shows the outline of the system used for the sea experiment. Water depth is approximately 1,000 m. The transmitter was moored near the sea bottom and it was located at approximately 6 m above the bottom. The receiver was suspended from R/V “KAIYO” by armored coaxial
cable. Direction and slant range between the transmitter and the receiver were controlled by changing the ship position and the length of cable. Wideband acoustic communication was carried out with using 80 kHz carrier frequency, 40 kHz bandwidth and quadrature phase shift keying (QPSK) as the modulation method.

In order to realize such a wideband communication, the wideband transducer with tilted toroidal beam\(^{10}\) was used. Fig. 2 shows directivity of transmitting voltage responses (TVR). Beam pattern of three frequencies (the centre frequency of 80 kHz and edge frequencies of usable band, 60 and 100 kHz) can be seen in this Fig. 2. Main beam is directed to 70° ±10° from vertical. This transducer uses free-flooded rings effect\(^{11}\) to widen its bandwidth.

The block diagram of the transmitter and the receiver are shown in Fig. 3. The transmitter is composed of single board computer (SBC), digital-to-analog converter (DAC), power amplifier, transducer, lithium-ion secondary battery pack and timer unit. Transmitting data is a compressed colour image in JPEG format stored in a hard disk drive (HDD) as a binary file, and pre-processed to modulate to pass-band signal. After mooring, the transmitter is activated by trigger pulses from the timer unit. Sampling frequency of DAC is 800 kHz that is 10 times over-sampling of the carrier frequency. The receiver is composed of SBC, analog-to-digital converter (ADC), band pass filters (BPF), pre-amplifiers, hydrophones, lithium-ion secondary battery pack and timer unit. At the experiment, the receiver was activated by trigger pulse from the timer unit that was synchronous to the timer unit of the transmitter. Four omni-directional hydrophones were used for receiving the acoustic signals. In accordance with number of the hydrophones, 4 units each of the pre-amplifiers and BPFs

![Block diagram of the transmitter and the receiver.](image)

were equipped. Sampling frequency of ADC was 800 kHz same as that of DAC. Each signal
from the four channels was passed through the band pass filter, amplified and digitized in 12 bit. Finally, digitized data of 4 channels data were stored in the HDD. After recovering the receiver, received data stored in the HDD were demodulated by software. Transmitted data was colour image, 320×225 pixels, compressed in JPEG format and 33,840 symbols of QPSK modulated data amount. The software demodulator is constructed by adaptive multichannel decision feedback equalizer (M-DFE) with phase compensator. The least mean square (LMS) algorithm is adopted as the update algorithm of the tap coefficients.

3. RESULTS AND DISCUSSIONS

The sea experiment was carried out and its outline is shown in Fig. 1. The receiver position was shifted along with the line of 77º from vertical by controlling the ship position and the cable length. Relative positions of the receiver are shown in Fig. 4. The x-y origin (0, 0) means the position of the transmitter, which is calibrated from the position measured by acoustic navigation system of R/V KAIYO. During this experiment, the depth of the receiver measured by CTD sensor was almost same as cable length, less than 0.5 m difference. So, it is assumed that the receiver was suspended just under the point of A-frame crane at the stern. The receiver position was calculated by offsetting the difference between GPS position and suspended position. Because all the recorded data were measured along the same line of 77º from vertical, there is no need for considering directivity pattern of the transducer. Slant ranges, which were obtained from the communication data, are approximately 410, 470, 560 and 620 m.

In these channels, QPSK communication data were obtained by changing the slant range along the line of 77º from vertical. In this experiment, the training sequence is composed of 1,024 symbol pseudo random sequence.

In this sea experiment, the total number of transmitted packet was 20. Fig. 5 shows the performance of the demodulator. Characteristics are shown when the number of the channels in use for the demodulation was changed from 1 to 4. The x-axis shows average input SNR and the y-axis shows symbol error rate (SER). In case of error free, its result is plotted as $10^{-5}$ of symbol error rate. The solid line is the calculated value with additive white Gaussian noise (AWGN) channel. The result of a single-channel demodulation was approximately 3 dB worse than AWGN case. The result of 2 channels demodulation was almost the same as AWGN case. The result of 4 channels demodulation was about 3 dB better than AWGN case. When the number of channels becomes larger, the result of demodulation performance

![Fig. 4 Relative position of the receiver while the experiment. (0, 0) as the transmitter position. Triangle mark: relative position of the receiver. Solid line: 77º from vertical.](image-url)
becomes better. 4 channels DFE demodulator shows the 6 dB better performances than single channel DFE demodulator.

In this experiment, the transmitter and the receiver to obtain communication data are located on the same line of the 77° from vertical. So, we can compare its performance according to just distance between the transmitter and the receiver. Fig. 6 shows the SER-slant range characteristics. X-axis is the slant range of transmission and Y-axis is SER. Single

![Graph showing symbol error rate vs average input SNR. Transmission rate: 80 kbps.]

**Fig. 5** Symbol error rate vs average input SNR. Transmission rate: 80 kbps.

![Graph showing symbol error rate vs slant range.]

**Fig. 6** Symbol error rate vs slant range.
channel DFE demodulator makes many errors at the distance longer than 470 m. But 4 channels DFE demodulator performs error free communication up to 570 m.

4. CONCLUSION

Wideband short range underwater acoustic data transmission experiment was carried out at 1,000 m depth area. Propagated data were recorded along the same direction, and these were recorded for each 4 channel respectively. By changing the number of the channels in use for the demodulation, the characteristics were obtained. As the number of the channels is increased, the demodulation performance is also increased. When average input SNR is larger than 10 dB, the result of 4 channels demodulation was approximately 6 dB better than single channel case. And it was confirmed that the error free communication can be done at the slant range of 560 m with 4 channels DFE demodulator. Now the authors are planning to develop the wideband receiver to be used under high pressure and are expecting that it will show a good performance in the deep sea operation in the near future.

REFERENCES

ROBUST ACOUSTIC COMMUNICATIONS OVER DELAY-LIMITED UNDERWATER CHANNELS

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Abstract: Two coded modulation schemes with the same bit-rate and decoding complexity are designed and compared. The first scheme combines Trellis Coded Modulation (TCM), symbol interleaving, and orthogonal frequency-division multiplexing (OFDM). The second system combines bit-interleaved coded modulation (BICM), based on a convolutional code, with OFDM. Both systems are combined with a space-time block code (STBC) when three projectors are used at the transmitter. Testing the aforementioned systems using both simulated and experimental data during the Rescheduled Acoustic Communications Experiment (RACE’08), the following result was obtained: coded modulation schemes emphasizing higher Hamming distance (e.g., BICM) yield a lower bit-error-rate (BER) when there is no option for many transmit and/or receive hydrophones. On the other hand, coded modulation schemes emphasizing higher free Euclidean distance (e.g., TCM) obtain a lower BER when many transmit and/or receive hydrophones are available.

Keywords: underwater acoustic communications, diversity, coded modulation, TCM, BICM, space-time codes, STBC, OFDM, MIMO.

1. INTRODUCTION

The aim of this work is to design a low-complexity, high-data-rate acoustic modem with robust performance under various channel conditions. A robust approach assumes a coded modulation scheme that provides good performance in both the additive white Gaussian noise (AWGN) and Rayleigh fading channels (and, consequently in Rice fading channels, which is an intermediate channel model between the other two). Consequently, coded modulation schemes should exhibit both large free Euclidean distance (minimum Euclidean distance between any pair of codewords) and Hamming distance (number of bits/symbols for which
Two codewords differ) [1]. Two methods are used to apply this approach. The conventional method, which induces no data rate penalty, is to employ TCM [2]. The innovative method for UWA communications employs bit-interleaved coded modulation (BICM) [3].

Improved link reliability can be accomplished by exploiting the time, frequency, and space diversity inherent in UWA channels [4]. Toward this end, two systems are designed and compared. The first system combines TCM based on an 8-PSK signal set, symbol interleaving, and orthogonal frequency-division multiplexing (OFDM). The second system combines BICM, based on a convolutional code and a 16-QAM signal set, with OFDM. Both systems are combined with a space-time block code (STBC) when three transmit projectors are used. Testing the proposed systems using both simulated and experimental data from RACE'08, the following result was obtained: the BICM scheme achieves a lower BER when the UWA channel exhibits a low spatial diversity order. Thus, coded modulation schemes emphasizing larger Hamming distance should be preferred. The TCM scheme, on the other hand, becomes a better choice when the UWA channel demonstrates a high spatial diversity order. In that case, coded modulation schemes emphasizing larger free Euclidean distance should be preferred.

2. SYSTEM DESIGN

Figure 1 (a) shows the block diagram of the transmitter of the proposed two single-projector systems. The proposed systems allow us to transmit different parts of a codeword across different coherent bands and coherent periods of the channel, thus exploiting both time and frequency diversity. The incoming bit stream is encoded by either TCM or BICM. The TCM scheme is based on a 2/3-rate, 8-state TCM encoder using an 8-PSK signal set followed by a symbol interleave. The BICM scheme combines a 1/2-rate, 16-state convolutional encoder, a bit interleaver, and a 16-QAM signal set based on Gray labelling. Note that both encoders achieve the same data throughput, 2 bits/sec/Hz, and the same trellis complexity, 32. The resulting coded symbols are grouped into vectors of \( N_c \) symbols each, where \( N_c \) is the number of sub-carriers. The symbols of each vector are simultaneously transmitted over the different \( N_c \) sub-carryers via IFFT operations. A cyclic prefix (CP), equal to the channel delay spread is inserted between successive vector transmissions in order to eliminate interference from previous vector transmissions caused by multipath propagation. Figure 1 (b) illustrates the receiver block diagram. After CP removal, FFT demodulation, and channel estimation, appropriate symbol/bit metrics are computed and deinterleaved before they are fed to the TCM/BICM decoder. Then, maximum likelihood (ML) decoding is performed by using the Viterbi algorithm.

![Block diagram of the proposed TCM-OFDM / BICM-OFDM system](image)

**Figure 1**: Block diagram of the proposed TCM-OFDM / BICM-OFDM system: (a) transmitter; (b) receiver.
In Figure 2 the TCM-OFDM and BICM-OFDM systems are extended so that three projectors are incorporated to exploit transmit diversity. In particular, TCM/BICM is considered as an outer code concatenated with an inner rate-3/4 STBC [5]. This approach provides increased reliability because full space and frequency diversity is achieved. In addition, receiver complexity is kept low since the STBC decoder retains a maximum likelihood decoder based on linear combining of the received signals. Detailed description of both the single- and multiple-projector systems is available in [7].

![Figure 2: Block diagram of the proposed TCM-STBC-OFDM / BICM-STBC-OFDM system.](image)

3. SIMULATION RESULTS

In all the following simulations, the bandwidth is $W = 4$ kHz, the number of sub-carriers is 128, and the cyclic prefix duration is 25 msec. For every system, information bits are grouped into packets of 30 OFDM symbols before transmission. The channel symbols are drawn from unit energy constellations, and when $N_r$ receive hydrophones are used, the transmit energy is divided by $N_r$ to compensate for the array gain. The physical path delays are spaced at integer multiples of $W^{-1}$; thus the number of physical paths is equal to the number of non-zero channel taps. The average power of the link between the transmitter and any receive hydrophone is one, regardless of the multipath structure. Consequently, the average received SNR for any system-channel combination is $1/N_0$, where $N_0$ is the average power of the additive white complex Gaussian noise process. In addition, we assume that the receiver has perfect knowledge of the values of each channel tap. Finally, when $N_r$ receive hydrophones are employed, the TCM-OFDM and BICM-OFDM systems are denoted as TCM1x $N_r$ and BICM1x $N_r$, respectively.

Figure 3 (a) compares the BER performances of TCM1x1 and BICM1x1 over Rayleigh block-fading channels with different delay spreads. The coded bits/symbols are interleaved within the packet with interleaving depth of 4 sub-carriers. When the channel has two physical paths, both TCM1x1 and BICM1x1 attain the same BER slope; however, TCM1x1 performs slightly better because it achieves a higher coding gain (4.59 as opposed to 3.2 for the BICM1x1). For all other channels, BICM1x1 performs better due to its higher coding diversity (8 as opposed to 2 for the TCM1x1).

Figure 3 (b) illustrates the BER performances of TCM1x1 and BICM1x1 over Rayleigh fading channels with different delay and Doppler spreads. Each physical path is characterized by a uniform Doppler power spectrum with maximum one-sided frequency $f_d$. To keep the system complexity low, intercarrier interference (ICI) is not compensated at the receiver. When the channel exhibits 1 msec delay spread, the BER performance of both systems is improved as the Doppler spread increases from 0 to 5 Hz, validating that both time and frequency diversity are exploited.

When two and six receive hydrophones are employed, each physical path is modelled with zero Doppler spread. The receiver performs maximum ratio combining (MRC) to achieve full receive diversity. Figure 3 (c) compares the BER performances between BICM1x2 and TCM1x2. For 1 msec delay spread, both systems achieve full frequency and space diversity; however, TCM1x2 performs better due to its higher coding gain. As the delay spread
increases from 1 msec to 10 msec, the available frequency diversity also increases, and thus BICM1x2 performs better. In Figure 3 (d), we observe the following: when six receive hydrophones are used, the frequency selective fading channel turns into an AWGN-like channel due to the ability of both systems to average out the channel fade by exploiting full space diversity. This is proved by the fact that the BER performances of both systems approach their corresponding performances for the AWGN channel as receive diversity increases. Consequently, TCM1x6 outperforms BICM1x6 since the system with the larger minimum Euclidean distance along any error event will always perform better.

![Figure 3: BER performances for TCM-OFDM and BICM-OFDM over various frequency-selective Rayleigh fading channels: (a) zero Doppler spread, one receive sensor; (b) various Doppler spreads, one receive sensor; (c) zero Doppler spread, two receive sensors; (d) zero Doppler spread, six receive sensors.](image)

4. EXPERIMENTAL RESULTS

The results of this section are derived by processing experimental data recorded during the Rescheduled Acoustic Communications Experiment (RACE) in Narragansett Bay, Rhode Island, in March 2008. The transmitter consisted of a three-sensor line array with 60 cm inter-
sensor spacing and a separate primary transducer, which was 1 m above the uppermost element of the line array and 4 m above the sea bottom. The depth at the transmitter site was 9 m. At 1 km range from the transmitter, the receiver was a 12-sensor line array with 12 cm inter-sensor spacing. The depth at the receiver location was approximately 10 m. The total emitted signal level was 185 dB re 1μPa at 1 m away from the source(s).

Since both the transmitter and the receiver were idle, any rapid amplitude fluctuations of the received signal (signal fading) are attributed to environmental changes. Figure 4(a) shows the estimated channel scattering function by processing data received on two consecutive days. The intensity plot is in dB scale. At 2:00 am of date 74 (March 14th 2008), a 2 msec delay spread was observed and a small Doppler spread due to calm seas. In contrast, an increase in both delay (by 1 msec) and Doppler spread (by 2 Hz) was noticed at 4:00 pm of date 75 due to higher sea-surface waves. Figure 4(b) illustrates the spatial coherence of the channel based on cross-correlating the output of different sensors of the array. The magnitude of cross-correlation functions is plotted. The averaging window was 0.8 sec. Note that significant decorrelation happens when sensors are separated by more than 48 cm, which corresponds to approximately 4λ at 12 kHz, λ being the acoustic wavelength. Consequently, using consecutive sensors is inefficient in terms of exploiting receive diversity.

Figure 4: (a) Intensity plots of the estimated scattering function. (b) Unbiased cross-correlation function between various sensor outputs.

Now we compare the performance between the TCM-OFDM system and the BICM-OFDM system. The bandwidth of the communication signals was 3906.25 Hz and the carrier frequency was 12 kHz. Packets with 128, 256, and 512 sub-carriers were tested, resulting in data rates up to 5900 bits/sec. Table 1 shows the number of error-free packets for each system. The label 1x1 indicates that one projector and one receive hydrophone (the 1st sensor from the bottom) were used for decoding each packet. Similarly, the labels 1x2 and 1x3 indicate that a combination of two (1st and 12th) and three (1st, 6th and 12th) receive sensors, respectively, were used to decode the packets. BICM clearly performs better than TCM, especially when one receive sensor is employed. When two or three sensors were employed,
BICM performs marginally better, but both systems show improved performance due to harvesting receive diversity.

Here, we report the BER performance between the TCM-STBC-OFDM and BICM-STBC-OFDM systems. The number of sub-carriers was 128. The maximum data-rate the three-projector systems achieved was 4100 bits/sec. Table 2 shows the number of error-free packets. As opposed to the single-projector systems, here TCM performs better than BICM.

<table>
<thead>
<tr>
<th>Date</th>
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<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>BICM</td>
<td>TCM</td>
<td>BICM</td>
<td>TCM</td>
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<tr>
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<td>22</td>
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<td>30</td>
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</tr>
<tr>
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<tr>
<td>March 25th</td>
<td>37</td>
<td>34</td>
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<td>37</td>
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</tbody>
</table>

Table 1: Number of received error-free packets for TCM-OFDM and BICM-OFDM for various numbers of receive hydrophones.

<table>
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<tr>
<td>March 24th</td>
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</table>

Table 2: Number of received error-free packets for TCM-STBC-OFDM and BICM-STBC-OFDM for various numbers of receive hydrophones.

5. CONCLUSIONS

Two systems with the same bit-rate and decoding complexity were designed and compared. The first system is based on TCM, while the second system employs BICM. Testing the systems using both simulated and experimental data from RACE’08, the following result was obtained: the BICM scheme performs better when the UWA channel exhibits a low diversity order (which is usually the case when spatial diversity is very limited). This is because BICM has a higher code diversity (Hamming distance of the code), which is a key parameter towards achieving robust performance in fading channels. The TCM scheme, on the other hand, becomes a better choice when the UWA channel demonstrates a high diversity order (which is usually the case when spatial diversity is sufficient). The reason is twofold: (1) spatial diversity averages out the channel fade, and therefore the effective channel seen by the receiver is an AWGN-like channel; (2) TCM has a higher coding gain, which is the key parameter to achieve better performance in AWGN channels.
REFERENCES


AN INVESTIGATION OF OCEANOGRAPHIC PARAMETERS AFFECTING ACOUSTIC MODEM PERFORMANCE FOR HORIZONTAL DATA TRANSMISSION

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Abstract: This study investigates the performance of underwater acoustic modems for long range horizontal communication in shallow water. The AQUAmode, a frequency shift keying (FSK) based long range acoustic modem developed by Aquatec Group Limited is tested to maximum range in approximately 30m of water on the coast of Western Australia. Accompanied by high frequency noise recorders, data from the trial indicated the sea-bed modem received data effectively at up to approximately 7km in low power mode despite unusually large levels of ambient noise in the area. Communication breakdown for short ranges was found to be caused by long reverberation times, easing with range. When the propagation path was near vertical, modem reception was at the highest possible level. Results from this trial conclude multipath interference to be the dominant source of communication breakdown, as modems were able to operate over long ranges in low SNR environments.

Keywords: Underwater, Acoustic, Modem, Subsea, Communication, Telemetry, Multipath
1. INTRODUCTION

Acoustic communication is a major contributor to subsea wireless telemetry, owing to the partial conductivity of seawater preventing efficient electromagnetic wave propagation. In many cases it has been used in relatively short range applications such as sub-sea sensor networks [1, 2], involving vertical transmission from the sea-bed to sea surface buoys. Data transfer to shore generally involves transmitting above the surface using radio waves due to its high speed and reliability.

This study seeks to aid in research for the oil and gas industry which is attempting to discard traditional forms of off-shore resource extraction methods by submerging all facilities. For this reason, conventional methods of system, environmental and pipeline monitoring need complete revision as RF transfer above the sea surface is no longer an option. This places a significant demand on the horizontal performance of acoustic modems to be used either primarily or as a backup system should an umbilical link fail.

The propagation of high frequencies involved in acoustics is complicated for horizontal transmission due to recurring reflections from the sea surface and sea bed arriving at unpredictable times compounded by refraction throughout a water column [3]. Data communication techniques have advanced considerably over the last decade to better adapt to these effects. However, emerging modems tend to use different principle encoding techniques and filtering, based mainly on the effects of the specific environment they are built for. It is even anticipated that this may compound problems further as an underwater communication standard is yet to be developed, creating the potential for modems in proximity to interfere with each other.

This study investigates the performance of an underwater acoustic modem designed for horizontal communication using frequency shift keying (FSK) techniques. By simultaneously measuring environmental parameters during telemetry, it aims to characterise the various mechanisms affecting transmissions.

2. TRIAL METHOD

For this study, the AQUAmodem from Aquatec Group Ltd was deployed to assess its performance for shallow water horizontal data transmission. Accompanying the modems during the deployment was external hardware built specifically to control and assess the modems during transmission. This included a Silicon Labs microcontroller used to communicate with modems via RS232 and added interfaces for integration with high level control via a PDA. This allowed high frequency recording of the signal using a high speed ADC, writing directly to USB flash disk. Whilst capable of recording at 192 kS/s, recordings for the deployment occurred in 20 minute blocks at 96 kS/s as the AQUAmodem utilises a low frequency range (7.5 - 12 kHz).

The trial was performed using a receiving modem deployed on the sea bed for the duration of the experiment with the omni-directional transducer facing directly upwards. The accompanying software recorded ambient noise whilst awaiting an acoustic command from a transmitter. All modem events were also recorded onto internal PDA memory. When the correct sequence was received, the modem responded with a similar short burst of acoustic data. The primary transmitting modem was configured to send an acoustic request every 20 seconds. Accompanied by a CTD profiler, this was lowered to typically half the water depth for approximately 10 minutes at various positions.
The deployment focussed on shallow water and was performed approximately 10km off the coast of Perth, Western Australia. The main transect was located over a 30m north-south depth contour. For investigation of modem performance over rugged terrain, the modems were also placed on either side of a collection of reef and rocks protruding from the sea surface. This is shown in Fig. 1. By using known bathymetry data for the area, the two dimensional propagation paths were extracted for each test position (TP). This is also shown, demonstrating the propagation paths for TP9 and TP11. The sea-bed substrate for the majority of the deployment was sand covering limestone. The maximum range was attempted at TP9 with a modem separation of 11.2km.

3. COMPARISON MODELS

Acoustic modems typically operate over a limited bandwidth between 1-50 kHz. These relatively high frequencies simplify computational predictions by allowing the application of Ray Tracing [3]. The ray code used in this study was Bellhop, written by Mike Porter at HLS Research. In addition to typical ray code, Bellhop considers each ray as a Gaussian beam of energy decaying over its path. This gives predictions of transmission loss at a receiver position and is particularly useful to predict multipath propagation. After extracting the bathymetry for TP11 as shown in Fig. 1, ray tracing was used to simulate a 9 kHz signal transmitted through the two dimensional model. Rays for the shallow water environment are concentrated, giving transmission loss depicted in Fig. 2.

Fig. 1: Trial area showing deployed positions of the receiver and subsequent testing positions for the towed transmitter (left). Extracted 2d propagation path (right).

Fig. 2: Ray trace model for uneven bathymetry at TP11 (left). Predicted signal loss generated by Bellhop (right).
Despite the long ranges, ray concentration at peaks shown in Fig. 2 (a) indicates signals converging at the receiver would place high demand on internal digital signal processing to correctly compensate with the effects of multiple arrivals. Transmission loss computation for receiver depth of 5m shown in Fig. 2 (b) indicates relatively uniform transmission loss throughout the shallow water environment typical of cylindrical spreading at range, despite the terrain.

4. RESULTS

During the trial, the transmitting modem did not report any successful communication with the receiving modem on the sea floor even when in close proximity and with signals clearly audible to the crew on deck. Whilst data was effectively transmitted to the sea-bed receiver, it was discovered that the response was not successfully decoded by the transmitter, resulting in a half-duplex communication link. A possible cause for this could be interfering vibration from the aluminium mounting plate on the sea-bed modem, as the systems were otherwise identical.

When analysing the reception at the sea bed, it was found that when transmitted horizontally, the signal was mostly undecodable by the receiving modem. However, as the range increased, telemetry resumed successfully. The success rate of the modems for each test position is shown in Fig. 3.

![Fig.3: Reception rate for modem deployed on the sea-bed.](image)

With the exception of near-vertical communication successfully receiving every transmission, performance of the modems gradually improved with increased range. The drop in reception beyond approximately 7km was followed by complete failure at further ranges.

Recordings from the sea-bed equipment were analysed to assess acoustics of the environment over the deployment period. It was discovered that several noise sources were present during the deployment, dominated by unusual amounts of vessel noise towards the end of the day. This was most likely due to deploying in close proximity of a local shipping channel, as well as a rare influx of recreational vessels in the area. Accounting for the drop in performance at TP7, the ambient noise level is depicted in Fig. 4 to exceed received signal strength over time during the communicating period. A summary of signal and noise levels is also shown.
An expected drop in signal strength is exhibited as the modems were placed further apart, following the predicted trend. These results demonstrated SNR levels sufficient for successful transmission at ranges up to 7km, consistent with the modem reception rate. Additionally, the FSK modem performance is shown to be effective in low SNR environments. The modem performance drop for close range communication was found to be unrelated to SNR, also inferring multipath propagation interference at close range.

Signal reverberation was investigated by analysing the tail of received signals. In normal operation over the deployment, modems transmitted for 1.27 seconds. The time taken for the transmitted signal power of known length to decrease to one standard deviation above the noise floor was measured. Shown in Fig. 5, the reverberation time is shown to decrease with range, a negative correlation with successful communication.

Maximum reverberation time during vertical transmission is expected, as reflections occur directly between the sea surface and sea bottom. As range extends beyond the depth of the water, it is shown that the column acts more like a waveguide and reception is minimally affected by long reverberation times. This significantly reduced inter-symbol interference, and modems were able to communicate effectively.

Upon closer analysis of signal patterns, the depth of water could be acoustically confirmed using the first reflection from the sea surface after a sea-bed transmission. Also shown in Fig. 5, following a signal termination, four symbols are clearly echoed in the same sequence as the final four symbols of the packet. The measured time delay of 42ms correlates with a depth of approximately 30m.
5. CONCLUSIONS

The shallow deployment of the devices caused significant multipath interference which was shown to decrease with range. Nearby vessel noise impacted modem performance throughout the trial, although successful reception was possible at ranges up to 7.2km with the AQUAmodem low power setting. No signals were detected by the ambient noise recorder at testing positions beyond this range. It is expected that given the unusually high ambient noise levels particularly during the later part of the day, successful communication may have been achievable at further ranges. Despite this, the modems have been shown to operate in very low SNR environments. However, inter-symbol interference at short ranges in shallow water has been shown to significantly inhibit performance.

By accessing debug information from the internal modem memory, data regarding both the bit error rates and raw recordings can be analysed to determine modem performance over the duration of the deployment. This can help confirm multipath interference as the primary cause for lack of modem communication in low ranges. Furthermore, by analysing modem recordings for similar transmissions, the cause of half duplex communication can be determined.

Future deployments will involve longer transects and deeper waters to help assess performance and characterise effects of other oceanographic parameters. Specifically, a long term deployment in deep water will give insight into the impact of wind noise and changes in sound speed profile. Eventually, effective characterisation of the parameters affecting successful underwater telemetry will aid in ensuring reliable subsea links, wherever needed.

Whilst analysis is ongoing, preliminary results have given strong indication that for shallow water deployments, multipath propagation causes significant inter-symbol interference. Environments close to shore have also been shown to exhibit high levels of man-made ambient noise. Although the modems in this trial were not designed for shallow water environments, they have been shown to operate effectively at long ranges where inter-symbol interference is reduced.

6. ACKNOWLEDGEMENTS

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REFERENCES

EXPERIMENT RESULTS OF TIME-REVERSAL COMMUNICATION IN THE DEEP OCEAN AT THE RANGE OF 300 KM

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Abstract: In our research program, a long cruising autonomous underwater vehicle (AUV) has been developed. Time-reversal techniques have a possibility to realize long horizontal communication with such AUV, by converging multipath signals to the focus and decreasing intersymbol interference (ISI). We have researched on time-reversal communication in the deep ocean and have proposed a method of combining time reversal and adaptive equalization. In the previous at-experiments, communications at the range from 10 to 100 km were demonstrated with synthetic time-reversal array. In this experiment, passive time-reversal communication was achieved with a real receiver array to the range of 300 km first as the experiment intended for communication. Several kinds of probe signals are used at the data rates of 50 and 100 bps. In spite of Doppler effect, the proposed method achieves demodulation with no error.

Keywords: time reversal, phase conjugate, underwater acoustic communication, adaptive equalization, Doppler effect
1. INTRODUCTION

In the Japan Agency for Marine-Earth Science and Technology (JAMSTEC), researches on underwater acoustic technologies are assigned to our research group. In our program, an autonomous underwater vehicle (AUV) was developed and the first in the world to successfully travel up to 300 km. One of our goals is to realize communication with such a long-cruising type AUV. However, as is well known, it is difficult to achieve communication horizontally in the ocean because many multipath signals cause intersymbol interference (ISI).

Time reversal communication has a possibility to overcome such a problem [1, 2], which is implemented as shown in Fig. 1. A probe pulse is transmitted from the original source, received at the time-reversal array (TRA), and time-reversed. When these time-reversal signals are modulated with digital data and transmitted sequentially, multipath signals are converged to the focus in space and time and data can be received without ISI; thus, communication can be achieved. Therefore, time reversal can be interpreted as a method of channel equalization using an acoustic medium itself. Such multiple-input-single-output (MISO) time-reversal communication is called active time-reversal communication. In the meantime, in the case of single-input-multiple-output (SIMO), it is called passive time-reversal communication [2]. In passive time reversal, a probe signal and data signal are transmitted continuously from a source, and received at an array. These signals are cross-correlated and summed over the channels. The correlation of two signals is equivalent to the convolution in which one of the two signals is time-reversed. Thus, the same focusing effect as active time reversal can be achieved by passive time reversal.

We previously proposed a method of combining time-reversal focusing and adaptive equalization [3-10]. In this method, after multipath signals are utilized by time reversal as much as possible, residual nonconvergent signals are removed by an adaptive equalizer. Then, communication can be realized even with a sparse TRA.

![Fig. 1 Concept of time-reversal communication.](image)

At-sea experiments at ranges of 10 to 100 km have been carried out [11,12]. It is demonstrated that time-reversal communication can be accomplished first in the deep ocean and outperforms the conventional method. In addition, with the use of real data obtained in the Project of Ocean Acoustic Tomography (OAT), the performance of time-reversal communication at ranges over 1000 km is predicted. From these results, it is clarified that the proposed method is effective.

In this paper, the results of our recent at-experiment at the range of 300 km are described.
2. EXPERIMENTAL PROCEDURE

In our first experiment [11], bi-directional active and passive time reversal communications were executed with one pair source-receiver systems while changing the depth of the TRA side system using underwater winch. As one of results, it is confirmed that the same converged signal can be obtained in fact with both active and passive time reversal. In the subsequent experiments at the range from 20 to 100 km [12], measurements for passive time-reversal communication were carried out with a single hydrophone suspended from the research vessel while changing its depth. These experiments were implemented, so to say, with a virtual time-reversal array.

The experiment described in this paper was executed at the range of 300 km in the 4000-deep and flat bathymetry area as shown in Fig. 2. In this experiment, a real receivers array was used, which was composed of 20 receiver systems. The receiver system consisted of a hydrophone, control unit, A/D converter, pre-amp filter, data memory, and battery, and works according to timer schedule. This 20-channels receiver array was moored at the point indicated as “Rx” in Fig. 2 (a). The intervals between the receivers were 6.0 m approximately. The array depth was 1,300 m approximately as shown in Fig. 2 (c), illustrated with the sound velocity profile around the mooring point. The hydrophone was HTI-90-U. The projector, EAI-500D, was used as a source, whose center frequency and bandwidth is 500 ± 50 Hz. The source system was suspended from the research vessel to the depth of 1,000 m approximately as shown in Fig 2(b), illustrated with the source velocity profile. During measurements, the ship position was kept as constant as possible, around the point indicated as “Tx” in Fig. 2 (a). As shown in these figures, the source and receivers are placed around the axis of SOFAR channel.

In this experiment, only passive time-reversal communication was performed. The same converged effect can be obtained with both active and passive time reversal. Therefore, the results can be interpreted as the performance prediction of active time reversal as well as passive time reversal itself.

![Fig. 2 (a) The experiment site. (b) The sound velocity profile around the source point. (c) The sound velocity profile around the receiver array point.](image-url)
In this experiment, four kinds of probe signals are used: (1) chirp pulse with duration 5.0 s, (2) chirp pulse with duration 10.0 s, (3) 8th M-sequence, and (4) 9th M-sequence. In the cases of (1) and (2), the sweep bandwidth is 475 to 525 Hz. In the cases of (3) and (4), M-sequences for the probe signals are filtered with 4th root-raised cosine filter. Their received signals are correlated with the original sequence before passive time-reversal process. In the meantime, data signals are filtered root-raised cosine filter. Through the cross-correlation of passive time reversal, data signals filtered with raised cosine filter can be obtained while multipath signals are eliminated by time-reversal focusing.

In all the cases, data were modulated with binary phase shift keying (BPSK) with the rate 50 bps. Therefore, the durations of the probe signals of (3) and (4) are 5.0 s and 10.0 s approximately. Additionally, in the cases of (3) and (4), communication with the rate of 100 bps was also executed. In that case, the durations are 2.5 s and 5.0 s approximately.

3. EXPERIMENT RESULTS

In this section, the experiment results are described, with demodulated symbols plotted on the constellation map. In this experiment, 2047 symbols were transmitted.

In Fig. 3 (a) and (b), the results are shown in case that the probe signal is the chirp pulse with the duration of 5.0 s and 10.0 s, respectively. In these figures and hereafter “TR only” indicates using only time reversal, and “TR+AE” indicates the proposed method of combining time reversal and adaptive equalization. In case of the proposed method, the initial 200 symbols are used for training of the adaptive equalizer.

In these results, the demodulated symbols are rotated in the case of TR. This phase rotation is supposed to be due to the ship drifting, although the ship position was kept as far as possible. Such Doppler effect is compensated in case of TR+AE.

In Fig. 4 (a) and (b), the demodulation results are shown when the probe signal is 8th and 9th M-sequence, respectively, and the data rate is 50 bps. In these results, phase rotations due to Doppler effect are observed in the cases of TR while such effect is compensated and demodulations are achieved with no error in the case of TR+AE. Comparing Fig.3 and 4, M-sequence pulses as probe signals give better results than chirp pulses. The correlation result of chirp pulse has longer sidelobes, which cause residual ISI after passive time-reversal

![Fig. 3 Demodulation results when the probe signal is a chirp pulse with the duration (a) 5.0 s and (b) 10.0 s.](image-url)
correlation process and degrade demodulation results. Comparing Fig. 4 (a) and (b), the length of M-sequence does not improve communication accuracy in this experiment conditions.

![Fig. 4](image_url) Demodulation results when the probe signal is the (a) 8th and (b) 9th M-sequence pulse with the data rate 50 bps.

In Fig. 5, the results when probe signal is M-sequence and the data rate is 100 bps are shown. In these results, the improvement with longer M-sequence probe signal is observed. Moreover, comparing the results in the cases of 50 bps, the result of Fig. 4 (b) has the best performance. If the source and receivers are stable and the environment is time-invariant, it is expected that longer M-sequence bring in the better SNR after the correlation. However, when the Doppler effect is accompanied, longer M-sequence is unfavourable. Therefore, shorter M-sequence with the data rate 100 bps brings in better results than 50 bps. Note that the effective data rate decreases as a matter of course when a longer probe signal is used.

![Fig. 5](image_url) Demodulation results when the probe signal is the (a) 8th and (b) 9th M-sequence pulse with the data rate 100 bps.

**4. CONCLUSIONS**

The experiment of time-reversal communication in the deep ocean was demonstrated with several kinds of probe signals. The time-reversal communication at the range of 300 km was achieved first as the experiment intended for communication. It is confirmed that the
proposed method has an impact to compensate Doppler effect. In this experiment conditions, the communication at the rate of 100 bps is robust rather than at the rate of 50 bps. Unfortunately, the longer-range measurements could not be executed due to the bad weather. The comparison with the conventional multichannel DFE has not yet been performed. It is expected that time reversal has an advantage at the longer distances where more multipath signals are received. These issues will be studied in future work.

ACKNOWLEDGEMENTS

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REFERENCES

CHARACTERISATION OF TIME-VARYING UNDERWATER ACOUSTIC COMMUNICATION CHANNEL WITH APPLICATION TO CHANNEL CAPACITY

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Abstract: The properties of the underwater acoustic communication channel at medium range and medium frequency are characterised by analysing a set of at sea data collected in coastal environment (depth 10-40 meters, range 1-3 km, frequency 11.2 kHz or 17.5 kHz). Based on an empirical mode decomposition, we infer that the channel behaves as trend stationary random process over a few minute duration. Moreover, according to statistical tests, the channel envelope is found to be in good agreement with a Rice process whose power ratio between the mean and the scattered part is time variant and delay dependent. As a conclusion, the results of attempts to apply some approximate bounds for the capacity of WSUSS Rice channel to the measured channels are given. It is found that the channel capacity is generally close to Shannon capacity when the channel response includes a few specular paths but significantly decreases with the number of paths and the diffuseness of the channel.

Keywords: Underwater Acoustic Channel, Propagation, Empirical Mode Decomposition, Capacity, Statistical Inference, Maximum Entropy.
1. INTRODUCTION

In order to anticipate in laboratory the performance of acoustic communication systems in real underwater environment, propagation channel models are essential. The underwater medium is indeed inhomogeneous, bounded by rough boundaries and in constant motion, so that transmitted signals suffer from strong fading, time spread and high Doppler. Underwater acoustic channel (UAC) modelling is usually either deterministic and physics-based or stochastic or a combination of both (the moment of the stochastic model being computed from the physical parameters).

Based on a set of measures collected at sea in a coastal environment, we here study the validity of stochastic modelling and infer some statistical properties of the shallow water UAC. The time-varying channel responses \( h(t,\tau) \) are computed in different scenarios during several minutes so that the distribution of the scattered random components as well as their time variation can be accurately estimated.

As a conclusion, on the basis of the results found in the statistical study, capacity approximate bounds of the measured channels are derived.

2. SEA EXPERIMENTS AND CHANNEL ESTIMATION

All the data presented in this contribution result from an experiment conducted in the vicinity of Brest in France (Atlantic Ocean) in October 2007. This experiment was set up as described hereafter:

- Transmission between a surface ship towing an acoustic source and a 4-element hydrophone chain submerged 80 meters from the shore.
- Distance range: 1000-3000m
- Ocean depth: 10-40m
- Calm sea, little swell
- Transmitted signal: QPSK modulation at baudrate \( R = 2.9 \) kBds or \( 4.35 \) kBds, carrier frequency at 11.2 kHz or 17.5 kHz, vocoded data+FEC.

The experiment was not originally designed for channel sensing so that no dedicated probe signal, such as maximum length sequence (LMS) data-aided adaptive channel estimation. All the transmitted data were perfectly known at reception. Note that when using wideband signals, the moving plate-forms and/or the medium motion induce a Doppler effect that needs to be mitigated in order to study the intrinsic properties of the UAC. This effect can indeed obscure the true channel Doppler spread for instance [4]. Doppler mitigation was thus performed by resampling/interpolation based on an open-loop scheme [1] for coarse correction and on a closed-loop [2] for fine time recovery. The channel was estimated with a resolution of \( 1/(2R) \) in the delay domain and updated at a rate of \( R/40 \).

Figure 1 shows 4 examples of estimated impulse responses.
3. SLOW VS FAST FADING

The observation of the measured impulse responses over 150 seconds suggests that signals propagating through an UAC are affected by fading phenomena of different scales. Fading is usually qualified as slow or fast to refer to the rate at which the magnitude and phase of the channel fluctuate compared to the transmission baudrate. Fast fading is predominant in the design of communication systems since the adaptive algorithms at reception must be tuned to its characteristics to ensure good performance. Slow fading is less critical as it represents only a long-term variation of the signal-to-noise ratio at reception. By looking at the envelope of the measured UACs (see Figure 2), it can be seen that the two fading types are combined in an additive way. The average value of the received power is time variant whereas the second order statistics on this received power seem to be invariant. On the observation window of 150 seconds, the UAC can therefore be considered as trend stationary and each path can be split up as:

\[ \hat{h}(t, \tau_k) = d_k(t) + w_k(t) \]  

(1)

where \( d_k(t) \) is a pseudo-coherent component that behaves almost as if the medium was deterministic and \( w_k(t) \) is the scattered random part. Note that \( d_k(t) \) can be considered as pseudo-deterministic and \( \hat{h}(t, \tau_k) \) as trend stationary only because the observation window is time limited compared to the fluctuation speed of the underlying physical phenomena. A longer observation window may lead to different conclusions.
In order to isolate $d_k(t)$ and $w_k(t)$ from $\hat{h}(t, \tau_k)$, an empirical mode decomposition (EMD) is performed on each path of the UAC. EMD is a method of signal decomposition, well suited to non stationary signals, that does not require any predetermined basis functions. The decomposition is designed to seek the different intrinsic modes of oscillations (or rotations in the complex case) in any data, based on the principle of local scale separation [5][6]. Each path is then expressed as:

$$\hat{h}(t, \tau_k) = \sum_{i=0}^{L_k} m_{i,k}(t) + \sum_{i=L_k}^{N_k} m_{i,k}(t) + r_k(t)$$ (2)

where $m_{i,k}(t)$ is the $i$-th of the $N_k$ modes, $r_k(t)$ is the residue component and $L_k$ is the decomposition order used to separate $d_k(t)$ and $w_k(t)$. These two processes varying at different paces (less than few seconds for $w_k(t)$ and 10 seconds to several minutes for $d_k(t)$), the choice of $L_k$ is based on a frequency criterion on the path envelope.
4. STATISTICAL INFERENCE

Thanks to the EMD, it is now possible to study the characteristics of the two fading components independently. Since fast fading is more critical for communication system design than slow fading, we here only focus on the characterisation of the scattered random components \( w_k(t) \). To model their probability density function (pdf), we applied the principle of maximum entropy \([7]\) constrained by some moments of different orders measured on the estimated impulse responses. Kullback-Leibler (KL) divergences between the model and the empirical pdf are then assessed for the tested moment orders.

<table>
<thead>
<tr>
<th>Moment order</th>
<th>0</th>
<th>( \leq 1 )</th>
<th>( \leq 2 )</th>
<th>( \leq 3 )</th>
<th>( \leq 4 )</th>
<th>( \leq 5 )</th>
<th>( \leq 6 )</th>
</tr>
</thead>
<tbody>
<tr>
<td>KL divergence</td>
<td>0.64</td>
<td>0.62</td>
<td>8.8 ( \times ) 10(^{-3} )</td>
<td>7.5 ( \times ) 10(^{-3} )</td>
<td>7.5 ( \times ) 10(^{-3} )</td>
<td>6.3 ( \times ) 10(^{-3} )</td>
<td>6.1 ( \times ) 10(^{-3} )</td>
</tr>
</tbody>
</table>

Table 1: Example of measured KL divergence for different constraints on the model

As illustrated in Table 1, the KL divergence converges as soon as the second order constraint is applied to the model of maximum entropy. This indicates that the \( w_k(t) \) can be well modelled by a stationary Gaussian process. This has been confirmed by a Kolmogorov-Smirnov \([8]\) test applied to the envelope of the \( w_k(t) \) that must follow, from the model, a Rayleigh distribution. 96% of the tested paths pass the test with a significance level greater than 5%. Consequently, each path \( \hat{h}(t, \tau_k) \) can be modelled as a Rice process with a slow time varying mean component. In addition, as shown in, the power ratio (Rice factor) between the mean component \( d_k(t) \) and the scattered part \( w_k(t) \) is delay dependent and tends to zero (Rayleigh fading) for the most delayed paths. Note that the fluctuation period of the mean component being on the order of several thousands to hundreds of thousands symbols, the communication channel can be well approximated by a wide-sense stationary process.

![Figure 4: Scattering function example of the \( w_k(t) \)](image)

To characterise into more details the fast fading components \( w_k(t) \), their coherence time with 50% correlation have also been measured. All results obtained are within the range of 300 to 600 ms. Figure 5 shows an example of the \( w_k(t) \) scattering function for a measured UAC impulse response.

![Figure 5: Measure of the Rice factor as a function of the path delays](image)
5. ERGODIC CHANNEL CAPACITY

Ergodic capacity of a transmission channel is here considered since it provides a simple bound, independent of the actual signaling scheme and receiver algorithm. In underwater acoustic, at our best knowledge, this capacity has only been computed by modeling the UAC by a band-limited AWGN channel [12]. In agreement with the results found in the previous sections, the UAC, after removing long term slow fluctuations, is here modeled as a WSSUS [11] channel with a possibly non zero mean.

Applying same techniques as in [9], an upper bound can be derived for the ergodic capacity of this channel:

$$\text{Capacity} \leq \max_{\alpha, \beta} \left\{ \int_{-1/2}^{1/2} \log \left( 1 + \alpha \frac{\sigma^2_H + |H(\nu)|^2}{\sigma^2_w} \right) d\nu - \alpha \int \int \log \left( 1 + \beta \frac{S_H(v, \tau)}{\sigma^2_w} \right) dw d\tau \right\}$$  \hspace{1cm} (3)

Here $H(\nu)$ stands for the Fourier transform of the mean of the channel response, $\nu$ is the normalized frequency, $\sigma^2_H$ the sum of the variance of all paths and $S_H(\nu, \tau)$ the scattering function [11] (with reduced units $\nu, \tau$). The noise is assumed to be Gaussian, zero-mean and i.i.d. with standard deviation $\sigma_w$ (such that $\sigma^2_w = N_0 W$; $W$ being the used bandwidth). The input symbols $s_k$ are assumed to be zero-mean, i.i.d. and such that $\mathbb{E}(|s_k|^2) \leq 1$ and $|s_k|^2 \leq \beta$ (both average and peak transmitted power are limited).

It can be seen that the RHS of (3) is the sum of two terms: a first term which is the capacity of a AWGN channel (constant or time variable, but perfectly known) and a second term which is the capacity loss due to the unknown fluctuations of the channel. It is also worth noticing that maximization with respect to $\alpha$ of the expression (3) is simple when $H(\nu)$ is null or constant which corresponds to at most one non zero-mean path.

![Figure 6: Channel capacity for a single path Rayleigh channel](image)

A first example of the resulting bound on capacity is given Figure 6 in the case of a single path Rayleigh channel with Gaussian scattering function. The two different plots correspond to two values of the coherence time $T_c$ the channel (in symbols); on each plot the red curves correspond to different values of the Peak-to-Average Power Ratio (PAPR) $\beta$. Capacity of AWGN channel is also plotted (black curve) since it provides an upper bound of the channel capacity in any circumstances.
It can be seen on this figure 6 that, for short coherence time $T_c$ ($R.T_c =10$ where $R$ is the symbol rate) the capacity is significantly lower than for the AWGN channel. For larger coherence time $T_c$ ($R.T_c =10$) the difference is smaller or even negligible. It also worth noticing that for large value of $R.T_c$ the capacity bound (3) is maximized for $\alpha$ equal to 1 (transmitted power is averaged limited, constant modulus modulation as PSK). On the contrary, for small values of $R.T_c$, it is no more possible to use constant modulus modulations (phase tracking becomes impossible) and incoherent spiky modulations, where information is carried by amplitude or epoch of spiky pulses, have to be used. For small value of PAPR $\beta$, the parameter $\alpha$ then have to be taken lower than 1 and the average power chosen lower than maximum allowable.

![Figure 7: Capacity of a WSSUS channel with a rectangular scattering function](image)

A second example is given in Figure 7 for a WSSUS Rayleigh fading channel with a rectangular scattering function defined as

$$S_H(\nu, \tau) = \frac{\sigma_H^2}{\Delta_s} \text{rect}\left(\frac{\nu}{D_s}\right) \text{rect}\left(\frac{\tau}{T_s}\right)$$

where $D_s$ and $T_s$ are respectively the Doppler and time spread and $\Delta_s$ is the spreading factor, $\Delta_s = D_s T_s$. It can be shown in this particular case that the capacity only depends on $\Delta_s$, $\beta$ and the SNR. Figure 7 indicates that the channel capacity is close to the AWGN channel capacity for small spreading factor $\Delta_s$, but significantly lower for larger spreading factor.

In practice, the acoustic channel analyzed in section 2-4 above corresponds to favorable situations (spreading factor $\Delta_s$ about $5.10^3$ according to Figure 5), even more favorable than the one on figure 6b-7b. Its ergodic capacity is therefore likely very close to the AWGN channel capacity. This is not always true since far more difficult UACs have already been observed, with spreading factor $\Delta_s$ up to 0.1 or higher, for which the channel capacity vs SNR will be similar to figure 6a and 7a.

6. CONCLUSIONS AND PERSPECTIVES

A set of statistical tools and methods has been presented to characterize a given UAC as a transmission channel to then bound its capacity according to the outcomes of this characterization. These tools have here been applied to a first set of at sea data: the
corresponding channel has been found in good agreement with a multipath Rayleigh or Rice fading channel with a small spreading factor and a capacity likely close to AWGN channel’s. Further works will be devoted to the analysis of other at sea datasets corresponding to more difficult channels. During some experiments, data transmission has been found to be only possible with data rates far lower than the ergodic capacity of the corresponding AWGN channel. The aims of these analyses will then be to determine whether the transmission rate has been limited by “poor” signaling schemes and transmission algorithms or by the channel itself. In other respects, it would also be interesting to address the impulsive characteristics of the ambient noise and its consequence on the channel capacity.

7. ACKNOWLEDGEMENTS

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REFERENCES

ACOUSTIC DATA LINK SYNCHRONIZATION INACCURACY DUE TO MULTIPATH EFFECTS ON WIDE-BAND SYNCHRONIZATION PULSES (SWEEP SPREAD SIGNALS)

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Abstract: The use of Sweep-Spread Technology enables the resolution of multipath arrivals in frequency domain. After synchronization of the receiver with the receive time of a dominant multipath arrival the influence of another (time-delayed) multipaths can be effectively suppressed. If the synchronization has been done accurately, the difference frequency of the dominant multipath arrival is equal to zero and the outputs of the receiving matched filter can be used for correct estimation of the received signal phase (the information). However, if strong interfering multipath comes to receiver closely to the dominant multipath arrival, the evaluation of the receive time of this arrival can become inaccurate. In result, the synchronization of the receiver becomes not correct and (during demodulation) the difference frequency of this arrival can differ from zero. Such effect can result into decrease of signal-to-noise ratio and into deterioration of acoustic data link performance. The paper represents the results of mathematical and numerical modelling of the synchronization inaccuracy arising due to multipath interfere contained in wide-band synchronization pulses (swept signals) with different parameters.
Abstract: The research project UUV Covert Acoustic Communications aims at furtive communication between unmanned underwater vehicles and a distant mother platform. Suitable modems were developed, which were tested in Norwegian coastal waters during a demonstration trial in September 2008. One modem was lowered into the water column from a mother ship, whereas the other modem was integrated in an autonomous underwater vehicle. In order to address the aspect of covertness, intercept sonobuoys were dropped in the area and their data processed by an automated energy detector and a sonar operator. The results show that it is difficult to communicate covertly under all circumstances, especially if the interceptor is close to the transmitter or experiences more favourable propagation conditions than the communicator. Conversely, covert operation is possible when the communicator experiences better conditions than the interceptor. In case of equal conditions, the communication can also be covert with the interceptor requiring some 6 dB higher SNR for detection than the communicator for decoding. Moreover, the communication link proves very robust with respect to interference whereas the energy detector suffers from multiple false alarms in the presence of transients unrelated to the communication system.

Keywords: Covert underwater communication, acoustic modems, orthogonal frequency-division multiplexing, interceptor.
1. INTRODUCTION

The future role of unmanned underwater vehicles (UUVs) is believed to include autonomous tasks in environmental monitoring, route survey, mine countermeasures, etc. Depending on the mission, there may be a need for a robust, covert communication link with a remote mother platform. In order to enable such a capability, suitable hardware and software were developed within the project UUV Covert Acoustic Communications (UCAC, 2005–2008). Several candidate covert modulation schemes were explored, and a pair of low-frequency (1.5–5.1 kHz) acoustic modems were designed and built. The frequency band was chosen so as to balance between the large bandwidth generally required for (spread-spectrum) covert communications, and a user requirement of working ranges up to tens of kilometers. Sound attenuation in the oceans, which rapidly increases with the frequency, prohibits the use of higher frequencies over such ranges.

Covert communication was defined in the project as the ability to communicate between the UUV and a mother ship, with a low probability of intercept by third parties. In order to keep the project unclassified, intercept techniques are limited to energy detection, and LOFAR and audio inspection by a sonar operator. A simple view considers a three-dimensional volume around the transmitter, not necessarily continuous, where intended receivers can detect and decode the message, and another 3D volume where an interceptor can detect the signal. Covert communication is possible, then, if the communicator is within the former volume and the interceptor outside the latter volume. In practice the location of possible interceptors is unknown, and covertness cannot be guaranteed. Much also depends on the prior knowledge of the interceptor and the tools it has available. For example, an energy detector that is perfectly matched to the frequency band and duration of the communication signals has a better chance of detecting information traffic than an energy detector operating with an integration window that is mismatched in frequency and/or time. If the opponent is a former friend, he may even be able to detect and decode the message coherently.

The present paper reports results from a demonstration trial executed in Norwegian waters in 2008. A communication link was established between a mother ship and a UUV, while intercept sonobuoys were dropped in the area and their data processed by an energy detector and sonar operator.

2. MODULATION SCHEME

Several candidate covert modulations were examined in the UCAC project [1–4]. One modulation scheme, multiband orthogonal frequency-division multiplexing (OFDM), was selected for modem implementation. Among the merits of this OFDM scheme are a short computation time, an excellent SNR performance, and a noise-like auditive character. Two data rates were implemented, a relatively high data rate of 80 bit/s, and a low rate of 4.2 bit/s. These rates are referred to as HDR and LDR, respectively. The corresponding signal lengths are 8 and 30 s in a frequency band from 1.5 to 5.1 kHz. Messages are protected by channel (turbo) coding and by repetition coding. The repetition coding is performed in time and frequency, which makes the OFDM scheme very robust against many forms of interference. For a detailed description of modulation and receiver the reader is referred to [2]. However, note that the implemented high-rate signal differs from the high-rate signal in [2] in that only one turbo code block is used.
3. INTERCEPTOR

One task of the demonstration trial was to gather information about possibilities to detect underwater communication sequences by an automatic interceptor in order to judge the level of covertness of the communication signal. Normally, visual signal inspection in combination with hearing should give the best results, as the human brain is faster and better in pattern recognition than automated algorithms. However, for large data records (e.g. many bearing directions, 24 hours-a-day observations, broad frequency range) an operator is overcharged and requires assistance from automatic algorithms that perform at least a preliminary selection of suspicious signal clips. Therefore, automatic algorithms scan the signal under test for features that indicate (hopefully, from the interceptor’s point of view) communication sequences.

The interceptor used within this project is a particular choice of energy detector. It is not specialized in communication sequences but integrates energy of arbitrary origin. Thus, an alarm can also be caused by active sonar use or any other sound source that produces acoustic energy on top of the ‘normal’ spectral background. However, these false alarms with respect to communication signals can be valid detections for a more general interceptor. The present interceptor segments the signal into short blocks that are transformed to the spectral domain. Based on a predefined number of short spectra in a sliding window, energy-related features (e.g. the variance) are calculated separately in overlapping frequency bands. A normalization step reflects that the intensity of the background noise depends on the frequency band and results in normalized feature matrices over time and frequency bands. The features are combined into a detection curve over time and a detector decides on the presence of communication signals—or more generally suspicious energy peaks.

4. DEMONSTRATION TRIAL

The final UCAC sea trial was conducted in September 2008. Experiments were performed in the North Sea west of Bergen, Norway, as well as in the nearby Bjornafjorden. One UCAC modem was integrated in a Hugin 1000 UUV, whereas the other one was deployed from a surface ship: see Fig. 1. Bi-directional communication was pursued. A typical experiment consisted of a modem operator on the ship requesting depth or heading information from the UUV. The query was encoded in an OFDM waveform, which was sent out to be detected in real time by the UUV modem. Following decoding of the query, the UUV would respond with its depth or heading, also encoded in an OFDM waveform, to be detected and decoded by the modem deployed from the mother ship. All detected waveforms were stored on hard disk for future analysis. Another experiment was for one of the modems to transmit a predefined sequence of nine HDR and nine LDR signals at a source level decreasing in 3-dB steps. The other modem would either be given the task to detect and decode all these signals in real-time, or it could be set in a continuous recording mode. The latter experiment was scheduled to enable a comparison between communicator and interceptor operating under identical conditions. To this end the OFDM receiver and interceptor software would be applied to the same recorded data.
5. RESULTS

Figure 2 shows the geometry of an experiment in the North Sea west of Bergen. The mother ship was moored close to the shore and its modem was used as the transmitter. A sonobuoy was dropped in the vicinity of the receiver modem on the UUV, but at a different depth. The sea was calm and there was low shipping traffic in the test area. A transmission loss calculation (right panel in Fig. 2) reveals that the intended receiver on the Hugin UUV and the sonobuoy experience approximately the same loss.

Nine pairs of communication signals (HDR/LDR) were transmitted at a decreasing source level. The interceptor—processing the sonobuoy data—detects the first three HDR and the first two LDR sequences (red marks in Fig. 3), although more sequences are visible and audible. The detection threshold for the interceptor is chosen so as to balance between detections and false alarms for the whole experiment. During this experiment, there was only one false alarm in 2 hours. When the same sonobuoy data are processed by the OFDM
receiver, the gain of its matched-filter is exploited and coherently detected signals, indicated by the green marks in Fig. 3, outnumber the interceptor detections. Moreover, the OFDM decoding is error free for all detected LDR sequences. So, for this experiment, under equal environmental conditions, the OFDM receiver has \( \approx 6 \) dB advantage in detection and demodulation of LDR communication signals in comparison with the interceptor detections.

Fig. 3: Comparison of interceptor (red) and OFDM receiver (green) detections. Expected signal arrivals are indicated in blue.

The detection performance of the OFDM receiver processing the data stored by the Hugin modem is comparable to that of the interceptor sonobuoy. The receiver looses its coherent detection advantage, presumably due to the self-noise of the UUV. Indeed, many false alarms result if the interceptor is applied to the UUV modem data, and the number of proper detections is low. The communication link is considerably more robust against noise interference than the energy detector.

Further data analysis shows that it is difficult to establish a bidirectional covert communication link under all environmental circumstances, especially if the interceptor is positioned close to one of the transmitters or experiences more favorable propagation than the communicator. Notice in this regard that propagation conditions relevant to the interceptor are limited to transmission loss and noise level. The communicator, on the other hand, is additionally confronted with multipath propagation and Doppler spreading. At a given SNR, an increasing delay-Doppler spread will lower the performance of the communication scheme, whereas the performance of the interceptor is unaltered as it just integrates energy in a wide time-frequency window.

Figure 4 shows example results of a scenario that is difficult for the communicator. The distance between the two modems was 19 km. The sonobuoy was positioned in between transmitter (Nøkken) and receiver (Hugin), at a distance of 4 km from Hugin. Halfway the sonobuoy recording, the modem depths were changed:
a) Nøkken 20 m; Hugin 17 m; Sonobuoy 60 m
b) Nøkken 80 m; Hugin 88 m; Sonobuoy 60 m

Blue marks indicate positions for expected queries from the mother ship Nøkken. If the UUV detects and decodes a Nøkken query, it responds (green marks). All responses were intercepted. Furthermore, some of the queries that were not detected by the UUV were also intercepted. The false alarm around $t = 670$ s is due to a sudden change in the background noise.

Fig. 4: Example of two-way communication in the North Sea. Blue marks indicate the known positions of queries from Nøkken. Green marks indicate responses from Hugin. The white curve is the scaled detector output compared to a fixed threshold.

On the other hand, covert communication is possible if the communicator experiences more favourable (noise) conditions than the interceptor. In case of a highly fluctuating noise background (caused by ship traffic, all kind of transients, sea state) the OFDM receiver proved to be very robust whereas the detector suffers from multiple (false) alarms not related to the communication link. Figure 5 shows an example.
This experiment was conducted in the Bjørnafjorden over a 10-km range between Hugin (TX, 90-m depth) and Nøkken (RX, 90-m depth). The sonobuoy (60-m depth) was not within the line of sight, but roughly 5 km closer to Hugin than the intended receiver. Figure 5 reveals a good performance of the OFDM receiver on Nøkken, detecting many HDR and LDR arrivals, and correctly decoding all detected LDR signals. Despite the favorably short distance of the sonobuoy from the transmitting modem, the interceptor job was very difficult owing to the noise conditions. There were two false alarms and no automatic detections of communication sequences, as seen in the right side of Fig. 5. However, upon close inspection by an operator, the first HDR/LDR pair in Fig. 5 is visible.

The general impression is that the environmental conditions were mostly quite difficult for the proposed, relatively simple interceptor. The interceptor performance is good for signals with a stationary background. Still, for such background conditions the SNR of the communication sequences can often be lowered to a value where the communication is said to be covert, at least at 4 bit/s. For channels with strong background interference the interceptor is highly susceptible to false alarms. On the other hand, under bad environmental conditions the OFDM receiver needs a sufficient SNR for detecting and decoding too. Then the communication signal is often (at least) visible in the spectrograms.

6. CONCLUDING REMARKS

Covert acoustic communication has been achieved between a UUV and a distant mother platform. Here, covertness is defined as the property of communication waveforms to be undetectable for energy detectors, and for sonar operators inspecting audiograms or listening to wavefiles. However, it is difficult to guarantee covertness as much depends on the location of possible interceptors relative to the communicators, and on the propagation conditions experienced by both. More often than not the assumption of stationary noise is violated by breaking waves, intermittent rain showers, shipping traffic, and especially platform noise such as produced by rudder action or the variety of acoustic instruments normally present on UUVs. The communication receiver proved very robust with respect to transient noise sources, which lead to false detections for the energy detector. On the other hand, the energy
detector is more robust with respect to large delay-Doppler spreads inflicted by the acoustic channel. In relatively benign communication channels with stationary noise, the 4-bit/s OFDM communication scheme [2] has a ≈6 dB advantage over the energy detector. Upon introducing transient noises the advantage grows larger, but if instead the delay-Doppler spread of the channel is aggravated, the balance shifts in favor of the interceptor.

In [5], Park concludes that it is relatively easy to communicate covertly when the interceptor is further from the source than the intended receiver, whereas it is almost impossible to be covert when the order is reversed. However, Park assumes stationary white Gaussian noise and does not consider the effects of delay or Doppler spreading. Although his conclusion may serve as a useful rule of thumb, we find that anything can happen in practice because the underlying assumptions are often not valid.

7. ACKNOWLEDGMENT

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REFERENCES

A VOLTERRA-WIENER NONLINEAR SIGNAL PROCESSING TECHNIQUE FOR HYDROACOUSTIC DOPPLER MITIGATION

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Abstract: A nonlinear signal processing technique for hydroacoustic Doppler Effect mitigation is developed. The synthesis method is based on the Volterra-Wiener theory of nonlinear systems combined with the Feedback System Inversion Lemma. A moving acoustic source transmits a reference signal, nonlinearly modulated by the motion characteristics. Such a reference signal can be a monochromatic or polychromatic tone as well as random band-pass signals with known statistical properties. A nonlinear Volterra model is developed for the Doppler Effect caused by the source’s motion signals (position and velocity) on the reference signal and its propagation through an infinite hydroacoustic medium. Such a process can be described by a strictly feed-through nonlinear operator in the Volterra framework. The importance of such a feed-through Volterra representation of a nonlinear process lies in the fact that the feedback system inversion lemma can be employed in order to generate a feedback operator. This feedback operator is required to demodulate the reference signal and obtain various source motion signals. The feedback system can be directly derived from the feed-through Volterra representation of the modulation process. Indeed, the feed-forward branch of this system is the inverse (approximate or exact) of the linear part of the Volterra representation, whilst the feedback branch is the purely nonlinear part of the Volterra representation. Only the linear part of the feed-through Volterra operator needs to be inverted for the construction of the signal processing system. This Doppler Effect mitigation technique is applied to a high-speed, high-frequency (260-375 kHz), hydroacoustic modem developed at Florida Atlantic University in collaboration with Edgetech. The Volterra kernels are derived based on field experiments performed with sinusoidal tones as reference signals.

Keywords: HERMES, hydroacoustic, modem, Doppler, Volterra
1. INTRODUCTION

Doppler spread arises commonly in underwater acoustic communication since it is usually employed to establish information transmission between two stations, which commonly experience relative motion. Relative motion between a source and receiver results in a Doppler shifted communication signal whose distortion is proportional to the ratio of propagation speed to relative platform speed. In the underwater acoustic channel the propagation speed is low relative to potential source receiver velocities leading to time-scale changes which are significantly greater than those encountered at radio frequencies. The magnitude of the time compression or expansion is .1% for slow moving systems but can potentially reach 1% when two fast moving underwater vehicles attempt to communicate.

The effects vary with time when accelerations due to speed or course changes are present. When both source and receiver are in motion the resulting acceleration may exceed 1 m/s². While Unmanned Underwater Vehicles (UUV) may operate at high speeds, wave-following buoys are subject to accelerations that may reach 4 m/s² at sea state 6 [1]. The effects of large time-scale changes will adversely affect the throughput of high-rate, bandwidth-efficient communication link unless tracked and removed.

2. FORMULATION OF THE PROBLEM

In the context of this work, a new Doppler Effect mitigation technique is applied to a high-speed, high-frequency (260-375 kHz), hydroacoustic modem developed at Florida Atlantic University in collaboration with Edgetech [2-5]. A message transmitted by HS-HERMES includes a trigger signal that allows several functions to be set up, prior to the body of the message that is contained in the data frame. Specifically, the trigger signal, \( s_{\text{trig}}(t) \), is generated as superposition of two components. The first is a single tone at \( f_{\text{cw}}(t) \) = 375kHz; the second is a chirp signal of 26kHz bandwidth centered at carrier frequency of 360kHz. The monochromatic component at 375kHz is used to track the Doppler shift. The signal component transmitted by the modem has the following form in phasor notation,

\[
s(t) = \exp(j2\pi f_{\text{cw}}t).
\] (1)

Without significant loss of generality, for such a narrow-band transmission the channel may be assumed to have no attenuation and to be noiseless. In effect, the received signal will obtain the following form,

\[
r(t) = \exp(j2\pi f_r(t)[t-t_d(t)]).
\] (2)

In (2), \( f_r \) is the frequency of the signal at the receiver and \( t_d \) is the signal transmission delay through the channel. For frequency \( f_r \) the following equation holds,

\[
f_r(t) = \frac{c-v(t-t_d)}{\lambda}, \lambda = \frac{c}{f_{cw}}.
\] (3)
In (3), \(c\) stands for the speed of sound in the water (approximately 1500 m/s in seawater). We assume that \(c\) is constant. Estimation of the relative velocity \(v\) between the transmitter and the receiver platform is the objective of this work. Based on (3), the convention for relative velocity \(v\) is to be assumed positive when the transmitter and receiver are moving away from each other. For transmission delay \(t_d\), the following equation holds,

\[
t_d(t) = \frac{L(t-t_d)}{c-v(t-t_d)}.
\]  

\((4)\)

\(L\) is the direct relative distance between the transmitter and the receiver. Due to the use of high carrier frequency, no multipath will be assumed. \(L\) is related to the relative velocity \(v\) according to the following,

\[
L(t) = L_0 + \int_0^t v(\xi) d\xi.
\]

\((5)\)

\(L_0\) stands for the initial distance between transmitter and receiver at \(t = 0\). The phase shift introduced in the signal due to the transmission delay can be calculated according to the following equation,

\[
\theta(t) = 2\pi f_r (t) t_d(t) = \frac{2\pi}{\lambda} L(t-t_d).
\]

\((6)\)

Using (3) and (6), (2) can be rewritten as follows,

\[
r(t) = \exp\left(j2\pi f_r (t)t - j\theta(t)\right) = \exp\left(j\frac{2\pi}{\lambda} \left[ ct - L_0 - v(t-t_d) t - \int_0^{t-t_d} v(\xi) d\xi \right]\right).
\]

\((7)\)

From the above it is readily seen that if relative velocity \(v\) is zero no Doppler shift arises. That is, Doppler shift is attributed exclusively to the relative motion between the transmitter and the receiver.

3. A NONLINEAR DOPPLER VELOCITY ESTIMATION ALGORITHM

Based on the Doppler model derived in the previous section, a nonlinear algorithm is proposed that delivers continuous time estimates of the relative velocity \(v\) continuously by exploiting the single-tone of the preamble signal. The algorithm is of the non-coherent type so that no external timing recovery is needed.

Before the method is presented, it is convenient to convert the time reference to receiver in (7). Dropping \(t_d\) in the expression for the received signal in (7) comes without loss of generality. Indeed, timing at the receiver will be lagging by \(t_d\). The inclusion in the expression is therefore redundant. \(t_d\) is expected to take small values relative to the time constants of relative motion between transmitter and receiver. For example, for \(L = 100\text{m}\) and \(v = 0\), transmission delay values in the vicinity of 150 ms will arise. In effect, (7) can be rewritten as follows,
The time derivative of (8) is,

\[ \dot{r}(t) = \frac{d}{dt} \exp(j\phi(t)) = \phi(t) \exp(j\phi(t)) = j \frac{2\pi}{\lambda} r(t) \cdot \left[ c - 2v(t) - \dot{v}(t) r \right]. \] (9)

For convenience, the following purely real signal \( a(t) \) is introduced,

\[ a(t) = c - 2v(t) - \dot{v}(t) = \frac{\lambda}{2\pi} \phi(t) \] (10)

Assuming that \( a(t) \) is real, the following identities can be readily verified,

\[ \dot{r}(t) = j \frac{2\pi}{\lambda} a(t) r(t) \Rightarrow \dot{r}(t) r(t) = j \frac{2\pi}{\lambda} a(t) r^2(t), \] (11)

\[ \text{Im}(\dot{r}(t)) \cdot \text{Re}(r(t)) = \frac{2\pi}{\lambda} a(t) \cdot \left[ \text{Re}(r(t)) \right]^2, \] (12)

\[ \text{Re}(\dot{r}(t)) \cdot \text{Im}(\dot{r}(t)) = \frac{2\pi}{\lambda} a(t) \cdot \left[ \text{Im}(r(t)) \right]^2. \] (13)

The relation in equation (11) can be used in order to recover signal \( a(t) \) according to the following,

\[ a(t) = -j \frac{\lambda}{2\pi} \frac{\dot{r}(t)}{r^2(t)} = \frac{\lambda e^{-j\phi(t)}}{2\pi j} \dot{r}(t). \] (14)

Furthermore, (12) and (13) can be used as basis for the relative velocity estimation algorithm. However, (11)-(13) involve nonlinear processing of the received signal \( r(t) \). Indeed, in the Volterra-Wiener theoretical framework, the processing in (11)-(13) is a second-degree, homogeneous system [6-8]. This fact is manifested by the transfer of signal power taking place between separate frequency bands. Assuming that the power of \( r(t) \) is concentrated in a relatively narrow band (whose width depends on the bandwidth of relative velocity \( v \)) centered at frequency \( f_{cw} = 375\text{kHz} \), the signal power of the processed signal is concentrated exclusively around \( 2f_{cw} \) and zero (in (12) and (13)). Therefore, by driving the signals in (12) and (13) through a low-pass filter (LPF) with cutoff frequency at \( f_{cw} \), \( a(t) \) can be obtained. Alternatively, the same signal may be obtained by use of (14). Since \( a(t) \) is given by (10), two cases may be distinguished for establishing an estimate \( u \) of relative velocity \( v \).

The first estimation strategy is derived directly from (10) if the relative acceleration is negligible,

\[ u_i(t) = \frac{c - a(t)}{2} - \dot{v}(t) \left( \frac{2v(t)}{t} \right). \] (15)
This means that the approximation in (15) is valid for a short period of time only. However, in the general case, one needs to solve the following linear, time-varying equation in order to obtain \( u \),

\[
\dot{u}(t) + 2u(t) = c - a(t) = 2u_0(t) \Rightarrow \dot{u}(t) + 2r^{-1}u(t) = 2r^{-1}u_0(t). \tag{16}
\]

A technique to tackle the above is to multiplicatively decompose \( u \) as follows,

\[
u(t) = b(t)u_i(t) \Rightarrow \dot{u}(t) = \dot{b}u_i + bu_i. \tag{17}
\]

Then by substitution of the above in (16) one obtains the following,

\[
\left(\dot{b} + 2r^{-1}b\right)u_i = 2r^{-1}u_0 - b'\dot{u}_i. \tag{18}
\]

A solution is obtained if \( b \) is calculated as the solution to the following homogeneous ODE, so that the left-hand side of (18) is zero,

\[
\dot{b}(t) + 2r^{-1}b(t) = 0 \Rightarrow b(t) = t^2. \tag{19}
\]

Then, \( u_j \) is calculated on the basis of (18) with the left hand side set to zero,

\[
2r^{-1}u_0 - b'\dot{u}_i = 0 \Rightarrow \dot{u}_i(t) = \frac{2r^{-1}}{b(t)}u_0(t) = 2r \cdot u_0(t). \tag{20}
\]

Equation (20) must be integrated numerically in real time to generate \( u_j \). At any time instant and assuming zero initial conditions, (17) provides the velocity estimate \( u \) as follows,

\[
u(t) = \frac{c}{2} - \frac{1}{r^2} \int_0^t \xi a(\xi)d\xi, t > 0
\]

(21)

In effect the proposed methodology consists of: a) obtaining signal \( a \) by propagating the received \( r \) through a second-degree, homogeneous nonlinear Volterra system as in (11)-(13); b) numerically evaluating (21) in order to obtain the Doppler relative velocity estimate \( u \) or, alternatively, employing (15), if the associated assumption holds.

4. SIMULATION RESULTS

A model was built in Matlab/Simulink© (Fig. 1) to process the field signals. The data were collected on June 12th, 2008. A source mounted on a kayak was moving slowly, and a receiver deployed off the side of a docked vessel was recording the data [2]. The source transducer was placed at 0.5 m below the surface and was placed at a distance of 50 m. These tests were performed at slack water between 10:45 am and 11:30 am. A single 561 ms record of the 375 kHz pilot tone is used in this paper. The pilot tone was transmitted at 0.36W of acoustic power. The signal-to-noise ratio was approximately 13 dB.
The output of the system is signal $a(t)$ in (12). Some additional processing is required including the limiters and the band-pass filter before propagating the complex signal through the derivative block. This is needed in order to normalize the amplitude envelope of the received signal and generate a tone containing the phase information. Only the low-pass component of $a(t)$ is needed, therefore it is propagated through an appropriate low-pass filter.

A typical velocity estimate obtained is shown in Fig. 2. This estimate was generated using (15). Indeed, the assumption for (15) is valid since the source was slowly moving and under calm water conditions. Furthermore, the Doppler mitigation processing will be reset at the beginning of each data frame’s reception; this reduces the time interval duration $t$ in (15) to a rather small value. Finally, the spike observed around second .37 sec is due to burst noise interference.

Fig.1: Doppler velocity estimation model.

Fig. 2: Doppler velocity estimation results.

5. CONCLUSIONS

A method for estimating the Doppler relative velocity in short-range high-frequency underwater digital communications is presented. The technique is based on the second-degree homogeneous nonlinear processing of a preamble signal transmitted in the beginning of each data frame. The method is finally validated by using recorded receiver signals from an earlier field campaign. In the future, the technique will be incorporated in the actual HERMES receiver module in order to assess real-time performance, especially in terms of processing latency introduced.
REFERENCES


Abstract: A mathematical model for the impulse response of a time-varying shallow water acoustic channel is proposed. The channel is modeled as a superposition of multiple propagation paths, whose lengths and relative delays are calculated from the channel geometry. Each path is characterized by a frequency-dependent path loss, and an additional random time-variation, expressed as a multiplicative distortion. Experimental signals collected during a 2008 test in the Narragansett Bay off the coast of North America are used to assess the statistical properties of the channel. Ricean distribution, conditioned on a time-varying mean, is found to be a good match for the path gain. Measurements of the average received power over short time intervals are made to assess the channel coherence, and to test the possibility of developing a feedback-based channel state prediction for power control or adaptive modulation.

Keywords: Underwater acoustic communications, statistical channel modelling, channel state prediction.
1. INTRODUCTION

High rate acoustic communication systems that are in use today use adaptive receiver algorithms to track the time-variation of the channel. While much progress has been made on this front, none of the existing systems exploit the knowledge of channel statistics. This is simply due to the fact that such knowledge is scarce, and there are no widely accepted models for the acoustic channel fading. However, efforts are underway to address this problem, e.g. [1]-[5]. These studies rely on experimental data collected in localized environments, and suggest different analytical models that fit the experimental measurements: while some authors find Rayleigh fading to provide a good match for their measurements [2], [3], others find Ricean fading to provide a better fit [1], [5]. Most studies consider short-term statistics, while some consider long-term statistics as well. In [3], long-term variations of a narrow-band channel are modelled by a log-normal distribution, similarly as in radio channels.

In this paper, we first model the time-invariant response of a wideband acoustic channel as a superposition of multiple propagation paths, where each path is approximated by a filter of an identical shape, but a different gain. The time-varying channel model is then obtained by associating a random multiplicative gain with each propagation path. This model is based on experimental observations. The measurements indicate a time-varying local average and an additional, more rapidly varying random component in each path gain. The latter constitutes fast fading which is caused by channel variations within a window of stationarity and is found to be well-matched with a Rayleigh distribution. Conditioned on the time-varying mean, this leads to an overall Ricean fading on each path. The time-varying mean causes random variations in the average (short-term) received power, which we use as a figure of merit to assess the overall coherence of the channel. Estimates of the Doppler power spectrum suggest coherence times on the order of a second. We conclude by discussing methods for predicting the large-scale channel variation, which may be useful in the design of feedback-based power control and adaptive modulation methods.

The paper is organized as follows. The deterministic model is outlined in Sec.2 and the experimental measurements are described in Sec.3. Sec.4 is devoted to statistical channel characterization and the issues of channel state prediction. Sec.5 summarizes the conclusions and outlines the future work.

2. CHANNEL MODEL: TIME-ININVARIANT CASE

We begin by establishing a time-invariant acoustic channel model that takes into account the physical laws of propagation, namely the fact that the path loss of an acoustic channel depends not only on the transmission distance, but on the signal frequency as well.

2.1 Single Path Attenuation

The acoustic path loss experienced by a signal of frequency $f$ travelling over distance $l$ is given by

$$A(l, f) = A_0 \cdot l^k a(f)^l$$

(1)

where $A_0$ is a constant scaling factor, $k$ is the spreading factor whose value is normally between 1 and 2 (for cylindrical and spherical spreading, respectively), and $a(f)$ is the...
absorption coefficient which is often expressed in dB per kilometer using the Thorp's empirical formula [6] as 
\[
\alpha(f) = 0.11 \frac{f^2}{1+f^2} + 44 \frac{f^2}{4100 + f^2} + 2.75 \cdot 10^{-4} f^2 + 0.003
\]
where \( f \) is in kHz. Using this value, we obtain the absorption coefficient in \( m^{-1} \) as

\[
a(f) = 10^a(f/1000)/10000
\]  
(2)

where \( f \) is in Hz. Unless otherwise stated, we will assume in what follows that the distance is given in meters and the frequency in Hertz.

2.2 Multipath Propagation

In an acoustic channel, propagation occurs over multiple paths, as illustrated in Fig.1. The \( p \)-th propagation path, denoted by \( l_p, p = 0,1, ... \) acts as a low-pass filter, whose transfer function can be modelled as [7]

\[
\tilde{H}_p(f) = \frac{\Gamma_p}{\sqrt{A(l_p,f)}}
\]  
(3)

where \( \Gamma_p \) is the cumulative reflection coefficient encountered over \( n_{sp} \) surface and \( n_{bp} \) bottom reflections along the \( p \)-th path. In particular, we model the surface as ideal, with a reflection coefficient \( \gamma_s = -1 \), while each bottom reflection is modelled by a coefficient \( \gamma_b(\theta) \), which is a function of the grazing angle \( \theta \), and under idealized plane-wave propagation conditions, has a solution of the form [8]:

\[
\gamma_b(\theta) = \begin{cases} 
\rho_b \sin \theta - \rho \sqrt{(c/c_b)^2 - \cos^2 \theta}, & \cos \theta \leq c/c_b \\
\rho_b \sin \theta + \rho \sqrt{(c/c_b)^2 - \cos^2 \theta}, & \text{otherwise}
\end{cases}
\]  
(4)

where \( \rho, c \) are the density and the speed of sound in water (\( \rho = 1000 \, g/m^3 \) and \( c = 1500 \, m/s \) nominally), and \( \rho_b, c_b \) are the density and the speed of sound in bottom. We will use the values \( \rho_b = 1800 \, g/m^3 \) and \( c_b = 1300 \, m/s \) to calculate the cumulative reflection coefficient of the \( p \)-th path as \( \Gamma_p = \gamma_{sp}^{n_{sp}} \gamma_{bp}^{n_{bp}} (\theta_p) \) where \( \theta_p \) is the grazing angle associated with the \( p \)-th propagation path. Note that this angle is the same for all reflections occurring along a given path for the geometry of Fig.1. The path length \( l_p \), the angle \( \theta_p \), and the number of surface and bottom reflections \( n_{sp}, n_{bp} \) can be calculated from the system geometry.

Fig. 1: Geometry of a shallow water channel is used to calculate path lengths and angles of arrival.
2.3 Channel Response

Given the impulse response $\bar{h}_p(\tau)$ of each propagation path, the resulting multipath response is obtained as

$$\bar{h}(\tau) = \sum_p \bar{h}_p(\tau - \tau_p) \tag{5}$$

where $\tau_p = l_p/c$ is the propagation delay associated with the $p$-th path.

In general, each path is characterized by an impulse response of a different shape, and this fact prevents us from obtaining a tractable, simple channel model. To explore simplified versions, let us express the transfer function $\bar{H}_p(f)$ so as to include the dependence on the reference path, which we take as $l_0$, the direct path:

$$\bar{H}_p(f) = \frac{\Gamma_p}{\sqrt{(l_p/l_0)^k a(f)^{l_p-l_0}}} \bar{H}_0(f) \tag{6}$$

The frequency-dependence that distinguishes the $p$-th path from the reference path is embodied in the term $a(f)^{l_p-l_0}$ in the above expression. If this term could be approximated as a constant, one could model all the paths by an impulse response of the same shape, and a different gain. The absorption coefficient $a(f)$ has a value very close to 1 for a broad range of acoustic communication frequencies. This fact may justify an approximation of the form

$$a(f)^{l_p-l_0} \approx a_0^{l_p-l_0} \tag{7}$$

We examine the viability of such an approximation in Fig. 2. Shown in the figure are the absorption coefficient $a(f)$, and the factor $1/\sqrt{a(f)}$ for several values of the path length difference $x = l_p - l_0$. This result indicates that the approximation (7) may indeed be valid, especially for small path length differences. The smallest path length difference shown, 15 m, corresponds to the relative path delay of 10 ms, a value that is within the multipath spread of the majority of shallow water channels. Note also that it suffices to judge the validity of approximation only within the frequency range occupied by a given system. Hence, we propose an approximation of the channel transfer function in the form

$$\bar{H}_p(f) \approx \bar{h}_p \cdot \bar{H}_0(f), \quad \bar{h}_p = \Gamma_p/\sqrt{(l_p/l_0)^k a_0^{l_p-l_0}} \tag{8}$$

where the constant $a_0$ may be taken as the absorption factor at some frequency within the operational bandwidth, e.g. the center frequency $f_c$. Moreover, a rough approximation $a_0 = 1$ should also be fine for a system operating in the range up to a few tens of kHz.

Fig. 2: Left: absorption coefficient $a(f)$ in $m^{-1}$. Right: Verifying the approximation (7).
The above approximation yields a simplified channel, whose impulse response is given by

\[ \bar{h}(\tau) = \sum_p \bar{h}_p \cdot \bar{h}_0 (\tau - \tau_p) \]  

(9)

Corresponding to this model is an equivalent baseband channel, whose impulse response with respect to a center frequency \( f_c \) is given by

\[ \bar{c}(\tau) = \bar{h}(\tau) e^{-j2\pi f_c \tau} = \sum_p \bar{c}_p \cdot \bar{c}_0 (\tau) \]  

(10)

where \( \bar{c}_p = \bar{h}_p e^{-j2\pi f_c \tau_p} \) are the equivalent baseband path gains, and \( \bar{c}_0 (\tau) = \bar{h}_0 (\tau) e^{-j2\pi f_c \tau} \) is the reference path response in baseband. To illustrate the model, we will focus on the geometry of the experimental channel.

3. EXPERIMENTAL CHANNEL

An acoustic communications experiment called “RACE 08” (R stands for Rescheduled) was conducted by the Woods Hole Oceanographic Institution in March 2008, in the Narragansett Bay, near the coast of Rhode Island, USA. The transmitter and receiver were deployed at 4 m and 2 m above the sea-floor respectively, and separated by 400 m in 10 m deep water. The channel probing signal was a length 4095 binary sequence, BPSK modulated into the carrier of frequency \( f_c = 13 \) kHz, repeatedly transmitted at a rate \( R = 1/T = 10\text{kbps} \) within about one minute. Transmit filtering was performed to compensate for the non-ideal transducer characteristic, resulting in an approximately flat overall transfer function within the signalling bandwidth 8-18 kHz. The entire one-minute long signal was transmitted every two hours during several days of the experiment.

Fig. 3 shows the time-invariant channel model (10) and the ensemble of estimated channel responses. Clearly, the channel is highly time-varying, but nonetheless, there is a large degree of similarity between the actual channel and the model. We thus propose a time-varying model of the form

\[ c(\tau, t) = \sum_p c_p(t) \bar{c}_0 (\tau - \tau_p(t)) \]  

(11)

where \( c_p(t) \) is a randomly time-varying path gain. In general, the path delays \( \tau_p(t) \) will also be time-varying; however, for the present experiment, where both the transmitter and the receiver were fixed on the sea-floor, no variation was observed in the path delays. Hence, we will assume that \( \tau_p(t) = \tau_p \).

Fig. 3: Left: impulse response of the time-invariant channel model (10). Right: ensemble of impulse response magnitudes estimated from a 40 second long recording using the RLS algorithm.
Fig. 4 illustrates the time-variation of the path responses $c(\tau,t)$ at several delays $\tau$. In particular, shown in the figure are the reference tap (at delay 0), and the two prominent taps at delays 0.5 ms and 0.7 ms. Several important observations can be made from this figure. First, we notice that the reference tap is very stable, while the others are not. The apparent variation of the reference tap gain is most likely due to the estimation noise, i.e. it is an artefact of the estimation process, not an inherent property of the channel (although it is possible that both are present). Secondly, we note that the other reflections exhibit a high degree of time-variability, which is evident in two forms: a time-varying local average (plotted in solid line in the figure), and a more rapidly varying instantaneous deviation about the average (plotted in dots). The latter includes estimation noise as well as any inherent rapid fluctuation. The time-varying local average is “local” in the sense that it represents an average taken over a short interval of time during which channel appears stationary in the wide sense. It is important to note that different taps exhibit different rate of variation, i.e. that they cannot be characterized by the same Doppler spread.

![Fig. 4: Variation of the path gains (delays 0, 0.5 and 0.7 ms) over a 40 second interval of time. Shown are the tap gains (dotted) and their local average (solid).]

4. STATISTICAL CHARACTERIZATION

Statistical channel characterization is concerned with determining two types of functions: the probability density function (pdf) and the power spectral density (psd). In order to provide a meaningful pdf characterization, one must first ensure that the process is stationary in the strict sense. However, we have already observed that on the time scales of interest to wideband communications, the path gain process is not stationary even in the mean. We thus proceed as follows: we first subtract the time-varying local average, and then estimate the pdf of the so-obtained process. Crucial in this procedure is the selection of the time-averaging window used to obtain the local average $\tilde{c}(\tau,t)$. Specifically, one would like to select the window size such that the resulting estimate of the pdf is consistent over time. Obviously, this window needs to be shorter for those tap gains that are varying more rapidly. The exact manner in which the averaging window is selected remains rather heuristic. In Fig. 5 we show the histogram of the path gain associated with the delay of 0.5 ms (the strong reflection). The window size has been chosen as 0.1 s, resulting in an obviously good match with a Rayleigh pdf for the path gain amplitude. Hence, we conclude that, conditioned on the mean value, this tap gain can be modelled as a Ricean fading process. The mean value, however, is itself a random process, characterized by a coherence time that is on the order of 0.1 s.
Similar observations can be made for the other tap gains. Again, it is important to note that each tap will be characterized by a different time-varying local average, and a different variance. Assuming a properly chosen window that reflects the coherence time of that path, the variance appears to be constant, at least over the one-minute interval under consideration. We estimate this variance as the time-average

$$\sigma^2(\tau) = \frac{1}{I} \sum_{i=1}^{I} |c(\tau, t_i) - \bar{c}(\tau, t_i)|^2$$

where the time index $i$ spans the observation interval (40 s) in steps of $1/R = 0.1 \text{ ms}$.

At this point, we have decomposed the path gain process into the sum of two parts: a randomly time-varying mean, and a zero-mean complex Gaussian process of a given variance. Following the general radio-communications literature, one could refer to these two processes as the slow and fast fading, respectively. Each of these processes can be characterized by its pdf and psd (we already have the pdf for the fast component). In what follows, we will focus on the psd characterization of the slow component, which we find to be the more interesting one for reasons that will become apparent shortly.

The notion of “fast” and “slow” is particularly meaningful from the viewpoint of designing a communication system: fast variations occur within a signal block (data packet), while slow variations may influence the average received power in a particular signal transaction. Because the fast variations are of a somewhat unpredictable nature, they must be dealt with at the receiver side. Slow variations, however, may be more predictable, and this fact can be exploited not only at the receiver side, but also at the transmitter side - if the propagation delay allows it. The time-varying mean of the path gains will lead to a time-varying average power. Judging by the result of Fig. 4, the average power can vary by as much as several dB over a time interval of a few seconds - long enough for a feedback mechanism to be implemented between the receiver and the transmitter. Such a feedback mechanism would allow for adaptive power control at the transmitter side. In particular, the transmitter could adjust its power so that the receiver always gets the minimum needed for a pre-specified quality of reception (within the constraints of maximal power, of course). Such a strategy would yield savings in the total transmission power consumed. Alternatively, the transmitter could take advantage of the times when the channel is in a “good” state by implementing an adaptive modulation scheme. Such a strategy would yield an increase in the average bit-rate, a crucial step towards approaching the channel capacity [9].

The possibility of implementing adaptive power control or adaptive modulation relies on the ability to estimate and predict the slow channel variation. Motivated by this fact and the potential gains of adaptive modulation, we focus on characterizing the time-varying average power. (A similar analysis can be conducted for individual path gains.)
Fig. 6 shows the time-coherence properties of the average power. The average power $\bar{P}(t_i)$ is obtained by filtering the instantaneous power $P(t_i)$, which is calculated from the impulse responses at times $t_i$ as

$$P(t_i) = \sum_j |c(j\Delta\tau, t_i)|^2$$

where $j$ ranges across the delay span of the channel response in steps of $\Delta\tau = 1/R$.

The 3 dB Doppler spread of about 0.2 Hz indicates a coherence time of 5 s. Hence, it may be possible to exploit some (albeit a small) amount of coherence within a time interval of a second or less, needed to implement a feedback between the transmitter and receiver at short range. For the geometry at hand, the two-way travel time over 400 m is about 0.5 s.

The easiest way in which the transmitter can decide on the channel state is to simply assume that it is identical to the one reported by the receiver. This type of channel state estimation will obviously be sensitive to the feedback delay. An improved estimate can be obtained by predicting the channel based on the delayed feedback. To test this possibility, we look at a predictor that operates on a series of average power measurements obtained every second. The measurements, made on a logarithmic scale, are denoted by $x(i) = 10 \log_{10}(\bar{P}(t_i)/\bar{P}(t_0))$. A one-step prediction is made as

$$\hat{x}(i + 1) = \sum_{m=0}^{M-1} a_m(i)x(i - m)$$

(14)

The $M$ model parameters $\{a_m\}$ are updated based on the error $x(i + 1) - \hat{x}(i + 1)$.

Fig. 6 shows the results of prediction. A predictor of order $M = 20$ is used, and its coefficients are updated by an RLS algorithm. The actual prediction obviously yields better results when the feedback delay is not negligible with respect to the channel coherence time.
5. CONCLUSIONS

A deterministic multipath model, based on the channel geometry and the frequency-dependent path loss in a wideband acoustic system, was augmented to include a randomly time-varying gain for each propagation path. Based on the 8-18 kHz experimental measurements in a 400 m long shallow water channel, the path gains were found to obey a conditional Ricean distribution, with conditioning on a time-varying local mean in sub-second stationarity intervals. Correlation properties of the average (short-term) received power were assessed via Doppler spectrum estimation, showing coherence times on the order of a second.

Considerable variation in the average received power serves as a motivation for investigating methods that would predict the channel state in order to implement feedback-based adaptive modulation and power control. We have briefly outlined a possible channel state prediction method, which demonstrates the possibility to exploit the limited channel coherence. More sophisticated prediction models, as well as a comprehensive analysis of adaptive modulation gains, are the subject of future research.

At present, our results are limited to the detailed investigation of a short segment of experimental data, and future work will enlarge this scope. In particular, the probability density function of the time-varying local mean, and the large-scale coherence properties of the channel need to be assessed. Establishing a relationship between the so-obtained experimental models and the physical processes that occur in the ocean remains a challenging task.

REFERENCES

Regular Session I

Acoustic Imaging
PASSIVE AND ACTIVE SONAR APPLICATIONS FOR A NON-UNIFORM AND LOW COST LINEAR ARRAY

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Abstract: one of the reasons to use a non-uniform linear array (NULA) is to reduce the cost and complexity of the array, since it can have fewer sensors than an ULA system.

In this paper, authors describe the construction of a low-cost, ten-hydrophones, non-uniform linear array to be employed in passive and active sonar applications. Many experiments were conducted, under controlled conditions (large pool), oriented to characterize both individual transducers as well as the complete array. Using the array in “passive” mode, employing an omnidirectional sound source and a non-impulsive signal, beamforming and inverse filtering techniques, the authors evaluated array directivity over a wide range of frequencies (1-23 kHz) and also its capability to estimate the direction of arrival (DOA).

Furthermore, this work also focuses on searching objects and identification. Different test were performed, in order to estimate system performances when used as the receiver of an active sonar. Using sine-sweep signals, and a specific real-time software developed by the authors, several measures were conducted in the shallow and very shallow water showing the ability of the system to detect objects in a 45° wide angle.

Keywords: non-uniform linear array (NULA), sonar imaging, not-impulsive techniques, wide-band measurements.
1. INTRODUCTION

The advantages of using arrays of transducers are well known and, probably, the “increase of directivity” is the most important. In fact the arrays are designed in order to maximize energy transmitted/received in a particular angle, simultaneously minimizing energy transmitted/received in other directions. This technique is known as array beamforming.

Considering an array of hydrophones, its performance will improve by increasing the number of transducers, which obviously also means raising the cost of the system.

In this work the authors describe the construction of a low-cost, ten-hydrophones, non-uniform linear array (NULA), suitable to be employed in passive and active sonar applications.

The study was divided into two parts: first the construction and acoustic characterization of the array, then its use in sonar applications. The array was assembled in the AIDA laboratory, while the tests for characterization and sonar application have been conducted in swimming pools. The authors have also conducted further tests, in addition to those already carried out previously [1], to validate the use of a sine sweep signal in active sonar.

Different projectors were employed during the tests, an ITC 1001 (for acoustic characterization) and an ITC 5264 (for active sonar). In both case sources are driven by a test signals generated by dedicated software running on a PC. Signals captured by the hydrophones were recorded and post processed in order to estimate inverse filters for each transducer. Furthermore, using Matlab™, it was possible to compare the directivity diagrams of the array for the theoretical case and the real one.

The second part of the study focused on using the system in passive and active sonar. In first case, the array was submerged in a large pool, then small iron objects were beaten to generate sounds other it. The post-processing of recorded signals, using beamforming techniques, allowed to estimate the direction of arrival of the sound. In the case of active sonar instead, the task was to test the capability of the system (NULA and software) to visualize a bottom profile and to discover submerged objects, placed at different angle under array. To do this, a dedicated real-time software was developed by the authors on the Linux platform.

2. ARRAY SET-UP AND CONSTRUCTION

The linear array is the simplest array geometry. In such an array all elements are aligned along a straight line and typically have an uniform inter-element spacing $d$. In practice the linear array represents the discretization of a continuous line at periodic locations in space. This discretization plays an important role in the “spatial aliasing” effect - the incident wavefront is sampled at specific locations and the polar patterns produced are affected by the Nyquist sampling criterion. The distance between elements, $d$, is related (inversely proportional) to maximum frequency detectable without “spatial aliasing”. Beam efficiency and beam directivity are affected negatively due to the introduction of secondary lobes at undesired angular locations.

In order to avoid these problems and to increase the flexibility of the system, a non-uniform linear array (NULA) was considered. In this configuration elements are placed with different distances, so if the total number of elements is kept the same, it produces an array with different lengths respect to the ULA configuration. Changing the length of the array will
affect the polar pattern beamwidth and sidelobe power, providing more flexibility in overcome the limitations of uniform spaced arrays.

The array developed in this work consist of ten Aquarian Audio H2a-XLR omnidirectional hydrophones. This kind of transducer is designed to provide high quality audio performance in a low-cost device and it can be interfaced directly with professional audio microphone preamps. It offers very good sensitivity (-180dB re: 1V/mPa; +/- 4dB 20Hz-4.5KHz), low noise, and can be employed over the entire frequency range of our interest (500 Hz ÷ 40 kHz). Hydrophones are mounted on a 2 m long aluminum frame (Fig. 1) in different positions (- 0.875, -0.455, -0.250, -0.105, -0.035, +0.035, +0.105, +0.250, +0.455, +0.875 meter w.r.t. the center). Flexibility is provided by a mounting system that allows to change easily the transducers positions according to different design strategies.

![Fig. 1: Ten-hydrophone arranged in the Non-Uniform Linear Array.](image)

The receiving system is completed by a high precision microphone preamplifier with ADAT outputs (APHEX 1788) and an ADAT to MADI converter (RME AD648), which is connected to the PC.

While the PC interface would allow expansion to 64 channels, using a single preamplifier limits this to 8 channels which means that only 8 of the 10 hydrophones can be used simultaneously.

3. TEST SIGNALS

The non-impulsive signals used in this research, as already mentioned, are sine sweeps, both logarithmic and linear. Sine sweeps (chirps) have been employed since some time for audio and acoustics measurements and characterization [2][3], but in recent years their use has increased thanks to the computational capabilities of modern computers. Recent research results allow for further refinements in sine sweep measurements, in particular when dealing with the problem of measuring impulse responses and distortion at the same time and when working with systems which are neither time-invariant, nor linear.

In underwater measurements (active sonar) also other not-impulsive signals as the MLS (Maximum Length Sequence) pseudo random signal can be considered. Various papers have studied the topic [4][5][6][7], comparing MLS to sine sweep [1] and demonstrating the latter's advantages.

The main advantage of the sine sweep method is its immunity to non-linear distortion. When using MLS signals this distortion can cause severe artefacts, appearing either as artificial background noise and, or worse, as spurious peaks which can be easily confused with reflections coming from non-existent objects (false echoes). With sine sweeps, instead,
these artefacts can be separated in the time domain from the “clean” linear impulse response, provided that a linear deconvolution technique is employed (instead of the circular deconvolution, which causes the not-linear artefacts to “fold back” and contaminate the response).

In practice, this is obtained very simply by linear convolution of the recorded signal with a suitable inverse filter. As demonstrated in [2], and confirmed independently in [3], this inverse filter is simply the time-reversal of the sweep signal itself (with a frequency-dependent gain factor in case of logarithmic sweeps).

In this work sine sweep signal is used to obtain an acoustic characterization of both each hydrophone, and the entire array. It is also used as source signal in active sonar application.

4. ACOUSTIC CHARACTERIZATION

To perform the acoustic characterization the array was placed at a depth (3 m) equal to half the depth of the pool (6 m) (Fig. 2 – left). This made it possible to minimize the effects of reflections on both the bottom and the surface. The source (ITC 1001) was positioned 3 m distant in front of the array. Feeding the projectors with a long linear sine sweep from 1 to 23 kHz, the frequency response for each receiver was measured. The following Fig. 2, on the right, shows the results of this measurement. The frequency responses of the ten hydrophones are very similar except for one transducer (shown in red) that has a higher attenuation at frequencies over 15 kHz.

![Fig. 2: Acoustic characterization: measurements set-up (left), frequency response for every hydrophone (right).](image)

The second part of this experiment was to verify the real directivity of the array. Maintaining always the same distance between source and array (3 m), the source was placed in seven well defined positions (named m4...m10) in front, on left and on right of the array centre. Recorded signals were processed applying beamforming and inverse filtering techniques. Using processing in Matlab™ it was possible to compare the directivity diagrams of the array for the theoretical case and the estimated one at different angles of beamforming. The next figure on the left shows the result with source placed in position m6 (0.5 m to the right of array centre) at a frequency of 9 kHz. On the right is represented the directivity vs. frequency.
5. “PASSIVE MODE”

Using the same array setup some experiments were carried out to estimate the capability of the system to find the direction of arrival of the sound. At various positions noises were produced beating two iron objects. The eight tracks were recorded and processed by beamforming techniques, also applying inverse filtering for each transducer. In Fig. 4 are shown the direction of arrival for two different positions “A” (13° on the right) and “B” (35° on the left) of the source. The analysis was performed in five octave bands 1, 2, 4, 8 and 16 kHz, but the most significant results were obtained in the last three. Obviously, working in a confined environment such as a swimming pool means there will be reflections on the walls. These effects are noticeable in the figure related to position “A”, where, at low frequencies (1 kHz band), a broad lobe is present at around -40°, caused by reflections on the lateral wall. This “false images” disappear at increasing frequency because of the increase in directivity of the array.

Fig. 4: DOA estimated: position “A” (left), position “B” (right)
6. “ACTIVE MODE”

A series of tests were also performed in order to verify the performance of a linear sine sweep signal and to analyse the array behaviour when used as the receiver of an active sonar.

A first measurement campaign was focused on validating the use of a sine sweep signal in active sonar. The authors conducted several tests, working first in a pool and then in a lake, at different depths of 2.2 m and 6.0 m respectively. Two hydrophones ITC 5264 equipped with parabolic reflectors were employed as transmitter and receiver respectively. The aim of the experiment was to identify a submerged target, a 0.35 m height iron box, using a 1 second long linear sine sweep signal, with a frequency increasing from 2.0 kHz to 42.0 kHz. The following Fig. 5 show the detected shape of the object placed on the bottom (pool on the left and lake on the right).

![Fig. 5: Shape of object placed on the pool bottom (left) and lake bottom (right).](image)

It is important to note that the image taken in the lake is well defined, even in presence of high background noise (boat engine noise). This confirms that the sine sweep technique yields a high Signal-to-Noise ratio. So as already explained by the authors in previous paper [1], high immunity to external noise, coupled with the very fast and easy processing required for the deconvolution of the impulse responses, make linear the sine sweep signal a very appealing choice for underwater measurements.

The purpose of the second measurement campaign, conducted only in the pool, was to test both the array system and the real-time software, properly developed for this application. In this experiment the receiver was the non-uniform linear array.

Transmitter and receivers were placed on a specially constructed raft, and targets were pulled, with uniform speed, at various angles underneath the system (Fig. 6).

The targets used were a small iron box and a human diver. Different frequency ranges were used with the upper frequency limited to 16 kHz, and using sweep durations of 0.5 or 1.0 second. The following figures show the real-time software output in two different experiments (the box and the diver), for each are depicted both the shape of the object (left) and energy reflected (right) at three different angles of beamfoming, 0° (vertical beam), 15° left and 15° right.
7. CONCLUSIONS

In this work the authors described the construction of a low-cost, ten-hydrophones, non-uniform linear array (NULA). Different tests to investigate directivity of the array were conducted and results confirmed a good agreement between the estimated values and theoretical ones. Furthermore, other experiments showed the capability of the system to detect real angle of incoming sound (DOA), especially at medium-high frequencies, where array directivity is greater. Also in this case there is perfect agreement between real and estimated arrival angle, and efficiency and accuracy increase using the inverse filtering technique. With respect to active sonar applications, tests based on sine sweep signals have shown that the use of such a system allows the identification of submerged objects, also placed in a wide angle under the array.
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REFERENCES


MINIMUM VARIANCE ADAPTIVE BEAMFORMING APPLIED TO A CIRCULAR SONAR ARRAY

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Abstract:

The minimum variance (MV) beamformer, also known as the Capon or minimum variance distortionless response (MVDR) beamformer, uses the recorded wavefield to compute a set of optimal weights to be applied to each sensor, before coherently adding the sensor outputs. The weights are chosen such that the variance of the output is minimized while maintaining unit gain in the view direction. The MV beamformer offers improved resolution and image quality compared to the conventional delay-and-sum (DAS) beamformer.

The MV beamformer was originally introduced for passive systems. When adapting the MV beamformer to active sonar, sub-array averaging is necessary in order to avoid signal cancellation of coherent signals, and to reduce the sensitivity to errors in the wavefield parameters. Sub-array averaging is a technique developed for flat, uniformly sampled arrays, and it is not immediately evident that this will give satisfactory results on a non-flat array. In this work, we have successfully implemented the MV beamformer on an active 1D circular sonar array suitable for fishery sonar. We demonstrate, through simulations, that it is feasible to apply the MV beamformer with sub-array averaging to a circular array. Our results have been verified using experimental data from the Simrad SX90 fishery sonar.

Keywords: Sonar imaging, Beamforming, Adaptive beamforming, Capon, Minimum variance, High resolution beamforming, MVDR
1. INTRODUCTION

The standard beamforming method in sonar imaging is delay-and-sum (DAS). Although adaptive beamforming methods have been extensively studied in recent years [1, 4, 5, 6], most of these studies assume uniformly sampled, linear arrays. In many scenarios such as surveillance and fishery sonar, linear arrays are not suitable because they do not provide a 360 degree field of view [2]. We have studied the performance of a particular adaptive beamformer, the minimum variance (MV) beamformer, also known as Capon or minimum variance distortionless response (MVDR). Through simulations, we have applied the MV beamformer to an active 1D circular array suitable for fishery sonar. Our results have been verified using data from the cylindrical Simrad SX90 sonar. In this case, we have coherently added the data from each horizontal row in order to synthesize a 1D circular array prior to MV beamforming [3].

The key to the MV beamformer is that it adapts to the data, allowing large sidelobes in areas where there is little energy, while suppressing interfering signals from off-axis directions [4]. This results in potentially improved image quality, with improved resolution and less reverberations and cluttering effects. Although the MV beamformer shows considerable advantages compared to the DAS, it suffers from two potential drawbacks; sensitivity to errors in the assumed wavefield parameters, and difficulty in handling coherent echoes. In the next section we describe the MV beamformer and explain how diagonal loading and sub-array averaging [4, 5, 6] are used to address these challenges. In Section 3, we present our results from simulations and experimental data. Finally, we give conclusions and discuss our findings in Section 4.

2. METHODS

Given an array of $M$ elements, the output of a general beamformer may be written as

$$z[n] = \sum_{m=0}^{M-1} w_m[n] x_m[n - \Delta_m],$$

where $x_m[n]$ is time sample $n$ from element $m$ and $w_m[n]$ is the weight applied to channel $m$ at time $n$. $\Delta_m$ is the time delay applied to the output from channel $m$ in order to focus the beam in a given direction. In conventional, non-adaptive beamformers, these weights are pre-defined. Using uniform weights is one option, but a more common approach is to use a window function such as a Hanning or Kaiser window to control the sidelobe levels. In this work, a Hanning window is used. In adaptive beamforming, the recorded energy field is used to adaptively compute the sensor weights. Eq. 1 may be written in vector form as

$$z[n] = w[n]^H X[n],$$

where $w$ is an $M \times 1$ vector containing the sensor weights and $X$ is an $M \times 1$ vector containing the recorded and delayed measurements from each sensor. $T$ and $H$ denote the transpose and the transpose conjugate operators, respectively.
The MV beamformer computes the aperture weights such that the variance of the output is minimized while maintaining unit gain in the direction of interest. The variance of $z[n]$ may be written as

$$E\{[z[n]]^2\} = w[n]^H R[n] w[n],$$  

(4)

where $R$ is the spatial covariance matrix defined as

$$R[n] = E\{X[n]X[n]^H\}.$$  

(5)

In practice we do not know the expectation of $X, E[X]$, so $R$ is replaced by its estimate, $\hat{R}$. The aperture weights are computed by solving the optimization problem:

$$\min_w w[n]^H \hat{R}[n] w[n] \quad \text{Subject to} \quad w[n]^H a = 1,$$

(6)

where $a$ is the steering vector. Since the data has already been delayed in Eq. 3, $a$ is simply a vector of ones. Eq. 6 has an analytical solution given by [7]

$$w[n] = \frac{\hat{R}[n]^{-1} a}{a^H \hat{R}[n]^{-1} a}.$$  

(7)

When computing the covariance matrix, spatial averaging is applied to ensure that $R$ is invertible. More importantly, spatial averaging has previously been presented as a way to modify the MV beamformer to handle coherent echoes [4, 6]. This is crucial in an active sonar environment. Spatial averaging, or sub-array averaging, is applied by dividing the array of sensors into overlapping sub-arrays of length $L$. The covariance matrix is computed for each sub-array, and the average covariance matrix of dimension $LxL$ is computed. For an $M$-element array and sub-array length $L$, averaging over $2K+1$ temporal samples, we have

$$\hat{R}[n] = \frac{1}{(2K+1)(M-L+1)} \sum_{k=-K}^{K} \sum_{l=0}^{M-L} \tilde{X}_i[n-k] \tilde{X}_i^H[n-k].$$  

(8)

In this work, no temporal averaging is used, so $K=0$. In Eq. 8, $\tilde{X}_i[n]$ is defined as

$$\tilde{X}_i[n] = \{x_{i}[n] x_{i+1}[n] ... x_{i+L_1-1}[n]\}^T.$$  

(9)

The sub-array averaging method was developed for uniformly sampled, linear arrays. This technique is not necessarily suitable for application directly to a non-flat array since the steering vector does not display the desired Vandermonde structure [8]. Davis [9] developed a pre-processing technique to overcome this obstacle. Our results indicate that for practical purposes and in our particular scenario, this pre-processing step is not necessary in order to obtain satisfactory results. Preliminary results indicate that a spatial re-sampling step prior to spatial averaging improves the resolvability of two point reflectors by 1-2 dB. A full investigation of the implications of non-flat array geometries is left for future work.

Diagonal loading, i.e. adding a small value, $\varepsilon$, to the diagonal of the covariance matrix as described in [4], is used to make the MV beamformer more robust against errors in the assumed wavefield parameters. We have chosen $\varepsilon = tr\{R[n]\} / 0.05 / L$, where $tr\{\}$ is the trace operator.
3. RESULTS

3.1. Simulations

Using a simulated environment developed by the Norwegian Defence Research Establishment (FFI), we have implemented and compared the performance of the DAS and MV beamformers by imaging point reflectors at a range of 100 m. A 1D circular array of diameter 37 cm consisting of 32 elements was used. The transmitted pulse was a 32 ms linear FM chirp with center frequency 27 kHz and a bandwidth of 1000 Hz. On reception, 12 elements at a time were used for beamforming. 288 beams were created, covering 360 degrees. A sound propagation speed in water of 1500 m/s was assumed.

In Fig. 1, to the left, the steered response from one point reflector is shown. The MV beamformer displays a much narrower mainlobe than the DAS beamformer, and the sidelobe levels are comparable even when using a Hanning window to reduce the sidelobe levels of the DAS beamformer. As the sub-array length increases, the mainlobe width decreases, improving the potential system resolution. To the right, two closely spaced point reflectors are imaged. This is a challenging scenario for the MV beamformer since the echoes are highly correlated. Although the point reflectors are too closely spaced to be resolved by DAS, the MV beamformer with a suitable sub-array length is able to separate the points by up to 19 dB ($L=7$). As the sub-array length is decreased, the performance of the MV beamformer tends towards that of the DAS. In fact, for $L=1$, the performance of the MV beamformer equals that of the DAS.

![Fig. 1: Left: steered response for a single point reflector is shown. The DAS beamformer with a Hanning window is compared to the MV with sub-array lengths 4, 6 and 7. Right: two closely spaced point targets. Although the targets are too closely spaced to be resolved by DAS, the MV with a suitable sub-array length clearly resolves the two targets.](image)

3.2. Experimental data

We have verified our results using experimental data from the Simrad SX90 sonar [10]. This is a 256 element 2-D cylindrical array with a diameter of 38.2 cm and a height of 56.1 cm. Since the physical array is two-dimensional and cylindrical, data from each row of the array was added coherently to synthesize a 1D array, before beamforming. In the experiment, a corner reflector was placed in the water column at 5 m depth and a range of approximately 104 m. The transmitted pulse was a 32 ms linear FM chirp with 700 Hz bandwidth and a center frequency of 27 kHz. As in the simulations, 12 elements at a time were used to create...
288 beams covering 360 degrees. Fig. 2 shows the resulting images, in dB, when using the DAS beamformer with a Hanning window (left), the MV beamformer with $L=5$ (middle), and the MV beamformer with $L=6$ (right). The top three images show a 90-degree sector of the 360 degree image, while the bottom images show a smaller region around the corner reflector. The size of the imaged reflector in the azimuth direction, defined by the -6 dB value, decreases from about 29.1 m for DAS, to 11.32 m for MV with $L=5$, and finally 6.8 m for MV with $L=6$. Data from 3 depth samples were averaged to find these numbers.

Fig. 2: Results using data from the Simrad SX90. The top images show a 90 degree sector of the image, and in the bottom images we have zoomed in on the main reflector. Left: DAS with a Hanning window. Middle: MV with $L=5$. Right: MV with $L=6$. The strong reflection is caused by a small corner reflector at a range of 104 m and a depth of 5 m. All images have been normalized by the peak value in the image.

4. DISCUSSION AND CONCLUSIONS

Results from simulations show that the MV beamformer outperforms the DAS beamformer with respect to mainlobe width as well as the ability to resolve two closely spaced objects. The MV beamformer also displays promising results with respect to resolution, when applied to experimental data from the Simrad SX90 fishery sonar.

Although the gain of using the MV beamformer instead of the DAS beamformer is not as large as for a uniformly sampled linear array, the MV beamformer offers a considerably narrower mainlobe width and comparable or lower sidelobe levels. A key step when adapting the MV beamformer to an active sonar system is sub-array averaging. Our results indicate that it is feasible to apply sub-array averaging directly to a 1D circular array and still get satisfactory results. The effect of applying sub-array averaging without taking into account the non-flat array geometry will depend on several parameters such as the bandwidth and the curvature and dimensions of the array. A full investigation of these effects is left for future work.

The length of the sub-arrays, $L$, should be chosen with care. A large $L$ gives a larger covariance matrix and therefore more degrees of freedom, resulting in better suppression of interfering signals and improved resolution. However, long sub-arrays also imply less spatial averaging and thus a less robust solution prone to amplitude loss caused by signal
cancellation. Our results indicate that $L = N/2$, where $N$ is the number of elements used for beamforming, is a suitable value.

Finally, it should be noted that while the complexity of the conventional DAS beamformer is in the order of $O(M)$, $M$ being the number of elements in the array, the complexity of the MV beamformer is in the order of $O(L^3)$.

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**REFERENCES**


SIMULATION AND EXPERIMENT ON CONVERGENCE CHARACTERISTICS OF SMALL UNDERWATER ACOUSTIC LENS

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Abstract: The authors have been researching about small acoustical lens. Final goal of this study is development of designing method of the acoustic lens for underwater acoustics and/or ultrasonic diagnosis equipment.

Two dimensional finite-differentials time-domain (2D-FDTD) calculation method for underwater acoustic lens is studied. In real applications, three dimensional FDTD method is required for the precise estimation because objects should be imaged have three dimensional shapes. Common 3D-FDTD, however, with orthogonal coordinates has disadvantage on large memory requirement and long calculation time. In order to overcome this disadvantage, simple 2D-FDTD calculation based on symmetry is proposed.

In this paper, basic algorism of the simple symmetrical 3D-FDTD with cylindrical coordinate and experimental results is described. The proposed 2D-FDTD calculated the convergence sound field of acoustic lens of 40mm in diameter using the same physical parameters with measurement of 2MHz. Simulation results agreed well with real sound field.

Keywords: small acoustical lens, 3D-FDTD, convergence sound field
1. INTRODUCTION

The authors have been researching about small acoustical lens. Final goal of this study is development of designing method of the acoustic lens for underwater acoustics and/or ultrasonic diagnosis equipment.

The many ultrasonic equipments are using widely in not only high–frequency imaging system with interluminal or catheter transducer[1] but also in HIFU[2]. In imaging applications, acoustic lens have a possibility for improving the resolution of ultrasonic medical probe. It, also, is most important element to increase the ultrasonic power in tissue for therapy. In this paper, a small acoustical lens of ultrasonic catheter type probe is described. The diameter of catheter type probe has to be less than 10mm in medical fields. Because of preliminarily study, a lens size in this study is scaled up about 4 times of real size.

There are many papers on focused sound filed, focused transducers and acoustic lens[3-9]. Fold[9] reported high resolution underwater imaging system with acoustic lens in sea[10]. The authors are also studying about acoustic lens for real time underwater imaging system[11].

A two dimensional finite-differentials time-domain (2D-FDTD) calculation method is commonly used for the characteristics estimation of acoustic lens. A three dimensional (3D) FDTD[12] is, however, required for the precise estimation in real imaging system, because objects be imaged have three dimensional shapes. A common 3D-FDTD with orthogonal coordinates has disadvantage on large memory requirement and long calculation time. In order to overcome this disadvantage, a simple 2D-FDTD calculation based on symmetry is proposed in this paper. The proposed 2D-FDTD calculates the sound field of the acoustic lens using the same physical parameters with measurement in water bath. A virtual spherical sound source whose amplitude distribution is equal to the sound propagation field of real sound source is also used for reduction of calculation.

2. 2D- FDTD WITH SYMMETRICAL CYLINDRICAL COORDINATES

In general, the calculated sound fields by 2D-FDTD do not have good agreement with that of actual 3D lens. A common 3D-FDTD of orthogonal coordinates, however, requires large memory and long calculation time. In order to overcome this disadvantage, simple 2D-FDTD calculation based on symmetry of cylinder is proposed in this paper. The basic equations of FDTD of cylindrical coordinates (z, r, φ) that includes attenuation factor η are obtained using the equation of motion and the equation of continuity in the weak loss medium,

\[ -\left( \frac{1}{\rho c^2} \right) \frac{\partial p}{\partial t} = \frac{1}{r} \frac{\partial}{\partial r} \left( r \frac{\partial v_r}{\partial r} \right) + \frac{\partial v_z}{\partial z} \]  \hspace{1cm} (1)

\[ -\rho \frac{\partial v_r}{\partial t} = \frac{\partial p}{\partial r} + \eta v_r \]  \hspace{1cm} (2)

\[ -\rho \frac{\partial v_z}{\partial t} = \frac{\partial p}{\partial z} + \eta v_z \]  \hspace{1cm} (3)

where \( p \) is sound pressure, \( v \) is the particle velocity, \( \rho \) is the density and \( t \) is time. The second part of the right hand side in eq.(2) to eq.(3) show an attenuation of the medium caused by absorption. An acoustic lens used in this study is symmetrical with z-axis, because two
surface of lens are a plane and an aspherical, respectively. If the plane wave or symmetrical wave incidents parallel to \(z\)-axis, the propagation sound field is symmetrical with \(z\) axis too.

The finite differential equations are obtained as the function of \(t\). Space and time are quantized as \(\Delta r = i \Delta r\) and \(\Delta z = j \Delta z\) and \(t = n \Delta t\). For simplification, \(\Delta r = \Delta z\) in this paper. New values of the pressure and the particle velocity are calculated from the solutions of discrete equations with the previous pressure and the particle velocity in FDTD.

\[
p^n(i, j) = p^{n-1}(i, j) - \rho c^2 \frac{\Delta t}{\Delta r} \left[ v_{r, i}^{-\frac{1}{2}} \left(i + \frac{1}{2}, j\right) - v_{r, i - \frac{1}{2}}^{-\frac{1}{2}} \left(i - \frac{1}{2}, j\right) \right]
\]

\[
- \rho c^2 \frac{\Delta t}{\Delta z} \left[ v_{z, j}^{-\frac{1}{2}} \left(i, j + \frac{1}{2}\right) - v_{z, j - \frac{1}{2}}^{-\frac{1}{2}} \left(i, j - \frac{1}{2}\right) \right]
\]

\[
v_{r, i + \frac{1}{2}}^{-\frac{1}{2}} \left(i + \frac{1}{2}, j\right) = v_{r, i + \frac{1}{2}}^{-\frac{1}{2}} \left(i + \frac{1}{2}, j\right) - \frac{\Delta t}{\rho \Delta r} \left[ p^n(i, j) - p^n(i + 1, j) \right]
\]

\[
v_{z, j}^{-\frac{1}{2}} \left(i, j + \frac{1}{2}\right) = v_{z, j}^{-\frac{1}{2}} \left(i, j + \frac{1}{2}\right) - \frac{\Delta t}{\rho \Delta z} \left[ p^n(i, j) - p^n(i, j + 1) \right]
\]

On the \(z\)-axis, the next equations is used for sound pressure,

\[
p_z^n(0, j) = p_z^{n-1}(0, j) - \rho c^2 \Delta t \times \frac{4}{\Delta r} v_{r, i}^{-\frac{1}{2}} \left(i + \frac{1}{2}, j\right)
\]

\[
+ \frac{1}{\Delta z} \left[ v_{z, j}^{-\frac{1}{2}} \left(0, j + \frac{1}{2}\right) - v_{z, j - \frac{1}{2}}^{-\frac{1}{2}} \left(0, j - \frac{1}{2}\right) \right]
\]

In the FDTD calculation, a virtual spherical sound source whose amplitude distribution is equal to the sound propagation field of real sound source was used for reduction of calculation. This technique is well known to reduce the calculation time.\(^9\) The distance between virtual sound source and the first plane of lens is 0.04\([m]\). The shape of virtual sound source is assumed to be sphere of 0.66\([m]\) in radius and the center is the same position of real sound source. Propagation sound amplitude in water bath is measured in the plane normal to the \(z\)-axis at 0.66\([m]\) from sound source. Amplitude of virtual sound is converted from measured amplitude in plane using propagation loss of difference sound path in water. Incremental step size \(\Delta r\) in calculation area is 37.5\([\mu m]\). The sound velocity of water is calculated by Greenspan’s equation\(^{13}\) at 18.0 deg. \([\text{C}]\), and sound velocity of lens was reported in previous work\(^{11}\). Normalized sound field in \(x\)-\(y\) direction is shown in Fig.1.

![Fig.1 Calculated sound pressure distribution at the focal point using 2D-FDTD.](image-url)
3. UNDERWATER EXPERIMENT IN WATER BATH

The lightly focused lens is originally designed by optical-lens designing software base on the ray theory. The first surface of lens is plane and the second surface has aspherical curve. Material of lens is acrylic resin and acoustic refraction index of it is 0.53. The effective diameter is 0.04[m] and focal length is 0.29[m] from the first plane of lens. Figure 2 shows the water bath of 1x2x1[m] to measure the convergence field of acoustical lens. The lens is fixed by lens holder with anti-diffraction membrane with sound absorber. The distance between sound source and acoustic lens is 0.7[m]. A 10 cycle burst-pulse of 2.0[MHz] is radiated from the transmitter whose diameter is 0.01[m]. Temperature in water bath is kept 37deg.[C] constantly. The sound pressure is measure by hydrophone using automatic controlled xyz-stage. Incremental step size is 1x10^{-3}, 1x10^{-3}, 1x10^{-3} [m] in x,y,z direction, respectively.

4. RESULTS

Normalized on-axis sound pressures are shown in Fig.3 for measurement and analysis. It is clearly shown that measured minimum points agree well with analysis results. Focal point for measurement and analysis is 0.328[m] and 0.320[m], respectively. Transverse beam patterns at the focal point are shown in Fig.4 for measurement and analysis. Measured -3dB beam width is 6.5x10^{-3}[m] and is almost the same as analysis of 6 x10^{-3}[m]. Focal point and beam width in both cases are shown in Table 1. Sound pressure distribution in z-y direction is shown in Fig. 8. Both patterns are normalized by sound pressure at focal point. It is clearly shown that measured pattern agree well with analysis results. Figure 5 shows the sound pressure distribution in x-y direction at focal point. Measured pattern also agree well with analysis results in this case. These figures show that the proposed 2D-FDTD simulation with virtual sound source calculate precise sound pressure field of acoustic lens at normal incident.

<table>
<thead>
<tr>
<th></th>
<th>Measurement</th>
<th>Analysis</th>
</tr>
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<tbody>
<tr>
<td>Focal point</td>
<td>0.328</td>
<td>0.320</td>
</tr>
<tr>
<td>Beam width</td>
<td>0.0065</td>
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Tabl.1 Focal point and beam width
Fig. 3  Normalized on-axis sound pressures for measurement and analysis

Fig. 4  Normalized sound pressures in x-y direction at focal points
Fig.5 Measured sound pressure in x-y direction at the focal point.

5. CONCLUSIONS

The sound field of small ultrasonic probe with an acoustical lens for diagnosis and/or therapy is calculated. The simple 2D-FDTD in cylindrical coordination based on the symmetrical property is proposed to reduce calculation time. A virtual spherical sound source is also used for reduction of calculation. Acoustic lens for small ultrasonic prove is designed and fabricated with acrylic resin. Because of preliminarily study, a lens size in this study is scaled up about 4 times of real size.

Convergence sound field is measured in constant temperature water bath. Measurement results have good agreement with 2D-FDTD analysis. This shows the validity of FDTD calculation method.

In future work, we will research incident angle dependency of lens based on the 3D-FDTD in Cartesian coordinate system with parallel computation. The centre frequency of pulse, also, would be increased and lens size would be reduced.

6. ACKNOWLEDGMENTS

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REFERENCES

MINIMUM VARIANCE ADAPTIVE BEAMFORMING IN ACTIVE SONAR IMAGING

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Abstract: The delay-and-sum (DAS) beamformer is commonly used in active sonar imaging. The large sidelobes of the DAS beamformer can be suppressed using aperture shading. This gives lower sidelobe levels at the cost of a wider mainlobe. The minimum variance (MV) beamformer uses the received data to adjust the weights of each receiving element to produce the least total output power while maintaining unit gain in the view direction. This suppresses signals from off-axis directions and allows large sidelobes in directions with no signal. We have developed and applied a robust MV beamformer to the Kongsberg Maritime HISAS 1030 sonar, which is an active sonar system. The transmitted signal was a 30 kHz band centered at 100 kHz and we used one 32 element receiving array. We first applied the MV and DAS beamformers to a simulated dataset. MV beamforming of two closely spaced reflectors showed better resolution and suppression of sidelobe levels compared to DAS. On a simulated scene with speckle, highlight and shadow the MV beamformer showed better lateral edge definition at the cost of slightly lower contrast. We also demonstrate these effects of the MV adaptive beamformer on real data collected by the HUGIN autonomous underwater vehicle carrying a HISAS 1030 sonar.

Keywords: Beamforming, Adaptive beamforming, Capon, Minimum variance, MVDR, High resolution imaging, Sonar.

1. INTRODUCTION
To focus an array of sensors towards a specific direction or point in space is known as beamforming. The standard beamforming method used in sonar imaging is delay-and-sum (DAS). Control of the sidelobe level of the DAS beamformer can be achieved by using aperture shading. This results in increased contrast at the expense of resolution. An adaptive beamformer uses the recorded wavefield to compute the aperture shading. Increased resolution can be achieved since an adaptive beamformer can suppress interfering signals from off-axis directions and allow larger sidelobes in directions in which there is no received energy. Adaptive beamforming methods such as the minimum variance (MV) beamformer [2], also known as Capon or Minimum Variance Distortionless Response (MVDR) beamformer, have successfully been used in passive wideband sonar imaging [1], and more recently been adopted to active wideband sonar imaging [3,4].

Systems with active transmission cause coherent received echoes. The standard MV beamformer assumes incoherent echoes. For simple scenarios like single point targets, the echoes will be incoherent. For more complex scenarios, the returning echoes are coherent and several measures need to be taken in order to handle this. These measures are various forms of averaging and smoothing of the estimates.

In this work we have applied the MV beamformer to simulated and real data. We have shown that by addressing the problems of coherence, the performance of the MV beamformer in an active imaging system is better or comparable with that of the DAS beamformer.

In the following section, a brief introduction to the MV beamformer is given. In Section 3, the simulation and experimental results are presented and discussed. A conclusion is drawn in Section 4.

2. METHODS

For both the DAS and MV beamformers we assume an array of $M$ elements, each recording signal $x_m[n]$. The output of the beamformer, $z[n]$, is a weighed sum of the time-delayed $M$ measurements

$$z[n] = \sum_{m=0}^{M-1} w_m[n] x[n - \Delta_m] = w^H[n] X[n] \ ,$$

where $\Delta_m$ is the time delay applied to signal $m$, $w_m[n]$ is the weight applied to sensor $m$ at time $n$, $w^H[n] = [w_0[n] \cdots w_{M-1}[n]]$ and $X[n] = [x_0[n - \Delta_0] \cdots x_{M-1}[n - \Delta_{M-1}]]^T$.

In the DAS beamformer the weights are independent of the received signal. We have used either a uniform or a smooth window with tapered edges. A smoothed window gives a lower sidelobe-level and a wider main lobe compared to a uniform window.

In the Capon beamformer the weights are calculated from the recorded signal by minimizing the variance (power) of $z[n]$ while maintaining unit gain at the focal point. This gives unique weights for all depths and viewing angles. The analytical solution of to the problem is given by [2]:

$$w[n] = \frac{R^{-1}[n] a}{a^H R^{-1} a} \ ,$$

where $R[n]$ is the spatial covariance matrix and $a$ is the steering vector.
In order to calculate these weights, the spatial covariance matrix has to be estimated. To estimate an invertible spatial covariance matrix from realistic signals, \( \mathbf{R}[n] \) in (2), both averaging of the observed signals in temporal and spatial domains as well as regularization is required. In the spatial domain, averaging is performed by dividing the aperture into (overlapping) subarrays and averaging the spatial covariance matrices of each subarray. The general covariance matrix estimate averaged over \( 2K+1 \) temporal samples and \( M-L+1 \) subarrays of length \( L \) is given by:

\[
\tilde{\mathbf{R}}[n] = \mathbf{I} + \tilde{\mathbf{R}}[n] = \frac{1}{(2K+1)(M-L+1)} \sum_{k=-K}^{K} \sum_{l=0}^{M-L} \mathbf{X}_l[n-k] \mathbf{X}_l^H[n-k],
\]

\( \mathbf{X}_l[n] = \left[ x_i[n-L] \cdots x_{i+L-1}[n-L] \right]^T \), \( \mathbf{I} \) is the identity matrix and \( \varepsilon \) is a constant. We have chosen this constant to be proportional to the power of the received data, i.e. \( \varepsilon = tr(\tilde{\mathbf{R}}[n]) \cdot \delta / L \), where \( tr(\cdot) \) is the trace operator and \( \delta \) is typically less than 1. A more thorough description of the algorithm, the estimation of the covariance matrix and the choice of parameters can be found in [5,6].

3. RESULTS AND DISCUSSION

We have simulated the Kongsberg Maritime HISAS 1030 sonar [7] with one transmitter and a 32 element linear receive array operating at 100 kHz. First we simulated a number of pairwise reflectors located at 50 and 70 meters range. The reflectors were separated by one meter. Fig. 1 shows the fields obtained using the DAS beamformer with rectangular and Kaiser 3.5 window shading, and the MV beamformer using a subarray length of 12 and diagonal loading with \( \delta = 0.2 \). The covariance matrix has been averaged over one wavelength in time. These parameters have been used for all MV images.

Fig. 1: Cut through focus at 50 meters (to the left) and at 70 meter (to the right) of two point reflectors separated by one meter.

At 50 meters range, we see that the DAS beamformer with a rectangular window and the MV beamformer are able to resolve the two reflectors, while DAS shaded with a Kaiser window cannot resolve them. At 70 meters range, only the Capon method can resolve the reflectors. The sidelobe level of the MV beamformer are seen to be much lower than both the DAS
beamformer with and without shading, and the width of the point response for the MV beamformer is much narrower than for the two others.

We have further simulated a scene with speckle, highlight and shadow. Simulated images from the same choice of beamformers are shown in Fig. 2. The images are normalized such that the mean intensity level is the same in the speckle region. The highlight region is in these images indicated with a white ring. Fig. 3 shows averaged cuts through the highlight and shadow regions. In the highlight, the response is averaged over a 0.6 meter wide band while in the shadow region the response from 43.5 to 45.5 meter is averaged. Fig. 4 shows averaged axial cuts through the centre of both the speckle, highlight and shadow regions. The response is averaged over a 0.6 meter wide band.

Fig. 2 shows that compared to the DAS beamformer, the MV beamformer produces images with better definition of both the highlight and the shadow region at a possible cost of a slightly lower contrast. Studying the details of these images as given in Fig. 3, the response for the MV beamformer in the highlight region has approximately 1-2 dB lower amplitude than the DAS beamformers. The true value of the highlight region should have been constant and at 15.6 dB. All the beamformers show a large variation in the amplitude response. The DAS beamformers seem to overestimate while the MV beamformer seems to underestimate the response. The width of the highlight region is overestimated for the DAS beamformers compared to the MV beamformer, the Kaiser shaded DAS beamformer producing the widest response. This result is in agreement with the simulated point reflector results given above. In the shadow region, the edges are best preserved by the MV beamformer. The filling of the shadow region is more severe for the DAS beamformer with rectangular shading compared to the two other. The axial cut given in

Fig. 2: Simulated images of speckle, highlight and shadow. DAS with rectangular shading (left), DAS with Kaiser shading (middle) and MV (right).
Fig. 3: Lateral cut through highlight (to the left) and shadow (to the right) for different choices of beamformer; DAS with rectangular shading, DAS with Kaiser shading and MV.

Fig. 4: Axial cut through speckle, highlight and shadow for different choices of beamformer; DAS with rectangular shading, DAS with Kaiser shading and MV.

Fig. 5: Images of a barge from the HISAS 1030 sonar for different choices of beamformer; DAS with rectangular shading (left), DAS with Kaiser shading (middle) and MV (right).

Fig. 4 shows that the response for the MV beamformer are close to the response of the Kaiser shaded DAS beamformer, and for most depths lower than the response for the rectangular shaded DAS beamformer.
Fig. 5 shows images from real data from HISAS 1030 [7], processed using the three different beamformers. Note that this is a sectorscan image, i.e. an image from a single ping. This is not to be confused with sidescan sonar images or synthetic aperture sonar images that are based on multiple pings. The images show a barge throwing a short shadow. As for the simulated images, the MV beamformer produces the image with the narrowest highlight region. The width of the shadow region is as wide as for the DAS beamformer with rectangular shading, but the filling of the shadow is not as severe and comparable with the DAS beamformer with Kaiser shading.

4. CONCLUSION

We have presented sonar images from simulated and real data. The images have been produced using three different choices of beamformers, DAS with rectangular shading, DAS with Kaiser shading and MV. We have shown that the MV beamformer produces edge responses that are better or comparable with a DAS beamformer without shading and a sidelobe level or filling of shadow region which is comparable to a Kaiser weighted DAS beamformer.

ACKNOWLEDGEMENTS

We kindly thank Kongsberg Maritime for providing experimental data from the HISAS 1030 sonar.

BIBLIOGRAPHY

REDUCED SCALE EXPERIMENT OF APLANATIC ACOUSTIC LENS FOR DESIGNING AMBIENT NOISE IMAGING SYSTEM

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Abstract: In our previous studies, we verified that a spherical biconcave lens with an aperture diameter of 2.0 m has a sufficient directional resolution (for example, the beam width is 1 deg at the target center frequency of 60 kHz) for realizing an ambient noise imaging (ANI) system. In this study, we designed an aplanatic lens correcting both spherical and coma aberrations with the same aperture for use in an ANI system. To confirm the directional resolution of the lens in a wide frequency band of 20-100 kHz in the original scale, we performed a reduced scale experiment of one-fifth space in a water tank. The lens, made of acrylic resin, has an aperture of 400 mm and a focal length of 500 mm. Burst pulses of 100, 200, 300, and 500 kHz, so that the frequency increased 5 times, radiated from the sound source. We measured the sound pressure after passage through the acoustic lens by moving the receiver around the image point. The sizes of –3 dB areas, whose pressure is 3 dB lower than the maximum at the image point, were smaller than those of a spherical lens, and the –3 dB areas did not overlap at the target center frequency of 300 kHz in the reduced scale. The beam patterns were expanded and contracted in proportion to the wave length. As the main lobe at 500 kHz was narrower than that at 300 kHz, we verified that this lens has a fine directional resolution over 60 kHz for use in an ANI system in the original scale.

Keywords: ambient noise imaging, aplanatic acoustic lens, directional resolution, reduced scale experiment
1. INTRODUCTION

In a traditional sonar system, target detection is hindered by ocean ambient noise, as noises distort the sound characteristics of target signals. There is a new method, developed by Buckingham et al., which views ambient noise as a sound source rather than a hindrance and which is neither passive nor active sonar [1]. This method is often called ambient noise imaging (ANI), and some experimental systems have been built that incorporate ANI. The Acoustic Daylight Ocean Noise Imaging System (ADONIS), consisting of a 3 m diameter spherical reflector with an array of 126 hydrophones attached to the focal surface, was built by Epifanio et al. [2]. Recently, 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, was built by Venugopalan et al. [3]. Both systems successfully detected silent target objects under snapping shrimp dominant noises.

An acoustic lens system could be a powerful choice for realizing ANI, because such a system would not require a large receiver array and a complex signal processing unit for two-dimensional beam forming, which could reduce the size and cost of the system. In our previous studies, we analyzed a sound pressure field focused by a single spherical biconcave lens constructed for an ANI system using the finite difference time domain (FDTD) method. Our aim was to develop a lens with a directional resolution similar to the beam width of ROMANIS, which is 1 deg at a frequency of 60 kHz. Upon analysis using the 2D or 3D FDTD method in the original scale and a reduced scale experiment in a water tank, an aperture diameter of 2.0 m with sufficient resolution was determined [4-8].

An aplanatic acoustic lens improves its convergence performance by correcting spherical and coma aberrations [9], and we expect the aplanatic lens to become the preferable choice for developing an ANI system. In this study, we designed an aplanatic lens with the same parameters as the spherical lens used in previous studies and evaluated its directional resolution. We then performed a reduced scale experiment on a scale of one-fifth of the original size of the lens in a water tank. We compared the sound pressure distribution and –3 dB area of the aplanatic lens with those of the spherical lens every 1 deg of incidence angle at the target center frequency, and the beam patterns of the aplanatic lens with those of the spherical lens at various frequencies to check the frequency dependences of directional resolution.

2. APLANATIC LENS AND EXPERIMENTAL SETUP

In this study, the aplanatic lens was designed with an aperture diameter of 2.0 m, a focal length of 2.5 m, and a center thickness of 0.05 m, which are the same as the parameters of the spherical lens used in previous studies [6-8]. Here, the refractive index is 0.562, and the speed of sound in water and in the lens is 1500 and 2670 m/s, respectively. The lens shape and ray paths obtained by Sato’s method [9] are shown in Fig. 1. This lens has a gentle aspherical curve on the entrance side but a steep curve on the exit side. The rays are concentrated on each different point at either a normal incidence or an oblique incidence of 7 deg. However, there are small aberrations in a large incidence angle, and the rays are not concentrated at all on a specific point at 15 deg. A lens, scaled down to one-fifth of the original size, with an aperture diameter of 400 mm, a focal length of 500 mm, and a center thickness of 10 mm, was made for this reduced scale experiment. The material used was acrylic resin.
The experimental arrangement of the sound source, the receiver, and the acoustic lens in the water tank is shown in Fig. 2. The water tank was 7 m wide, 9 m long, and 5 m deep. The distance between the sound source and the center of the acoustic lens was 7 m. Burst pulses at 100, 200, 300, and 500 kHz were radiated from the sound sources. Here, the frequencies are multiplied five times to correspond to the frequencies of 20, 40, 60, and 100 kHz in the original scale. The sound pressure after passage through the acoustic lens was measured by moving the receiver. The diameter of the receiver was 4 mm, and the receiver had a flat frequency response up to 800 kHz. The x-y-z stage controlled the position of the receiver within an accuracy of ±0.01 mm. The acoustic lens was arranged perpendicular to the z-axis.

3. RESULTS

The relative pressure distributions around the image point (maximum pressure point) on the x-z plane at y=0 are compared in Fig. 3 at the frequency of 300 kHz, which was the target center frequency of 60 kHz in the original scale. Whenever the incidence angle increased by 1 deg, the image point shifted to the right by about 9 mm. Generally, the 3 dB beam width of a beam pattern is used to determine the directional resolution of a sonar system. This beam width is also used to estimate the resolutions of ADONIS and ROMANIS. In this study, we used the –3 dB area, the pressure of which was 3 dB lower than the maximum at the image point, to evaluate the directional resolution of the lens, as equivalent to the 3 dB beam width. The –3 dB areas on the x-z plane at y=0 are shown in Fig. 4 in the case of the center frequency of 300 kHz. All the –3 dB areas of the aplanatic lens were smaller than those of the spherical lens. Each –3 dB area of the aplanatic lens was between about 53% and 55% of that of the spherical lens. We confirmed that the effect of correcting aberrations in the aplanatic lens improved the convergence performance. We can also see that each –3 dB area had a width of about 7 mm and that the –3 dB areas did not overlap in the experimental results of the aplanatic lens. These results verified that sufficient resolution for an ANI system could be realized using the aplanatic lens at small incidence angles.
Fig. 3: Experimental results of sound pressure distributions on the x-z plane at y=0 in the case of the center frequency of 300 kHz. The results of the spherical lens are for (a) 0 deg, (b) 1 deg, and (c) 2 deg. The results of the aplanatic lens are for (d) 0 deg, (e) 1 deg, and (f) 2 deg. The contour lines are drawn every 1 dB.

Fig. 4: Experimental results of –3 dB areas on the x-z plane at y=0 in the case of the center frequency of 300 kHz. The mark “*” is the image point (maximum pressure point) and the gray zone is the –3 dB area. The results of the spherical lens are shown in (a), and those of the aplanatic lens are shown in (b).

To check the frequency dependence of directional resolution, we charted the experimental results of beam patterns at normal incidence and various frequencies, as shown in Fig. 4. The
main lobes of the aplanatic lens were narrower than those of the spherical lens, and the side lobes of the spherical lens were larger than those of the aplanatic lens at all frequencies. These results also confirmed that the effect of correcting aberrations in the aplanatic lens improved the convergence performance. The figures show that the beam patterns were expanded and contracted in proportion to the wave length. It is likely that the –3 dB areas will not overlap at frequencies over 300 kHz, because the main lobe at 500 kHz was narrower than that at 300 kHz.

**Fig. 4**: Experimental results of beam patterns at normal incidence and various frequencies. (a) 100 kHz, (b) 200 kHz, (c) 300 kHz, and (d) 500 kHz.

4. CONCLUSION

In this study, we designed an aplanatic lens that corrects spherical and coma aberrations for an ANI system and evaluated its directional resolution in a reduced scale experiment at one-fifth of original size. The experimental results verified that the present lens has sufficient resolution for an ANI system, because the –3 dB areas did not overlap every 1 deg of incidence angle at the target center frequency of 300 kHz, which corresponded to the 60 kHz in the original scale. In addition, the main lobe at 500 kHz was narrower than that at 300 kHz.
in the beam pattern of the reduced scale. This suggests that a lens of 2.0 m aperture will have fine directional resolution over 60 kHz in the original scale. Thus, this lens has a suitable directional resolution for use in a wide frequency band when the sound source of an ANI system is the natural background noise generated by living organisms such as snapping shrimp.

Future studies are still needed to measure the sound pressure field at oblique incidences of large angles to estimate the field of view of the present lens. Tests are also needed to confirm the angle that will cause –3 dB areas to overlap. The aplanatic lens appears to have a wide field of view in light of the fact that it corrects coma aberration.

REFERENCES

DERIVING FLOW VELOCITY MEASUREMENTS AND SUSPENDED SEDIMENT FLUX USING BACKSCATTER RECORDED WITH A MULTIBEAM ECHO-SOUNDER

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Multibeam Echo-Sounders (MBES) are routinely deployed to provide high-resolution bathymetric imaging, and modern data handling and storage technologies now permit the logging of the backscatter information from the water-column. This paper describes a novel methodology that derives estimates of flow velocity across the two-dimensional MBES swath by correlating backscatter magnitudes recorded from the static deployment of the MBES.

The correlation method used is similar to the Particle Image Velocimetry techniques that have been developed to derive velocity vectors across a two-dimensional area of flow containing seed particles. The method presented herein differs in that the suspended sediment particles are smaller than the acoustic wavelength, and hence the correlation magnitude relates the similarity of areas of spatially-averaged random backscatter in the Rayleigh regime between successive pings rather than the reflections from individual particles. As such, the method can be used to quantify the dynamics of suspended sediment that track the flow structure, and builds upon our previous work that has quantified suspended sediment concentrations from the backscatter recorded with an MBES. By enabling the simultaneous measurements of suspended sediment concentration, flow velocities and bathymetric data, the MBES is demonstrated to be a versatile tool for the holistic surveying of sediment transport fluxes in aquatic environments.

Results are presented from the deployment of a RESON 7125 MBES system at the confluence of the Mississippi and Missouri rivers, USA. The data was recorded from a boat moored at-a-point over a large sand-dune. Flow structures containing suspended sediment are clearly observed, developing and advecting with flow over the dune. The velocities
derived from the MBES backscatter correlation method closely match the velocity profiles recorded with an acoustic Doppler current profiler at the same location.

**Keywords:** Sediment Transport, Flow Velocities, Multibeam, Imaging

1. **INTRODUCTION**
This paper describes a method of estimating suspended sediment flow structure velocities from the backscatter recorded using a multibeam echo-sounder (MBES). This work was undertaken as part of a partnership project between the University of Leeds and RESON Inc.(funded by the Natural Environment Research Council, UK). Methods of data analysis have been developed that derive flow velocities from tracking of areas of suspended sediment scattering volume data obtained using RESON’s 7125 MBES system. Results from the field deployment of the 7125 at the confluence of the Mississippi and Missouri rivers are presented.

The research team at Leeds University’s School of Earth and Environment has extensive experience of quantifying and modelling the links between flow and suspended sediment dynamics in a wide range of environments using a variety of innovative approaches. The potential for using MBES to identify suspended sediment structures was identified by the group whilst collecting bathymetry data in the Paraná river system along the border of Paraguay and Argentina [1]. The objective of the current project is to extend the use of the MBES, to an instrument that is simultaneously capable of providing information regarding the suspended sediment concentration and associated flow structures within the water column as well as the high-resolution bathymetry data. The aim is to develop methodologies that will enable a wide range of users to exploit a single instrument capable of providing holistic measurements of spatio-temporal bathymetry, suspended sediment concentration and flow structure velocities, and thus providing a powerful tool for the management and modelling of complex fluvial systems.

2. **METHODOLOGY**
The aim of the partnership project was to develop methods of simultaneously estimating the suspended sediment mass concentration and flow velocities from the backscatter received and recorded by the MBES system. Details of the method for estimating the concentration have been described by Simmons et. Al. [2] using the RESON 8125 MBES at the Paraná field site.

2.1. **7125 MBES DESCRIPTION**
RESON’s 7125 MBES produces a fan of beams over a two-dimensional swath of 128°. The beam-forming algorithm produces either 256 or 512 beams when operating at 400kHz or 256 beams at 200kHz. The angular spacing between the beams can be set to uniform or ‘equi-distant’ mode (for uniformly spaced bathymetry samples). The transmit and receive beam-patterns combine to give approximate 3dB beam-widths at nadir of 0.5° x 1.0° at 400kHz and 1.0° x 2.0° at 200kHz. The magnitude data are recorded along each beam with a sampling
distance of approximately 0.0214m (1500ms\(^{-1}\) sound speed). Fig. 1 shows a visual representation of the backscatter magnitude for a single ping obtained from a vessel moored at a point over the lee-side of a dune at the confluence of the Missouri and Mississippi rivers. The MBES was operated at 400kHz with 256 beams of equal angular spacing. The solid line is the bathymetry data obtained with the MBES bottom-finding algorithm. The depth of the bed at nadir is just over 7m and the data above the solid line shows the backscatter from material within the water-column and also the interference from the strong bed reflections via the side-lobes. The area of interest for concentration and flow velocity estimation is defined by the area within the arc formed by the component of the nearest bed reflection to the transducers. This area can be seen in Fig.1 where a large suspended sediment flow structure can be observed above the bed. (advecting with flow from left to right).

Fig. 1 Back-scatter magnitude of a single ping collected at the confluence of the Missouri and Mississippi. The orientation is parallel to the flow (left to right) of the and the position is above the lee-side of a sand dune bedform.

2.2. SCATTERING VOLUME

The first stage of the processing is to form an estimate of the scattering volume, \(S_v\), across the two-dimensional area of the MBES swath, which can then be tracked through the volume to estimate flow velocities. For a suspension of sediment of constant grain size and type, the linear value of \(S_v\) is proportional to the mass concentration. \(S_v\) is a function of the square of the mean back-scatter voltage per unit volume, corrected for spreading, attenuation and absorption losses and the sonar settings such as power, gain, pulse length, and time-varying gain. It is derived from the sonar equation as:

\[
S_v(r, \theta) = RL(r, \theta) - SL(\theta) - S_p \log_{10} r + 40 \log_{10} r - 10 \log_{10} V(r, \theta) + 2\alpha r
\]

(1)

where RL and SL are the receive and source level intensities in dB, \(S_p\) is the time-varying gain spreading coefficient, \(\alpha\) is the absorption (dB/m), \(r\) is the range to the sampling volume at an angle, \(\theta\), to the nadir and \(V\) is the range cell volume, defined as:

\[
V(r, \theta) \propto r \frac{\theta_{\text{Nadir}}}{\cos(\eta)} \phi
\]

(2)

where \(r\) is the range to the sampling volume, \(\phi\) is the along-track 3dB beam-width, \(\theta_{\text{Nadir}}\) is the across-track 3dB beam-width at nadir and \(\eta\) is the angle from the nadir.
2.3. SEDIMENT BACKSCATTER STATISTICS

The backscatter magnitudes for uniformly distributed spherical-scatterers are Rayleigh distributed. The standard error for n samples of a Rayleigh distribution is approximated by Eq.(3) [3]:

$$\sigma \approx \frac{V_{\text{rms}}}{2\sqrt{n}}$$

where $V_{\text{rms}}$ is the expected back-scatter voltage and n is the number of samples. Hence, for 100 samples a 5% error is expected for the mean. This equates to an error of approximately 10% in Sv. The processing routine therefore allocates, to the position of an individual sample in a beam, the mean or median of all the positions of the samples within a specified spatial radius. This process is repeated for all samples within the beams of the two-dimensional fan. Hence, a two-dimensional spatial-averaging algorithm is performed for the full swath on each individual ping.

3. FLOW VELOCITY ESTIMATION

A two-dimensional mean quadratic difference (MQD) method has been developed that estimates velocities by tracking areas of scattering volume data between successive pings. The cross-correlation method works in a similar manner to Particle Image Velocimetry (PIV). PIV is a commonly used velocity measurement technique in fluid mechanics and has become popular and successful owing to its simplicity, accuracy and low-intrusiveness. The difference of the method described herein is that PIV tracks seeded particles much larger than the wavelength of light, whereas this method tracks areas of volume reverberation from scatterers that are generally much smaller than the acoustic wavelength.

The first step in the estimation process is to convert the averaged (linear) scattering volume levels (obtained using the method outlined in section 2) from polar to Cartesian co-ordinates by performing a two-dimensional interpolation. This allows rectangular windows of data in one ping to be compared with the same-sized windows of data (at different positions) in the next ping. A value based on the MQD of the two windows is then derived. The velocity vector is then estimated by determining the window within the next ping with the lowest MQD and joining the centre of the area in the first ping with that of the window in the second. The process is then repeated across a grid of different areas in the first ping to produce a two-dimensional matrix of velocity estimates. A refinement has been made by including a gradient-ascent algorithm to find the peak correlation and thus save processing time. This was possible as the correlation peaks were found to be smooth over a large area. Cubic interpolation is performed on the samples around the peak to refine the vector estimate on a sub-pixel level. Temporal averaging is then performed with a simple moving-average for each vector component. The MQD method was chosen over cross-correlation methods to enable windows of different sizes near the edge of the swath to be compared with one another.

4. FIELD RESULTS

Fig. 1 shows the raw backscatter data of a single ping at the confluence site collected in October 2007. The data were collected with the survey vessel moored at a point, positioned over the lee-side of a sand-dune with the transducers orientated parallel to the flow. A lee-
side slope, with smaller superimposed bedforms, is clearly visible at an angle of around 7-8°. However, it was later determined (from examining the bathymetry and velocities in the swath) that the MBES was effectively mounted at an angle of around 5.1° to the vertical (possibly due to the anchoring of the vessel) giving a lee-side slope of around 2-3°.

The data were processed to give values of Sv which were then interpolated onto a Cartesian grid with 0.025m spacing. The MQD was calculated for windows of 80 x 80 samples (2m x 2m) and the velocity vectors estimated for each point. The resultant vectors were then averaged using a moving-average window of 9 pings (Ping repetition rate was 10Hz). Fig. 2 shows the mean vectors (positioned relative to the transducers at 0m, 0m) over a period of 11s. The vertical components have been multiplied by a factor of 5 and are relative to the mean vertical component across the swath.

Fig. 2 Mean flow structure velocity vectors over a period of 11s. The vertical components are relative to the mean and are multiplied by a factor of 5.

Fig. 3 shows the velocity vectors at a single frame in the sequence. The colour scale here depicts an estimate of the mass concentration (mg/l) based on the Sv values and a model of the grain-size variability. Again, the vertical components of flow have had the overall mean value subtracted and are exaggerated in scale by a factor of 5 to demonstrate their relationship to the advecting suspended sediment flow structure. A relationship can clearly be seen between the components of velocity and the structures of suspended sediment close to the bed.
5. CONCLUSIONS

This paper has demonstrated the potential of MBES systems to estimate flow structure velocities in two dimensions. The ability of MBES systems to measure high-resolution bathymetry along with estimates of concentration and velocity enables improved monitoring, modelling and management of environmental systems.

REFERENCES


ASSESSMENT OF SHALLOW WATER PERFORMANCE USING INTERFEROMETRIC SONAR COHERENCE

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Abstract: Interferometry is used for bathymetric mapping with side scan and synthetic aperture sonars. The principle is based on estimating the time difference between arrivals for two vertically displaced receivers. An important additional feature with interferometric sonar is that an estimate of the signal-to-noise ratio can be derived from the coherence between overlaid data series from the two receivers. Interferometric sonar coherence therefore allows for an effective real-time estimate of quality for side scan sonar and synthetic aperture sonar measurements.

We have conducted a series of trials in shallow waters along with Kongsberg Maritime using a HISAS 1030 interferometric sonar mounted on a HUGIN autonomous underwater vehicle. We have also developed a tool for shallow water performance assessment based on a simple ray model. In this paper we first demonstrate how the interferometric sonar coherence relates to the image quality of synthetic aperture images. We then associate the coherence from real data with results from the model in order to better understand the different signal contributions. For the shallow water environment of investigation, we identify multipath returns as the limiting factor for image quality.

We conclude that the interferometric sonar coherence yields an effective estimate of quality for interferometric side scan sonar and synthetic aperture sonar measurements. It allows for a simple real-time estimate of the imaging range and in combination with simulations provides the means required for autonomous on-the-fly change of the sonar geometries and settings for optimal performance.

Keywords: Sonar performance, cross-correlation, coherence, very shallow waters
1. BACKGROUND

Side scan sonar (SSS) and synthetic aperture sonar (SAS) are two important technologies for imaging the seafloor. In very shallow waters their performance strongly depends on the environmental conditions. A primary source for this limitation is the contamination of the returned signal by multipath returns. The identification of dominating multipaths has earlier been investigated through advanced experiments [1][2]. In this paper we combine measurements with a dedicated model description in order to relate the loss of performance to the correct source and environmental condition.

2. MEASUREMENTS

We have conducted a series of measurements in very shallow waters along with Kongsberg Maritime using a HISAS 1030 interferometric sonar [3] mounted on a HUGIN autonomous underwater vehicle. In Fig. 1 we present two sample images from the same scene with water depth around 9 m. While the first image has good quality all the way out to 150 m range, the quality of the second image is poor above roughly 55 m range. In Fig. 2 we present a measurement-based signal-to-noise ratio (SNR) of the data used to generate the two images and observe that the image degradation corresponds to a decrease of the SNR that at 55 m range crosses roughly 2 dB.

For AUV operations the capability of estimating the imaging range and if possible optimise the sonar performance can be of great importance. We will adopt the SNR of the beamformed signal as a measure of imaging potential and identify the main limiting factors to the SNR.

The measurement-based SNR is obtained from the time series of the two vertically displaced arrays of the interferometric sonar. Assuming a Gaussian distribution of both signal and noise amplitudes, the SNR is given by the maximum normalised correlation factor $\mu$, the coherence, when correlating the two time series [4]:

$$\text{SNR} = \frac{\mu}{1 - \mu}$$

3. MODEL

We have developed a ray tracing tool for modelling the SNR in shallow waters. The SNR is estimated from the ratio of the direct bottom return to the sum of all other returns, including the direct surface return and returns involving two or more reflections from bottom and surface. A simplification is adopted in that only one non-specular reflection is allowed for each ray. This significantly reduces the computational task, while preserving a reasonable accuracy as most of the energy is reflected around the specular direction. The performance of the model is indicated by the similarity between the measured and modelled SNR of Fig. 2.
Fig.1: Two SAS images taken one week apart of the same scene by HISAS 1030 on a HUGIN AUV using identical settings. The water depth is 9 m and the vehicle depth 3 m. The range spans from 0 m to 150 m from left to right. The main environmental change is a reduction of the wind speed from 13 m/s of the upper image to 4 m/s of the lower image.

Fig.2: The measurement-based SNR for beamformed (sidescan) lines of sonar data (i.e. prior to SAS image processing) is presented in black. The solid and dashed lines give the results for the good and poor image respectively. The grey lines correspond to simulations for the environmental conditions of the two measurement campaigns.
The raytracer is based upon the analytic solution for piecewise constant gradient of the sound speed from [6], sec 6.2. For bistatic bottom scattering we selected Ellis’ model [7] combining Lambert scattering valid for intermediate angles with a facet term for specular reflection. The model is extended to smaller grazing angles by including Del Balzos plateau as a correction term to the Lambert scattering [8]. For bistatic surface scattering we choose the APL-UW94 backscattering model [9] as a basis and performed a brute expansion to a bistatic model by mimicking the layout of Ellis’ model. The model results show good resemblance in forward and back-scattering benchmarking. Finally additive noise is mimicked by adding a noise level that prevents the sensor from obtaining a SNR of more than 0 dB at 300 m.

In the model computations we assume bottom consisting of gravel (as indicated from the SAS-image), adopt wave height estimated from wind strength, and a sound speed profile recorded with a CTD profiler. The sonar is simulated with a transmitter beamwidth of 15° steered 15° below the horizontal plane and a receiver beamwidth of 28° centred around 22° below the horizontal plane.

4. ANALYSIS

The modelled SNR were presented together with the measurement-based SNR in Fig. 2, and a good resemblance between the two can be observed, in particular in the range interval of 30 m to 100 m. We believe that the discrepancies at longer range are effects of the increasing sensitivity to inaccurate assessments of the sound speed profile, additive noise, bathymetry and/or bottom type for long range and very small grazing angles. The discrepancies between measurements and model at short range we believe is related to a combination of an overestimated modelled SNR and an underestimated measurement-based SNR. The modelled SNR is unrealistic high in some areas where the direction of the multipaths coincide with zeros in the transmitter or receiver beampatterns. The resulting complete loss of a multipath only occurs in our simplified 2D model, as both a 3D model and inclusion of diffuse scattering at all interfaces would smooth the directivity of the contributions. Furthermore, the SNR estimated from coherence measurements is possibly too low in the short range region. This could be caused by baseline decorrelation and processing induced decorrelation [5], and will be investigated further.

We now return to the two images and try to identify the effect causing the huge difference in their SNRs. By adjusting one parameter of the modelled environments at a time and evaluating the related effect on the SNR, we observe that the main origin of the changing SNR is the different sea states of the two days with wind speeds of 13 m/s for the good image and 4 m/s for the poor image.

Multiple paths with the same round-trip distance are characterised by their different angles out of the transmitter and into the receiver, as illustrated by the round-trip paths in the bottom panel of Fig. 3. By shaping the transmitter and receiver beam patterns beneficially, it is therefore possible to reduce the effect of undesired multipaths. In Fig. 3 we visualize the contribution of the dominating multipaths for the modelled sonar and environments. All multipaths contributing 10% or more to the total contribution at any distance is included in the figure, with the results for the good quality image in the top panel, and the poor quality image in the centre panel. The individual multipaths of each colour are illustrated for one roundtrip distance in the lower panel. We observe that the relative contribution of the multipaths are stronger in the poor quality image, and that multipaths with two bottom and one surface scattering give the main contributions to the SNR in our examples. Under windy
conditions a better imaging range is achieved, probably as a result of damping and spreading of the multipath returns at the surface.

Fig. 3: Individual multipath contributions of the images for windy and calm conditions in the top and centre panels respectively. The direct return is indicated with the black solid line and the total multipath contribution with the black dotted line. Furthermore, all multipaths contributing 10% or more to the total multipath contribution are plotted in different colours. The yellow line indicates an assumed additive noise floor. The lower panel indicates the path corresponding to each colour, exemplified for round-trip ray paths of 65 m length. The paths are also indicated by the legend, stating the order of bottom (b/B) and surface (s/S) scatterings for each colour, and with the single non-specular scattering indicated by an uppercase letter.
5. CONCLUSION

We conclude that the interferometric sonar coherence yields an effective estimate of quality for interferometric side scan sonar and synthetic aperture sonar measurements. It allows for a simple real-time estimate of the imaging range and in combination with simulations provides the means required for autonomous on-the-fly change of the sonar geometries and settings for optimal performance.

REFERENCES

Regular Session II

Acoustic studies of marine mammals
PRELIMINARY RESULTS OF ACOUSTIC SURVEYING FOR BEAKED WHALES IN THE CORAL SEA NEAR AUSTRALIA

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Abstract: A combined visual and acoustic survey for beaked whales was conducted over a three week period in August, September 2008 in the Coral Sea off the east Australia. The survey area is used for naval exercises and the survey was intended as a trial of methods of detection of beaked whales for the purpose of determining their distribution across the area. Two types of acoustic system were used: a towed array with four closely spaced hydrophones and two autonomous drifting recording systems with hydrophones suspended to 400 m depth. Water depths in the area vary from 300 m to more than 3,500 m. Data were sampled at 300 kHz (towed array) and 192 kHz (drifting recorders) and recorded on computer hard disks. Data were analysed using a range of acoustic analysis software, automated using matched filter techniques. Preliminary results have found many detections of acoustic signals with similar wave form, frequency range, frequency sweep, number of cycles and repetition rate as those reported for beaked whales in the northern hemisphere. Thousands of clicks typical of beaked whales have been counted, compared with only six sightings of 12 individual beaked whales. Higher densities of detections were found in areas of steep bathymetric slopes than over the deep plains.

Keywords: beaked whales, beaked whale surveys, beaked whale acoustics, passive acoustic monitoring.
1. INTRODUCTION

Beaked whales are particularly elusive whales. They are small and rarely seen and their distributions are poorly known. The predominance of beaked whales in strandings that have occurred at times of a number of naval exercises in the Northern Hemisphere has led to a need for improved knowledge of their distributions for management of impacts. The effectiveness of visual surveying for beaked whales is, however, too limited to be useful because they are so difficult to detect visually. A few years ago data from DTAG sound recorders placed on two species of beaked whale, *Ziphius cavirostris* (Cuvier’s beaked whale) and *Mesoplodon densirostris* (Blainville’s beaked whale) showed that their vocalisations are distinctively different in several acoustical characteristics from those of other toothed whales, providing a reliable means of detection and identification [1]. The recordings also showed that these two species vocalised repeatedly when several hundred metres down during deep foraging dives. This paper describes some preliminary results of a survey for beaked whales in an area of the Coral Sea off east Australia used for naval exercises. The survey was intended as a trial of equipment and techniques.

2. METHODS

A three week survey was conducted in August, September 2008 in the Coral Sea off the east coast of Australia. The survey covered a box extending about 171 km E-W and 60 km N-S (Fig. 1), the coordinates of the NW corner of the box being 22°20´ S and 54°00´ E. Water depths varied from 300 to over 3,500 m and included the continental slope in the western part of the area and coral islands with steep sloping edges to the east. A Royal Australian Navy landing craft, HMAS *Labuan* was provided for the survey which involved both visual and acoustic monitoring. Two types of acoustic passive recording systems were used: a towed array and two drifting recording systems (“acoustic loggers”). The towed array supplied by Ecologic UK Ltd., consisted of two mid frequency channels (500 Hz – 30 kHz) and two high frequency channels (500 Hz – 150 kHz) and depth sensor. The hydrophones were closely spaced at the end of the 400 m cable. Data were recorded on hard disk via a data acquisition card sampling at 300 kHz.

The two drifting systems (“acoustic loggers”) were developed by the Centre for Marine Science and Technology at Curtin University for the purpose and comprised a computer controlled recording system in a container suspended below surface buoys and with a hydrophone at a depth of 400 m. A radio beacon transmitted the GPS position of the buoys allowing the survey team on the ship to keep track of their position. The logger systems were deployed for a period of a few days at a time, and then recovered, the data downloaded and the systems redeployed. While the towed array provided continuous monitoring, the much deeper hydrophones of the drifting systems were considered to provide an advantage in the detection of clicks from beaked whales foraging at depth.

Data have been analysed using MATLAB and the following acoustic software packages: Adobe Audition (Adobe Systems), Ishmael [1] and PAMGUARD. Click detections were compared with those presented by Johnson et al. [2] and samples that were verified as typical of beaked whale were used in matched filtering to automate the analysis.

Acoustic detection ranges of beaked whale clicks are limited because of the higher absorption of sound at the high frequencies of the vocalisations, and because of their narrow
beam pattern. Zimmer et al. [3] estimate that the range for 50% probability of detection using a receiver depth of 100 m 1.5 and 3.8 km, and detection beyond 4 km would be very unlikely. These estimates are for low background noise and shorter ranges are to be expected for higher noise levels as conditions (e.g. wind speed) changes. Hence a detection within our survey area would be localised to a small area compared to the size of the survey area, just by the limiting range of detection.

Fig.1: Map showing the location of the survey area and the bathymetry. Water depth varies from 300 m to over 3,500 m. The NW corner of the survey area is 22°20´ S, and 134°00´ E.

3. RESULTS

The visual survey resulted in 75 sightings of more than 500 individuals of a range of cetacean species including several species of dolphins, pilot whales, sperm whales, humpback whales, and minke whales. There were six sightings of 12 individual beaked whales of unidentified species. The acoustic survey detected sounds typical of a range of species.

A large number of clicks were detected, including some typical of sperm whales and some typical to dolphins. Some clicks were found to be remarkably similar to those reported by Johnson et al. [2] for the two species of beaked whales and were therefore identified as beaked whale sounds. They consisted of tone bursts sweeping up in frequency over the range from about 25 kHz to 50 to 60 kHz with durations of 200 to 300 μs. The envelopes of the tone bursts showed varying shapes, but generally within the range shown in [2]. Inter-click intervals were also similar. Using the automated detection techniques, thousands of clicks typical of beaked whales have been detected. Individual beaked whales are known to produce large numbers of clicks while foraging, so the actual numbers of individual numbers...
of beaked whales detected will be much fewer than the number of clicks recorded. The highest density of clicks in the area surveyed has been over the steep slopes of Cato Is., a coral island, the lowest over the deep plains.

4. ACKNOWLEDGEMENTS

Funding for survey was provided by the Royal Australian Navy (RAN), who also funded the development of the drifting acoustic recordings systems and the purchase of the towed array and other equipment used on the survey. In particular, the role of the Commander Stephen Cole, the RAN Environmental Manager in obtaining the funding and providing general support and encouragement is much appreciated. We are also particularly indebted to the Commanding Officer and crew of HMAS Labuan for their enthusiastic participation and support throughout the survey.

REFERENCES

Correlation of the Broadband Spectral Characteristics of Bottlenose Dolphin Signatures with Dolphin Behavior in the Mississippi Sound

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Abstract: A series of acoustic and visual observations were made of bottlenose dolphins in areas around the Mississippi sound. A portable acoustic monitoring system recorded echolocation clicks, wideband burst pulses and narrow-band frequency modulated whistles. The system was located on a small boat which came in close contact with different concentrations of bottlenose dolphins. Observations of concurrent dolphin behavior where categorized according to the operational definitions used by Jones and Sayigh (Marine Mammal Science, 2002). The broadband spectra of the recorded signals are calculated and the spectral variations (level and frequency) correlated with observed dolphin behavior. Work supported by the Institute for Marine Mammal Studies
Brief sound intensity discrimination by a bottlenose dolphin

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Abstract: Studies on a dolphin's biosonar target recognition are focused on acoustic cues associated with individual echoes returning from ensonified targets. Differences in echo highlight intensities, waveforms as well as frequency spectra are all considered to be possible acoustic cues for target discrimination. An echolocating dolphin is capable of detecting a 1-dB difference in target strength of stationary targets. Around 1-2 dB intensity threshold difference was also determined at bottlenose dolphin discrimination of brief sounds. To measure intensity discrimination thresholds in dolphins, comparison targets or sounds are normally presented to a dolphin simultaneously from two directions (at some azimuth separation). In present study we compare a bottlenose dolphin intensity threshold for brief sounds presented simultaneously from different directions to the intensity threshold for the same brief sounds presented from the same direction with a small delay relative to each other. We found that the intensity threshold for high frequency clicks (similar to the bottlenose dolphin sonar clicks) presented to the dolphin from two transducers separated in azimuth by 30 degrees was around 2.5 dB. At the same time the threshold intensity difference for the same clicks transmitted from the same transducer with delay as small as 20-30 microseconds was found to be just about 0.4 dB. The results suggest that the bottlenose dolphin is capable of very fine intensity discrimination of highlights within a single target echo. The results also indicate the bottlenose dolphin auditory time resolution to be as high as 20-30 microseconds.
Boat effects on the behaviour of Indo-Pacific Humpback (Sousa chinensis) and Irrawaddy (Orcaella brevirostris) dolphins in Cowie Bay, Sabah

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Abstract: Lack of information about the effects of anthropogenic activities on marine mammals was known. Different types of boats give different impacts on the behaviour of Indo-Pacific Humpback (Sousa chinensis) and Irrawaddy (Orcaella brevirostris) dolphins in Cowie Bay, Sabah. The study was conducted during Spring and Neap tides from April to September 2008 and it consists of 24 days. The objective of the study was to determine the effects of fish trawler, speed boat, tug boat and passenger boat on the behaviour of the two species. Dolphins’ behaviour described as follows, negative, positive, and neutral behaviours. Effects were described as follows when pods of dolphin moved away (negative) from the boat, moved towards (positive) the boat or dolphin remains perform own (neutral) activities. Direct and indirect impacts, long term impacts were observed during data collection period. Direct impacts; dolphin shows tendency to show positive effects on trawler boat. Speed boat and passenger boat gave negative effects while tug boat gave neutral effects on the two species. Data analyses showed that positive effect was more frequently observed on trawl boat, negative effects on passenger and speed boat, neutral effects on tug boat. When trawlers were present, dolphins shows positive effects by chasing and fed behind them. Previous studies described this as indirect negative impacts (long term impacts) of this activity on dolphins normal feeding behaviour. Limiting the number of boats by enforcement could be effective in minimizing the long term impacts on the dolphins. Keywords: marine mammals, Sousa chinensis, Orcaella brevirostris, Cowie Bay, behaviour, impacts.
Regular Session III

Ambient noise
INTERACTION BETWEEN ACOUSTIC FIELDS OF THE AMBIENT NOISE AND POINT SURFACE SOURCE

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Abstract: In measuring acoustic field vector properties either complete or partial cancellation between ambient noise energy flux and the oppositely directed one, radiated, for instance, by tone or noise-shaped point source can be found. In that case there is no matter coherent or incoherent fields interact. The phenomenon of interest is named compensation of opposing energy fluxes. The paper is focused on interaction between vertical component of the surface noise energy flux and that of bottom-reflected noise-shaped interfering signal from on-water source in the deep open ocean. Bottom-reflected weak broadband interfering signal measures the extent of resultant compensation. As the vertical projection of the signal energy flux makes the interference maximum, its power appears to be comparable to that of the surface noise and complete compensation takes place at a given frequency. Once it is the interference minimum, just partial compensation can be found. No compensation was observed for co-directional energy fluxes.

Keywords: Acoustic energy fluxes, underwater ambient noise, cancellation between energy fluxes
INTRODUCTION

Acoustic energy fluxes of different signal or noise origin cross each other in underwater waveguide. Making signal measurements in terms of scalar approach, for example, by means of hydrophone arrays, one can detect a signal against noise background in a given frequency band. However, vector measurement devices may probably provide a different result. Since cross-spectrum signal-ambient noise energy density turns to zero under certain conditions (i.e. at specific frequency and direction a zero net energy flux is to be found).

The paper is focused on interaction between vertical component of the ambient dynamic noise energy flux and noise-shaped signal from a point on-water source. The authors studied phenomena resulted from simultaneous interference and compensation manifestation in deployments done in different regions of the Earth Ocean. Eight-channel freely drifting combined telemetric systems built by the Acoustic Ocean Noise Laboratory (the Pacific Oceanological Institute, Vladivostok) were employed [1]. The experiment presented below was done in the Sea of Philippine, the deployment site was 18°53′N, 126°38′E. Measurement system combined sensors were 150 and 500 m deep. The whole of eight channels were at work. Under current meteorological conditions of the experiment with moderate rain vertical ambient noise energy flux spectrum \( S_{PV_z} (f) \) was 60 to 68 dB re 1μPa²/Hz in the 50 to 800 Hz frequency band. At the depth of 500 m the vertical component of the noise-shaped point source signal spectrum was 1 to 10 dB greater than that of the surface noise in the continuous part of the spectrum. At this depth the noise and signal energy fluxes are co-directional. The experimental data processing used Fourier transformation of pressure \( p(t) \) and particle velocity projections \( V_x(t), V_y(t), V_z(t) \) data series. Pressure-particle velocity spectra \( S_{PV_i} (f) \), corresponding phase-difference spectra \( \Delta \phi_i (f) = \varphi_{Pr}(f) - \varphi_i(f) \), and simple one-point coherence function spectra \( \gamma_i^2 (f) \) were computed from the following expressions:

\[
\begin{align*}
S_{PV_i} (f) &= |S_{PV_i} (f)| e^{-j\Delta \phi_i (f)}, \\
\Delta \phi_i (f) &= \arctan \left( \frac{\text{Im} S_{PV_i} (f)}{\text{Re} S_{PV_i} (f)} \right), \\
\gamma_i^2 (f) &= \frac{|S_{PV_i} (f)|^2}{S_{PV_j} (f) S_{PV_j} (f)} , \quad i = x, y, z.
\end{align*}
\]

(1)

The discussion below uses only \( \gamma_z^2 (f) \) and \( \Delta \phi_z (f) \) for special features of interaction between opposite energy fluxes are most pronounced in these spectra. The first description of the compensation phenomenon observed in deep open ocean and shallow water was presented in [2,3]. A kind of “scent” of compensation is occasionally shown in later papers by different researchers (e.g. Fig.6 in [4]) although with no any analysis.

RESULTS OF THE EXPERIMENT

Conditions. Ocean depth at the deployment site is about 4900 m. Wind speed of about 7 m/s and well-developed wind-generated waves. Surface homogeneous water layer with sound speed of 1536.8 m/s is about 100 m thick; the minimum sound speed at the depth of
underwater sound channel axis - 1000 m - is 1482.7 m/s; sound speed at the bottom of 1542.0 m/s is greater than that at the surface.

Measurements done at the depth of 150 m. General time-dependence of the acoustic field properties is shown on sonograms in Fig.1. They present the acoustic pressure spectrum $S_{p^2}(f,t)$, the vertical coherence function $\gamma_z^2(f,t)$, and the phase difference between acoustic pressure and pressure gradient vertical projection $\Delta \varphi_z(f,t)$ as functions of both time and frequency.

**Fig.1: Sonograms of traveling on-water source; a) pressure autospectrum $S_{p^2}(f,t)$; b) vertical projection of the coherence function, $\gamma_z^2(f,t)$; c) phase difference between pressure and z-projection of the pressure gradient $\Delta \varphi_z(f,t)$. Exponent averaging over 10 s, frequency bin of analysis 1.2 Hz, measurement point depth 150 m, engine on-time $t=570$ s.**
Distant shipping noise of 90 dB re 1μPa^2/Hz is well-pronounced on the autospectrum sonogram $S_{p^2}(f,t)$ in the 10 to 100 Hz frequency band during the entire measurement. Slight ‘scent’ of tropical shower is visible in the autospectrum from 200 to 500 s. Vertical component of the coherence function appears to be more affected by the rain. In Fig. 1a moving vessel $(t > 570 s)$ can be identified by its broadband noise spectrum with discrete lines all over it. No interference is shown in the autospectrum $S_{p^2}(f,t)$ since on-water source moves in deep ocean. However, $\gamma_z^2(f,t)$ and $\Delta \phi_z(f,t)$ sonograms clearly show periodic variations in coherence and phase difference at $t > 570 s$. The paper studies the properties of interference introduced by on-water noise source as well as influence of periodically changing energy flux on surface noise characteristics. Consider several specific spectra chosen from a vast body of sonograms (see Figs. 1b,c). Fig.2 shows $\gamma_z^2(f)$ and $\Delta \phi_z(f)$ spectra of underwater ambient noise at $t = 560 s$.

That is the point a shower turns to a moderate rain. Here the coherence function is about 0.7 at high frequencies. Average $\Delta \phi_z(f)$ is -90° that agrees with downward surface-generated dynamic noise energy flux [1]. In Figs. 3b and 4b, for clarity sake, the phase difference between the pressure and vertical component of the pressure gradient is analyzed. Moving research vessel produces periodical changes in $\gamma_z^2(f)$ and $\Delta \phi_z(f)$ spectra seen in Fig.3, in contrast to Fig.2.

Fig.4 shows $\gamma_z^2(f,t)$ sonogram measured by sensor 2 at the depth of 500 m. Synchronous 150- and 500-m sonograms are shown in Figs. 1 and 4. At $t < 570 s$ high-coherence rain-produced area is well-pronounced. Ambient noise $\gamma_z^2(f)$ and $\Delta \phi_z(f)$ spectra measured at the depth of 500 m agree with those at the depth of 150 m shown in Fig.2. However, no periodical changes in either $\gamma_z^2(f)$ or $\Delta \phi_z(f)$ spectra are seen above 570 s. This is because grazing angles at 500 m are greater than at 150 m.
Fig. 3: Spectrum of the vertical projection of (a) the coherence function; (b) the phase difference of on-water source against the noise background obtained from Fig. 1b,c at t = 990 s. $K_i = f_i / f_0$, $f_0 = 758.7$ Hz. The conditions are the same as in Fig. 1.

Fig. 4: $\gamma_z^2(f,t)$ sonogram. Measurement point depth 500 m. The conditions the same as in Fig. 1.

Contribution of these rays to downward-directed energy flux component is comparable to the surface noise or 3 to 10 dB greater and depends on frequency, e.g. between 500 and 550 Hz the excess is about 10 dB. In this situation bottom-reflected energy flux is less than downward-directed sum of the surface noise flux and noise-shaped signal one. The coherence function decay similar to that shown in Fig. 3 for the depth of 150 m is not found in Fig. 4, since ship- and surface- generate energy fluxes are co-directional.
CONCLUSIONS

The paper presents the study of interaction between statistically independent acoustic fields by the example of compensation detected in cross-spectral energy density function of the field produced by opposing energy fluxes. Vertical projections of coherence function and phase spectra are under consideration. The paper analyses both situations, 1) with opposing signal and noise energy fluxes, and 2) co-directional ones. In the first case – the measurement point is 150 m deep – cross-spectral energy density of bottom-reflected noise-shaped signal is either comparable to that of the surface noise (in interference maximum) or less (in interference minimum). At this depth a steady compensation between noise and interfering signal can be observed. In the second case – the measurement point is 500 m deep – there is no compensation between co-directional noise and signal. This is the case noise-shaped signal spectrum level is either comparable or greater than the ambient noise level.

The result implies building a criterion to detect a low-power signal against noise background. It can use measurements of ambient noise properties including phase difference between pressure and particle velocity (or pressure gradient) and one-point coherence function.

Above-discussed phenomenon of compensation detected in cross-spectral energy density function of opposite energy fluxes field provides a base to a novel approach to several problems in applies underwater acoustics including problem of detection. From the other hand, any hydrophone system appears to be incapable in monitoring compensation phenomenon.

REFERENCES

Passive acoustic thermometry of liquids: effects of strong absorption, near thermal fields, radiation pattern of antenna.

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Abstract: Passive acoustic thermometry (PAT) is an emerging technology for noninvasive measurement of inner temperature. It is based on registration of thermal acoustic radiation entering from the depth of investigated medium. The potential sensitivity (~0.1°C) and resolution (~1 mm) makes PAT attractive for many practical applications such as medical diagnostics, for example. The theoretical basis of PAT is the relation between measured thermal acoustic radiation and spatial distribution of thermodynamic temperature. This forward problem solution was obtained earlier with the help of ray theory. However, ray theory sets aside some important effects. In this work we used more general approach to the solution of PAT forward problem in liquid-like media. Our approach was based on the fluctuation-dissipation theorem applied to hydrodynamic equations. We have obtained common integral relation between one-dimensional temperature profile of investigated medium and acoustic pressure measured above the surface by acoustics radiometer. Using this theory we have analyzed the influence of quasistationary field to thermal acoustic emission, as well as the effects of strong acoustic absorption and wide radiation pattern of receiving antenna which were not taken into account via ray theory. The conditions were defined when the obtained solution transforms to more particular result of ray theory. We have also studied the role of the above effects in PAT measurements of subsurface human body temperature. New method for control of PAT sounding depth was suggested which allows one to retrieve the temperature profile inside the investigated medium by solving obtained integral equation.
CASPIAN SEA UNDERWATER ACOUSTICAL CHARACTERISTIC MEASUREMENTS

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Abstract: Measurement and analysis of physical and chemical characteristics of sea water are of crucial importance in many applications. ANZALI is an active and noisy port located in south west of the Caspian Sea. There, however, are limited published data about physical and underwater acoustical characteristics of the port.
During February and March 2007, measurement of underwater acoustical noise and Transmission loss in the zone, was conducted using two boats, a hydrophone array system and an acoustical projector. Some associated quantities such as temperature profile; wind and water flow direction and magnitude were measure, prior to which, water temperature measurement had been done in some different successive seasonal conditions. It is worth to mentioning that, water depth of the zone changes slowly from 100 to 200 meter.
Following preprocessing of measured data, Power Spectrum (PS) and Probability Density Function (PDF) estimation algorithms were applied to the data.
We recognize a significant increase in PS levels below 200Hz in different weather conditions and industrial activities. Considering seasonal temperature measured data; we conclude that water temperature, between surface and depth of 50 meter drops sharply from 27 to 8 centigrade at summer and slightly from 10 to 8 centigrade at winter. While in depths below 50 meter, water temperature is not considerably affected by air temperature and varies slowly from 8 to 7. In addition, we found that in shallow regions the Transmission Loss is in accordance with the spherical spreading model that is reasonable regarding temperature profile and sea bed materials.

Keywords: Caspian Sea, ANZAI, Underwater noise, Power spectrum, Hydrophone
1. INTRODUCTION

ANZALI is an active and noisy port located in south west of the Caspian Sea. On account of cumulative acoustical and chemical pollution that affect marine life in similar ports, periodic measurement of the physical and chemical characteristics of the sea is necessary in this region. Since wind, shipping traffic and water temperature profile have a considerable influence on total background noise level, they have to be simultaneously measured and the results must be compared with valid published levels, WENZ for instance [1].

Thus far, published research results on physical specifications of Caspian Sea are very limited, among which we can mention international marine cruise done in 1996 [2] and some coastal measurements near to NOSHAHR rivers estuary [3]. In this presentation, measurement and analysis of underwater acoustical parameters of ANZALI port is investigated.

2. EXPERIMENT DETAILS

Main parts of underwater acoustical noise and Transmission loss measurement were accomplished on 17 February 2007 and the complimentary experiments were done on 11 March 2007 and 28 July 2008. Experiments area has approximately 200m depth and, as shown in fig.1, is located at 15km to the north east of the port.

Two boats were used for transmitting and receiving signals in different distances. The Receiver system consists of a hydrophone array (two 8101 and one 8105 B&K standard hydrophones), preamplifiers, filters, data acquisition unit (12 bit, 20 KHz for each hydrophone) and portable computer. The hydrophones were hanged in 2.4, 8 and 28 meters depths behind the first boat. A balancing weight put below the lowest hydrophone to neutralize water current effects.
Calibration of hydrophones was done according to respective procedures. Transfer function of other electronic devices was measured by injecting white noise into the inputs and measuring outputs amplitudes and phases. The response of the Receiver hardware is flat within 30-10000 Hz frequency band which means that the estimated PS is correct excluding above - mentioned frequency interval.

During 3 hours, several data measurements were done in all channels simultaneously. The duration of data batches was 40 sec with 12 sec time interval between sequential batches.

Temperature, salinity and sound velocity profiles was measured in deepest point of the zone using standard CTD meter (SIS CTD PLUS 1000 ITS-90) at its sequential time mode. In addition, wind and water current speed and direction were measured using portable devices. These measurements were repeated several times in different seasons.

The experiment consists of several phases. In underwater background noise measurement phase, vessel’s engine became silent and the support vessel had kept away more than 6 km. Wind velocity and direction, water temperature, salinity profile, and water current were measured parallel to underwater noise measurement. Dividing the amplitude of measured data by sensitivity and gain coefficients, we converted them to sound pressure that is shown in Fig. 2. Data health check and suitable preamps gain choice were implemented by hearing measured data and showing time domain representation. So we found that channel number 3 installed near to surface, have some impulsive noise that might have been possibly caused by surface perturbation and cable vibration.

In the second step, support vessel towed a small 1800Hz sine wave generator and passed in front of receiver system with fixed 3.2 m/s speed directly toward the end point. The projector source level had been measured 140dB re 1μPa at 1800Hz. Thus the measured data were a combination of the vessel noise and the narrowband projector noise.

![Time domain representation of two hydrophones' signals hanged in depths of 5m (ch1) and 28m (ch2)](image)

**Fig. 2: Time domain representation of two hydrophones’ signals hanged in depths of 5m (ch1) and 28m (ch2)**

### 3. MEASUREMENTS

In Fig. 2 time domain representation of underwater background noise relative to 1μPa are shown in two channels suspended in 8 and 28m depths.

Measured temperature and salinity profiles in two different seasons are shown in Fig. 3. In addition, our last measurement in Noshahr [4] and International cruise [2] surveys are
presented so as to compare the results. The cruise comprise several measurements in 18 different points among which only one (37.5N 50.7E) was located near to Anzali port.

Underwater current varied from .15 to .5 m during experiment and its direction was approximately parallel to coastline (290 degree with respect to North Pole). Wind speed was 5 miles per hour that caused surface waves with height of less than 30cm (equal to 2 Beaufort scale). The narrow band projector received signal level in successive saved batches is depicted in Fig. 4.

4. ANALYSIS AND CONCLUSION

Welch method which is one of the well-known PS estimation methods was applied to the measured noises. In order to detect narrow band component produced by man made noise the frequency resolution has been set to 1Hz at first and thence to 10Hz to give a smooth estimate of PS. Fig. 5 represents the PS estimate of measured background noise.
Fig. 4: Measured level of the narrow band 1800Hz projector passing in front of the receiver system in a straight line. Batch numbers 20, 30, 40 and 50 are corresponding to 320, 733, 1320, and 1560 meters Transmitter-Receiver distances respectively.

Since background noise PS does not vary in short period of time, PS estimation ought to be applied to successive data batches to recognize any probable effect of transient signals resulting from either hidden industrial activities or passing boats. So we found three narrow band components whose centre frequencies vary slowly during 420 sec.

Fig. 5: PS estimates of the outputs of two hanged hydrophones using Welch algorithm (freq. resolution 10Hz, Hanning window, signal duration 40 sec, time 15:10).

Regarding our measurements and international cruise data, we came to realize that water temperature profile above 40m depends heavily on air mean temperature in different seasons, but it is pretty stable at farther depths and slowly decreases to reach 7 degrees centigrade at a depth of 900m. Fig 6 shows the result of calculated sound propagation velocity using temperature and salinity data. Based on these results, we may expect two different sound propagation situations: First, in summer, downpour of surface generated noise makes the propagation loss have higher values compared to that by spherical spreading model; in our experiment (Fig 4) we measured 35dB/decade loss. Second, in winter, due to insignificant variation in sound velocity profile, spherical model is an appropriate one for propagation of
sound with sources located in any depths; the last experiment[4] showed 18dB/decade loss in winter.

![Calculated Sound velocity profile during the experiments](image)

**Fig.6: Calculated Sound velocity profile during the experiments**

**REFERENCES**


Regular Session IV

Reverberation
NON-STATIONARY PROPAGATION OF CLUTTER IN A DISPERSIVE MEDIUM

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Abstract: We show that phase space methods are effective in understanding wave and noise propagation in a dispersive medium. The phase space distributions we use are the Wigner distributions of position and wave number or time and frequency. In addition, we show that the Wigner distribution offers approximations that are accurate and easy to apply.

Keywords: dispersion, propagation, Wigner distribution
1. INTRODUCTION

Our aim is to study how a noise field propagates in a dispersive medium. Suppose, for example, we know the statistics of the noise at a point in space, then, we aim to obtain the statistics of the noise at a later time at an arbitrary position. In this paper we consider the medium to be deterministic and the randomness comes in only in the initial distribution. We first study deterministic propagation of a wave by phase space methods. Subsequently we apply the method to noise. For wave equations with constant coefficients

\[
\sum_{n=0}^{N} a_n \frac{\partial^n u}{\partial t^n} = \sum_{n=0}^{M} b_n \frac{\partial^n u}{\partial x^n}
\]

the standard Fourier method of solution is obtained by substituting \( e^{ikx-i\omega t} \) into Eq. (1) to give [4,6,7]

\[
\sum_{n=0}^{N} a_n (-i\omega)^n = \sum_{n=0}^{M} b_n (ik)^n
\]

This is an algebraic equations between \( k \) and \( \omega \). One solves for \( k \) in terms of \( \omega \) or \( \omega \) in terms of \( k \); which is done depends on the type of initial conditions. Also, in general the relation between \( k \) and \( \omega \) is complex and we write

\[
\omega(k) = \omega_n(k) + i\omega_t(k)
\]

\[
k(\omega) = k_R(\omega) + ik_I(\omega)
\]

Of fundamental significance is the group velocity

\[
v_g = \omega_R(k)
\]

and similarly one defines the "group slowness", or the "unit transit time" as

\[
\tau_g(\omega) = k'_R(\omega)
\]

We now discuss two types of initial conditions and the corresponding phase space distribution. If we know the field as a function of position at a given time \( t_0 \), \( u(x,t_0) \), then the field at a later time is given by [4,6,7],

\[
u(x,t) = \frac{1}{\sqrt{2\pi}} \int S(k,t_0) e^{ikx-i\omega_0(t-t_0)} dk
\]

where \( S(k,t_0) \) is the initial spatial spectrum, obtained from \( u(x,t_0) \), by

\[
S(k,t_0) = \frac{1}{\sqrt{2\pi}} \int u(x,t_0) e^{-ikx} dx
\]

Defining

\[
S(k,t) = S(k,t_0) e^{-i\omega_0(t-t_0)}
\]

shows that \( u(x,t) \) and \( S(k,t) \) form Fourier transform pairs for any \( t \),

\[
u(x,t) = \frac{1}{\sqrt{2\pi}} \int S(k,t) e^{ikx} dk
\]

\[
S(k,t) = \frac{1}{\sqrt{2\pi}} \int u(x,t) e^{-ikx} dx
\]

For this case the phase space is defined as that of position and wave number [1]

\[
W(x,k,t) = \frac{1}{2\pi} \int u^*(x-\lambda/2,t)u(x+\lambda/2,t)e^{-i\lambda} d\lambda
\]

which can equivalently be expressed in terms of the spatial spectrum, \( S(k,t) \).
\[ W(x,k,t) = \frac{1}{2\pi} \int S^*(k + \theta/2,t)S(k - \theta/2,t)e^{-i\theta} d\theta \]  

(13)

One can express the Wigner distribution at time \( t \) in terms of the Wigner distribution at time \( t_0 \), \([2, 5]\)

\[ W(x,k,t) = \frac{1}{2\pi} \iint W(x',k,t_0)e^{i\theta(k-x')}e^{i\omega\theta(k-\theta/2)-(\omega(k-\theta/2))t} d\theta dk' \]  

(14)

This equation is exact. We have shown that a excellent approximation is \([2,5]\)

\[ W(x,k,t) \approx e^{2i\omega(k-k_0)(t-t_0)}W(x-v_0(k)(t-t_0),k,t_0) \]  

(15)

For the second case suppose we are at a given position, \( x_0 \), and produce a signal as a function of time, \( u(x_0,t) \), then the signal at a different spatial point, \( u(x,t) \), is given by

\[ u(x,t)=\frac{1}{\sqrt{2\pi}} \int F(x_0,\omega)e^{ik(\omega(x-x_0))} d\omega \]  

(16)

where \( F(x_0,\omega) \) is the initial time spectrum at \( x = x_0 \),

\[ F(x_0,\omega) = \frac{1}{\sqrt{2\pi}} \int u(x_0,t)e^{i\omega t} dt \]  

(17)

If one takes

\[ F(x,\omega) = F(x_0,\omega)e^{i\omega(x-x_0)} \]  

(18)

then \( F(\omega,x) \) and \( u(x,t) \) form Fourier transform pairs

\[ u(x,t)=\frac{1}{\sqrt{2\pi}} \int F(x,\omega)e^{-i\omega t} d\omega \]  

(19)

\[ F(x,\omega)=\frac{1}{\sqrt{2\pi}} \int u(x,t)e^{i\omega t} dt \]  

(20)

Explicitly

\[ u(x,t)=\frac{1}{2\pi} \iint u(x_0,t')e^{ik(\omega(x-x_0))} d\omega d\omega' \]  

(21)

For this case one defines the time-frequency Wigner distribution of the wave at a particular \( x \),

\[ W(t,\omega,x) = \frac{1}{2\pi} \int u^*(x,t-t/2)u(x+t/2)e^{-i\omega t} dt \]  

(22)

Also

\[ W(t,\omega,x) = \frac{1}{2\pi} \int F^*(x,\omega+\omega')F(x,\omega-\omega')e^{-i\omega' t} d\omega' \]  

(23)

One can express the \( W(t,\omega,x) \) in terms \( W(t,\omega,x_0) \) \([2, 5]\)

\[ W(t,\omega,x) = \frac{1}{2\pi} \iint W(t',\omega,x_0)e^{-i\omega(t'-t)}e^{-i\omega'((x-x_0)')} dt' d\omega' \]  

(24)

In analogy with Eq. (15) we have shown that an effective approximation is \([2,5]\)

\[ W(t',\omega,x_0) \approx e^{-2i\omega(x-x_0)}W(t-t_0)(-\omega)(x-x_0),\omega,x_0) \]  

(25)
2. PROPAGATION

In this paper, due to space limitation, we deal with case two above. We define the autocorrelation function by

\[ R(t_1, t_2; x) = E[u(x, t_1)u^*(x, t_2)] \]  
and define the Wigner-Ville spectrum as the ensemble average of the Wigner distribution,

\[ \overline{W}(t, \omega) = \frac{1}{2\pi} \int E[u^*(x, t - \tau/2)u(x, t + \tau/2)]e^{-i\omega \tau} d\tau \]  
Taking the inverse we have

\[ R_x(t + \tau/2, t - \tau/2; x) = \overline{W}(t, \omega, x)e^{i\omega \tau} d\omega \]  
and

\[ R(t_1, t_2; x) = \overline{W}(\frac{t_1 + t_2}{2}, \omega, x)e^{-i(t_2 - t_1)\omega} d\omega \]  
Suppose noise is generated at a particular position, \( x_0 \), and we wish to determine the statistics of the noise at another position, \( x \). We take the ensemble average of the Wigner distribution Eq. (24)

\[ \overline{W}(t, \omega, x) = \frac{1}{2\pi} \int \overline{W}(t', \omega, x_0)e^{-i\omega (t' - t)} e^{-i[k^*\left(-\omega + \omega' / 2\right) - k\left(\omega - \omega' / 2\right)](x-x_0)} dt' d\omega' \]  
Using the fact that

\[ R(t + \tau/2, t - \tau/2) = \overline{W}(t, \omega)e^{i\omega \tau} d\omega \]  
one can show that

\[ R_x(t + \tau/2, t - \tau/2) = \left(\frac{1}{2\pi}\right)^2 \int \overline{R}(t' + \tau/2, t' - \tau/2)e^{i\omega (t' - t)} e^{-i[k^*\left(-\omega + \omega' / 2\right) - k\left(\omega - \omega' / 2\right)](x-x_0)} dt' d\omega' d\omega \]  
3. EXAMPLES

As a first example consider the case where there is no dispersion. In that case

\[ k(\omega) = \omega c \]  
where \( c \) is the velocity of propagation. Hence

\[ k^*\left(-\omega + \omega' / 2\right) - k\left(\omega - \omega' / 2\right) = \frac{1}{c}\left((-\omega + \omega' / 2) - (\omega - \omega' / 2)\right) = \frac{\omega'}{c} \]  
Using Eq. (30) this leads to

\[ \overline{W}(x; t, \omega) = \overline{W}(x_0; t - (x - x_0)/c, \omega) \]  
and

\[ R_x(t + \tau/2, t - \tau/2) = R_{x_0} (t - (x - x_0)/c + \tau/2, t - (x - x_0)/c - \tau/2) \]  
\[ R_x(t_1, t_2) = R_{x_0} (t_1 - (x - x_0)/c, t_2 - (x - x_0)/c) \]  
This shows that the noise field changes as expected, namely, the statistics are the same except translated by the amount of time it takes to go from \( x_0 \) to \( x \).
Now consider white noise. We first point out that for white noise one has
\[ R(t_1, t_2) = N_0 \delta(t_1 - t_2) \] (39)
and therefore
\[ E[F^*(t - \tau)F(t + \tau/2)] = N_0 \delta(\tau) \] (40)
Substituting this into Eq. (27) we have
\[ \overline{W}_{WN}(t, \omega) = \frac{N_0}{2\pi} \] (41)
which shows that for white noise the Wigner distribution is constant in frequency for all time.

Now consider the propagation of white noise. Using Eq. (30) we have that
\[ \overline{W}(t, \omega, x) = \frac{1}{2\pi} \int \int \overline{W}(t', \omega', x_0) e^{-i\omega'(t' - t)} e^{\left[ k^2 \left( \frac{\omega + \omega'}{2} \right)^2 - k^2 \left( \frac{-\omega + \omega'}{2} \right)^2 \right]} \left( x - x_0 \right) dt' d\omega' \] (42)
\[ = \frac{1}{2\pi} \frac{N_0}{2\pi} \int \int e^{-i\omega'(t' - t)} e^{\left[ k^2 \left( \frac{-\omega + \omega'}{2} \right)^2 - k^2 \left( \frac{-\omega + \omega'}{2} \right)^2 \right]} \left( x - x_0 \right) dt' d\omega' \] (43)
Or
\[ \overline{W}(t, \omega, x) = \frac{N_0}{2\pi} e^{\left[ k^2 \left( -\omega x + k(x - x_0) \right) \right]} \] (44)
We see that if the dispersion relation is real (no attenuation) we have that the noise field remains white noise but if there is attenuation then
\[ \overline{W}(t, \omega, x) = \frac{N_0}{2\pi} e^{2k^2 \left( -\omega x + k(x - x_0) \right)} \] (45)

4. CONCLUSION

We have shown that an effective way to handle the propagation of nonstationary noise in a dispersive medium is to consider phase space methods. In a future paper we will present the results for case one described above and also the propagation of other statistical properties.

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References

Investigating scattering variability via reverberation inversion

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Abstract: This paper carries out joint inversion of reverberation and short-range propagation data to investigate the ability to resolve spatial and frequency-dependent variations in seabed scattering and geoacoustic parameters. The data were collected on the Malta Plateau, Strait of Sicily, using a towed source-receiver system as part of the BASE04 experiment. The source transmitted an 850-1750 Hz linear frequency-modulated signal at one-minute intervals along the track. The signals were recorded at a horizontal array (towed ~370 m behind the source) over a 25-s time window to include both short-range propagation data (with no signal clipping) and long-range reverberation data. The seabed properties at the experiment site are known to vary with range, with a soft, low-speed sediment layer pinching out along the track. To meaningfully resolve spatial/frequency variability via inversion requires rigorous uncertainty estimation to differentiate variability from inherent inversion uncertainties. To this end, a Bayesian inference approach is developed and Markov-chain Monte Carlo methods are applied to estimate properties of the posterior probability density (PPD) which quantify parameter estimates, uncertainties, and inter-relationships. Given the strong inter-parameter correlations inherent in reverberation data, an efficient Metropolis-Hastings sampler is applied in a principal-component parameter space, employing a proposal distribution based on a linearized PPD approximation. Statistical tests are applied to determine whether variability in parameter estimates are significant, given the inversion uncertainties.
A SHALLOW-WATER RANDOM REVERBERATION MODEL

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Abstract: Reverberation is a unwanted signal for active sonar-system. There are appeared many models described average reverberation intensity decaying and reverberation envelope statistics respectively in order to understand the reverberation phenomenon. In this paper, a phenomenological model of long-range, low-frequency reverberation in shallow water, based on reverberation intensity decaying model and random bottom scattering law, is developed to describe the statistical characteristics of the sound field scattered by sea bottom. Reverberation intensity decaying model and reverberation statistics model were combined together successfully. The random reverberation signal was simulated according to the new model and its corresponding probability density function was analysed. The result shows that: the shallow water diffuse reverberation envelope is Rayleigh distributed when the number of scattering patch is large enough.

Keywords: Reverberation, statistics, shallow water
1. INTRODUCTION

In published shallow water reverberation models[1], phenomenological models based on normal mode theory can describe the average reverberation intensity decaying well. Bucker and Morris[2] were the first ones who suggested that the reverberation field can be obtained by phenomenologically relating normal mode scattering to plane wave scattering coefficients. Later Zhang and Jin[3] extended the formulation to allow the water column to be stratified. Ellis[4] summarized this approach and extended it to be obtain time domain reverberation through ray-mode analogies. Kevin[5] developed this method by describing the temporal characteristics of monostatic reverberation as a function of source bandwidth, source-receiver depth, and the propagation characteristics of range-independent shallow water. Li[6] calculated the coherent reverberation intensity decaying by coherent summing of normal modes. Recently, Grigor’ev, etc. [7] suggested the statistical characteristics of reverberation can be described by random bottom scattering Law.

Unmatched reverberation distribution can lead to an increase in the probability of false alarm[8]. There are a lot of efforts on the reverberation statistics research. Traditionally, it is assumed that the reverberation envelope satisfied Rayleigh distribution. While Rayleigh distribution can not describe reverberation envelope for some cases when there are not enough scatterers contributing to the reverberation in given resolution cell. It is found that Non-Rayleigh distributions, such as K, Weibull, Rayleigh mixture or log-normal distribution can describe these cases, especially K distribution[9,10].

In this paper, we start with a phenomenological reverberation model, which can predict temporal narrow-band random reverberation signal and the average reverberation intensity. The random reverberation signal was simulated and it’s envelope statistics was analysed according to the established model in Section 3. Finally, discussion and conclusion were addressed in section 4.

2. PHENOMENOLOGICAL REVERBERATION THEORY

In shallow water, the bottom scattering can be considered as the dominant scattering source, especially for a relatively smooth sea surface or for a water sound speed profile with a negative gradient[11]. It is reasonable to only consider bottom backscattering when model the shallow water reverberation. And the shallow water reverberation can be framed as Figure 1.

Fig 1. Monostatic reverberation diagram

Fig 2. Scattering element dividing
Consider Pekeris waveguide, bottom depth is $H$, source depth $z_s$ and receiver depth $z_r$. Then acoustic pressure at bottom scattering element due to a point source of unit strength in middle-far range (>5$H$) can be written as[12]

$$p_{in} \approx i\pi \sum_{m=1}^{M} \phi_m(z_s)\phi_m(H)H_0^{(1)}(K_m r)$$  \hspace{1cm} (1)

where $r$ is the range, $\phi_m$ are the mode functions, $H_0^{(1)}$ is the zeroth Hankel function of the first kind, and $K_m$ is the mode wave number including attenuation,

$$K_m = k_m + i\delta_m$$  \hspace{1cm} (2)

here $\delta_m$ is image part of wave number. $M$ is the number of trapped modes in the waveguide.

According to reciprocal theorem, the acoustic pressure at receiver $z_r$ from scattering element $dS$ can be written as

$$p_{scan} \approx i\pi \sum_{m=1}^{M} \phi_m(z_r)\phi_m(H)H_0^{(1)}(K_m r)$$  \hspace{1cm} (3)

Based on the bottom backscattering phenomenological diagram, the monostatic reverberation pressure can be written as

$$p_{rev}(\omega) = \int_S p_m T_{mn} p_{scan} \sqrt{dS}$$  \hspace{1cm} (4)

where $dS$ is scattering element, $T_{mn}$ is transfer matrix between incident modes and backscattered modes.

When the phenomenological theory is used to explain reverberation average intensity decaying, $T_{mn}$ is commonly represented as Lambert Law or other empirical backscattering law. The random bottom backscattering law was used to explain the random reverberation pressure here

$$T_{mn} = \zeta_{mn}(r) F(\theta_m, \theta_n)$$  \hspace{1cm} (5)

here, $\zeta_{mn}$ is a stochastic process that describes the relative fluctuations of the field scattered from mth to nth modes, $F(\theta_m, \theta_n)$ is a deterministic function characterizing the angular redistribution of the scattered field, $\theta_m$ is grazing angle corresponding to the mth normal mode.

The scattering field given by eq.(4) is for CW wave, the scattering field of a pulse signal $s(t)$ with a spectrum $s(\omega)$ can be obtained by Fourier transform

$$p_{rev}(t) = \int d\omega [s(\omega) p_{rev}(\omega) \exp(i\omega t)]$$  \hspace{1cm} (6)

Substituting eq.(4) into eq.(6), and making use of the narrow-band approximation[1], set

$$k_m(\omega) \approx k_m(\omega_0) + (\omega - \omega_0) \frac{\partial k_m(\omega_0)}{\partial \omega}$$  \hspace{1cm} (7)

Then, we get

$$p_{rev}(t) = \int_S s(t - t_{mn}) p_{in} T_{mn} p_{scan} \sqrt{dS}$$

$$= -\pi^2 \int_S \sum_{m}^{M} \sum_{n}^{M} s(t - t_{mn}) \phi_m(z_s)\phi_m(H)H_0^{(1)}(K_m r)T_{mn} \phi_n(z_r)\phi_n(H)H_0^{(1)}(K_n r) \sqrt{dS}$$  \hspace{1cm} (8)

where

$$t_{mn} = \left[ \frac{\partial k_m}{\partial \omega} + \frac{\partial k_n}{\partial \omega} \right] r = \frac{r}{u_m} + \frac{r}{u_n}$$  \hspace{1cm} (9)

here, $u_m$ is group velocity of the $m^{th}$ normal mode.
In eq.(8), we neglect the pulse spreading caused by the \( (\partial^2/\partial \omega^2) \) km term, the modification of the pulse amplitude spreading should be considered for very long range and very wide band pulse. For such a case, it has been discussed in Ref.[5,14].

In eq.(4) and eq.(8), \( S \) is the insonified annular surface with area

\[
S = 2\pi \Delta r_{ac}
\]  

(10)

where \( \Delta r_{ac} \) is the effective width of the ring,

\[
\Delta r_{ac} = c_{ac} \tau / 2 << r
\]  

(11)

Here \( c_{ac} \) is effective value of the sound speed, and \( \tau \) is the duration of the received pulse. Assume the bottom rough surface can be described by Goff-Jordan spectrum[15].

\[
P_n(2k_0) = \pi L[1 + (2k_0L)^2]^{-3/2}
\]  

(12)

where parameter \( L \) is the horizontal correlation scale, in general, \( L \) ranges 10-20 m. Hence, in modeling the reverberation signals, one should break down the insonified surface into individual scattering patch (elementary areas) of small width \( L \), within which incident and scattered sound fields are constant.

\[
p_{rev}(t) = -\pi^2 \sum_j \sum_m \sum_n s(t-t_{mn}) \phi_m(z_s) \phi_n(H) H_{0}^{(1)}(K_m r) T_{mn} \phi_n(z_r) \phi_n(H) H_{0}^{(1)}(K_n r) \sqrt{dS_j}
\]  

(13)

here \( D \) is the number of scattering elements in backscattering area. The method of scattering element dividing can be found in Figure 2.

### 3. Reverberation Envelope Statistics

In this section, the random reverberation signal was simulated based on eq.(13), and the probability density function (PDF) of it’s envelope was analysed. In our numerical simulation example, the source function used to simulate the reverberation signal is a Gaussian impulse with amplitude \( A \), center angular frequency \( \omega_0 = 2\pi f_0 \), width \( \tau \), and delay \( t_0 \). Given by

\[
s(t) = A \exp[-\tau^2(t-t_0)^2 - i\omega_0 t]
\]  

(14)

Consider a Pekeris waveguide with water depth \( H = 100 \) m, \( c_0 = 1500 \) m/s, \( c_b = 1623 \) m/s, \( \rho_h = 1.77 \) g/cm\(^3\), and the center frequency \( f_0 = 300 \) Hz, the source depth \( z_s = 25 \) m, the receiver depth \( z_r = 50 \) m. The mode functions and eigenvalues used to evaluate the random reverberation signal are computed with KRAKEN[16]. In the Goff-Jordan spectrum, the horizontal correlation scale \( L = 20 \) m. so the area of the scattering element is about \( L \times L = 400 \) m\(^2\). For the initial signal, \( A = 1; \tau = 0.2 \) s.

The random reverberation signal \( p(t) \) was shown in Figure 3, where Rayleigh distributed amplitude parameter \( \sigma = 0.001 \) (\( \approx \sqrt{-27} \) dB ). The average reverberation intensity line \( I(t) = p^2(t) \) was calculated by using determinate transfer matrix \( T_{mn} \) instead of \( T_{mn} \).

\[
T_{mn} = \sqrt{\pi/2} \sigma F(\theta_m, \theta_n)
\]  

(15)

And make incoherent sum of each modes to get average reverberation intensity. The normalized random reverberation signal \( p_N(t) \) was gotten in the following way,

\[
p_N(t) = p(t) / \sqrt{I(t)}
\]  

(16)
The normalized random reverberation signal can be found in Figure 4. The PDF of the reverberation signal was shown in Figure 5. The result shows that the PDF of reverberation signal can be described by Rayleigh distribution with power $2\sigma_R^2 = 0.68$.

![Fig 3. Random reverberation signal and its average decaying line](image)

![Fig 4. Normalized reverberation complex envelope](image)

![Fig 5. The probability density function of the reverberation envelope](image)

4. DISCUSSION AND CONCLUSION

A phenomenological model of random reverberation in shallow water is developed with narrow-band approximation. The model can predict both average intensity and random reverberation signal. Numerical simulation result shows that the diffuse reverberation signal envelope can be described by Rayleigh distribution. This result is consistent with traditional reverberation envelope statistics.

For high-resolution active sonar systems, there may be too few scatterers in the resolution cell so that the central limit theorem cannot be satisfied. The new model developed in this paper can include this case by decreasing the number of scattering element. And its corresponding distribution envelope can also be explained by the model described in this paper, this requires the work in the future.

5. ACKNOWLEDGEMENTS

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REFERENCE


Regular Session V

Seafloor characterization
Seafloor characterization using backscattered angular responses obtained with a forward looking sonar

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Abstract: An angular analysis of backscatter echoes from the seafloor is presented. The study is based on data collected at sea with a prototype of multibeam front-scan sonar system (COSMOS - Project partially funded by the European Commission in the MAST III programme - Contract n° MAS3-CT97-0090 - DG12 - ESCY). Thanks to the front-looking geometry of the system, local angular responses of the bottom have been collected over a wide range of incidence angles. The data were collected in regions with high variability of sediments. The responses have been used to characterize the different types of seafloors. We report here the procedure for classifying the backscattered responses and the subsequent characterization. A Principal Component Analysis is performed. The responses are then projected on the first Eigen vectors. An original process is defined to identify clusters in that space. The typical responses corresponding to these clusters are exhibited and compared with backscattering strength models. Maps with different types of classified sediments are finally presented.
SonarClass: A MATLAB TOOLBOX FOR THE CLASSIFICATION OF SIDE SCAN SONAR IMAGERY, USING LOCAL TEXTURAL AND REVERBERATIONAL CHARACTERISTICS.

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Abstract: A Matlab toolbox for the acoustical classification of Side Scan Sonar imagery has been created and is presented in the current paper. SonarClass is based on a methodology that concerns extraction and analysis of local textural and reverberational characteristics and consists of 3 main stages: 1) selection of training samples, 2) feature selection and calibration according to the training samples so that the bottom classes are optimally discriminated into the feature space and 3) supervised classification of the SSS image using the selected features and a variety of classifiers. Second order (or textural) statistical parameters are derived from the well known Grey Level Co-occurrence Matrices (GLCMs) and the two-dimensional Fourier spectrum whereas first order (or tonal) statistics are simple calculations of elementary statistical features and central moment estimations. The originality of the method lays on the fact that 1. it applies automatic selection of the best combination of parameters, through an iterative optimization process and 2. it performs calibration of the offset (d) and theta (θ) parameters that control the GLCM efficiency. The main target of this methodological scheme is to ensure that the researcher has maximum supervision over the classification process and generalized classification rules can be straightforward realized. The created Matlab toolbox and the graphical interfaces are particularly comprehensible and the user has full control over the classification process. The options of overall image classification or individual bottom type detection are feasible on user’s demand and unsupervised classification abilities are also apparent.

Keywords: Sidescan sonar, texture analysis, image classification, SonarClass, GLCMs optimization
1. INTRODUCTION

Sidescan sonar has been an important tool for seafloor survey over the past few decades. Due to the highly textured appearance of sonar images, texture analysis techniques become natural choices for sidescan sonar image analysis. Grey level cooccurrence matrices (GLCMs) and Fourier Transform approaches are among the methodologies that have mostly been used for textural analysis and image classification. Statistics over GLCMs are very powerful texture descriptors but they need a sufficient amount of input parameters to be specified a priori. In works of Ph. Blondel et.al these parameters are specified by performing repeated tests until optimal separation between textural units is observed [1], while R.Jobanputra et. al. have arbitrarily set standard values [2]. G.Y. Ojeda et.al extracted various image description features using both textural and grey level first order statistical parameters and investigated their ability to differentiate classes by plotting diagrams between most of the feature combinations [3]. John Preston extracted a vast amount of features concerning many first and second order statistical parameters and then reduced the data dimensionality to only three components by using PCA [4]. Xiaoou Tang et.al. proved that Fourier Transform magnitudes contain enough texture information, if the right feature extraction algorithm is used [5].

In this study we have proceeded to three innovations: 1) grey level first order (tonal), second order (textural) and Fourier transform methodologies were all together used to extract a variety of image description features, 2) optimal feature selection is ensured through a computerized process that tests the discrimination ability of all the possible combinations between the features and 3) numerous GLCM input parameters (offsets, directions and other special treatments) are used to produce different feature values and the optimal ones are selected via innovation no 2. Tests upon artificial textures and case studies [6] showed that the proposed methodological scheme provides optimal separation between bottom classes and thus reliable classification seafloor maps can be produced.

2. METHODOLOGY OVERVIEW

The process of pattern recognition in the context of the SonarClass software involves five main steps: 1. manual selection of a limited number of small characteristic regions from each desired sea-bottom class (training samples), 2. extraction of a large number of first order (tonal) and second order (textural) statistical parameters from each training sample, 3. automatic selection of the combination of parameters that provide the highest discrimination between the acoustic types (optimization process), 4. extraction of these parameters from sub-regions throughout the whole SSS image and 5. supervised classification of the image. In practice, steps 1 to 3 compose the Calibration module while steps 4 and 5 the Classification module of the SonarClass software. SonarClass can be considered a sea bottom classification software that parallels the methodologies adopted by the broadly used TexAn [1] and QTC [4] softwares, but through a more explorative way. Five GLCM properties, four simple first order gray statistics and two 2D FFT spectrum descriptives are in use by the SonarClass software. An in depth description of them can be found in [6]. Fig. 1 describes in brief the methodological scheme adopted by the SonarClass software.

A key feature of SonarClass toolbox is the emphasis that has been given to the way that GLCM features are calculated. In particular, only the 5 more popular features out of the 12 that Haralick [8] introduced are considered. Symmetrical and normalized cooccurrence
matrices are used so that matrix elements refer to joint probability densities. Averaging the GLCMs by using four angular directions (0°, 45°, 90° and 135°) is an established method to ensure insensitivity to pattern rotation [1]. In our study a new approach is developed, according to which more angular directions are analyzed and the maximum statistical values of the θ directions are recorded apart from the averages ones. These treatments of angular directions will be referred in the text as “max θ” and “average θ” treatments. In [6] it has been proven that the “max θ” treatment very often gives much better results than the classic “average θ” treatment.

![Fig.1: An overview of the methodological scheme adopted by SonarClass.](image)

### 2.1. THE CALIBRATION MODULE

Using a large amount of features for classification purposes is not always efficient as some of them may be correlated or may reduce the discrimination of the classes within the features’ space. Data reduction methods such as Principal Component Analysis is a common technique [4] but the components that are created have no particular physical meaning and they cannot be used to train a classifier that will be applied to a different data set, thus making them applicable only for unsupervised classification purposes. Taking the previous into account, careful selection of the features is always important to optimize a classifier’s efficiency. The methodology followed in SonarClass involves 3 stages: 1) feature extraction from the training image samples, 2) estimation of the discrimination power for all the possible combinations (not only per two) between the n features and finally 4) selection of the combination that provides the highest discrimination power score. The calibration practice is firstly applied to the GLCM features, estimated for a variety of d and θ values and for both ‘max’ and ‘average’ ‘0 treatments’, as requested by the user. After the GLCM properties (d, 0 and θ treatment) and the combination of GLCM features that provide the best possible discrimination power have been found, the process is repeated including the rest of the features. Finally the top combinations of all the features are shown to the user who can decide which suits him best. The motivation for this methodological approach is that as far as we know the d and θ values are usually selected either arbitrarily by setting d = 1 pixel and θ = 4 angular directions, or by performing protracted manual tests to specify the best ones. When dealing with a large amount of features, possible combinations between the features are too many to be tested out manually. For example the possible combinations between 8 features are 255 and between 11 variables are 2047. Notwithstanding, finding out the features that provide the best between classes discrimination is vital for the optimization of the classification process. Consequently, in SonarClass, a computational procedure is employed
that automatically tests all the possible combinations between the \( n \) features and decides which one provide the higher score. Then, this combination is considered as an appropriate one for the classification process.

Discrimination power measure is provided by the cluster silhouette values. The silhouette value for each sample point is a measure of how similar that point is to points in its own cluster (class) compared to points in other clusters, and ranges from -1 to +1. The sensibility of the discrimination power to changes of the offset, \( \theta \) and \( \theta \) treatment properties can be very clearly seen in experiments that have taken place in [6] making their careful selection an issue of importance.

![Figure 2: The SonarClass software as used for creation of a bio-habitat map in Laganas Gulf, Zakinthos Island, Greece [7].](image)

The “Calibration module” of SonarClass is responsible for the implementation of the above mentioned procedure. It has a graphical user interface (Fig 3) that allows the user to control the number of offsets and directions to be included in the process and the theta treatments to be considered. In addition, each class can be analyzed individually so that different feature combinations are to be used for the classification of every bottom type (Fig 4). The top ten combinations of features and all their characteristic parameters can then be visually explored via the “EXPLORE RESULTS” tool (Fig 3). The preferred combination of features can then be selected to be used in the “Classification Module” for the classification of the raw SSS image, into as many classes as reported by the training samples.
2.2. THE CLASSIFICATION MODULE

At this point we have extracted GLCM, first order and Fourier statistical features from the sample data set and we have concluded in which features to use in order to provide optimal categorization between classes. The training of the classifier is done using the selected features as extracted from each sample region. Then the sidescan sonar image is divided into either distinct or sliding windows (sub-regions or texels), and each one is classified according to the decision rules created by the classifier. If a window consists of more than 50% white or black areas it is considered as ‘no data’ area and is excluded from the analysis.

The Classification module gives the option of supervised or unsupervised classification (Fig 4). In the first case the features to be included in the analysis are supplied by the calibration module whilst in the second the features are manually selected as well as the GLCM properties. In the “unsupervised” case, the data can be dimensionally reduced using a
variety of data reduction techniques. In each case the user can choose between a number of classifiers and decide whether the data will be standardized or not. Finally, the user selects the SSS imagery to be classified, the window sizing (in pixels) and the overlapping window area if sliding windows are desired. The results can be viewed and stored in comma delimited matrices and indexed-color georeferenced classification images.

3. CONCLUSIONS

In this paper a fully developed MATLAB toolbox for side scan sonar imagery classification has been presented. The implemented methodology involves three image analysis approaches (first order statistics, GLCMs and simple Fourier Transform approaches) and a computerized GLCM calibration and feature selection procedure. We have introduced a methodology that automatically compares all the possible combinations between the features and finds those that provide the best separability between the various sea bottoms classes, based on collected image samples. The same samples are used to train a classifier. The simultaneous use of a variety of feature extraction methodologies in combination with the calibration process has been proved to be a promising aspect. The SonarClass MATLAB toolbox and its convenient graphical interface have already been successfully used for the classification of SSS records from a variety of sea bottom environments.

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Acoustical tools comparison for the mapping of Posidonia oceanica fields: vertical single beam echosounder vs. sidescan sonar uses.

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Abstract: Posidonia oceanica fields in the "Cabo de Gata" marine natural park, located in the southeast Mediterranean Spanish coast, have been mapped by means of three different acoustical tools: a vertical singlebeam scientific echosounder, a sidescan sonar and the same side scan sonar with diminished incidence angle. Each acoustical data have been analyzed by statistical and echogram image processing methods. A selection of the measured transects have been dived and recorded with a video camera in order to validate the predictions from the extracted acoustical data. Results obtained from the statistical analysis carried out with open and alternative algorithms and image processing methods from each acoustical tool have been compared. The modified use of the sidescan sonar, by approaching the beam axis to the normal incidence, reveals as a very good method for seagrass acoustical mapping.
Regular Session VI

Signal and image processing
ANALYSIS OF ACOUSTICS SIGNALS ECHOS FROM CYLINDRICAL ELASTIC SHELLS BY HHT AND THT

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Abstract:
The interest of Time-Frequency Representation (TFR) compared to time or frequency representations, is their ability to quantitatively resolve changes in the frequency content of nonstationary signals such as underwater reflected acoustics signals. However, a limit of such TFRs is they produce, in some cases, representations that are meaningless or difficult to explain and to analyze. In this work two new TFRs based on the Empirical Mode Decomposition of Huang, called respectively Hilbert-Huang Transform (HHT) and Teager-Huang Transform (THT) are investigated. The two TFRs are applied to analyze echo signals of cylindrical tube. Resulting TFRs are compared to that of Wigner-Ville distribution. Both HHT and THT give qualitatively and quantitatively better results than the Wigner-Ville distribution and the spectrogram in terms of pertinent and accuracy time-frequency features, used in targets classification.

Keywords: Time-frequency representation, HHT, THT, Underwater signal echoes, cylindrical shells.
1. INTRODUCTION

Use of underwater sound for detecting and locating submerged targets was introduced more than 80 years ago. The problem of discrimination of immersed targets was initiated by Hoffman [1] who investigated time-domain approaches followed by Chesnut and Floyd who tested multiple frequencies based techniques [2]. Time-frequency (TF) approaches have also been used for target classification [3],[4] and have given high potentiality for discrimination between solid and hollow targets as well as for determining the target material [5]. For example in [5] a Wigner-Ville distribution (WVD) as a TF description is used. WVD has been shown to be a relevant for understanding of echo formation mechanisms and for surface waves that circumnavigate the targets [3],[4]. In [7] a Sonar target classification based on the TF projection filtering [8] has been proposed. The WVD associated to the Impulse response (IR) (acoustic response) of a Sonar target generates a TF plane (image) showing different patterns. These patterns can be classified into two categories 1) Interferences due to the bilinear nature of the WVD [6]. 2) High energy pattern: the first one, non dispersive, is associated with the specular echo on the target and the two following patterns correspond to the arrival of surfaces waves (antisymmetric Lamb waves) that circumnavigate the target [7]. The two pertinent patterns for classification are the specular reflection and the Lamb waves. The function of a TF filter is to extract from the signal to be analyzed the pertinent patterns. The filter is designed from the WVD of a reference signal and more particularly from its TF support R containing the relevant information. This region R is derived manually (isolation of the echoes by an expert operator). A limit of the WVD is the severe cross terms as indicated by the existence of negative power for some frequency ranges. Although most of these difficulties can be avoided by using proper kernel functions, the method is still Fourier based; therefore all complications associated with Fourier transform still exist. To circumvent this difficulty two new TF methods based both on the Empirical mode decomposition (EMD) of Huang [10] called Teager Huang Transform (THT) [9] and Hilbert Huang Transform (HHT) [10] are investigated to analyze IR of Sonar target. The EMD does not make any assumption about the stationnarity or the non-linearity of the analyzed signals, and avoids the interference problem of the WVD. The EMD decomposes a signal into oscillatory modes called Intrinsic Mode Functions (IMFs). The aim is to determine IMFs that characterize the Sonar target. In this paper we investigate the THT and HHT to analyze echoes signals scattered from cylindrical object. We examine their ability for characterising Lamb waves, which can propagate in the object. We compare the results of TF representations of HHT and THT to those of the WVD and spectrogram.

2. EMD-BASED APPROACHES

2.1. Hilbert-Huang Transform:

This TFR is based on the EMD and the Hilbert transform (HT). The EMD is defined by an algorithm, called sifting process, which decomposes any time series to a set of IMFs that are conform to HT [10]. Using HT, IMFs produce IFs and Instantaneous Amplitudes (IAs). The resulting energy-TF representation, called HHT, is comparable to WVD. The interest of EMD based methods is that IMFs are derived from the signal itself. This means that no artificial basis functions are used and so the extracted IFs are physically meaningful. The
EMD can be seen as a type of wavelet decomposition, whose sub-bands are built up as needed, to separate the different components of a signal $s(t)$. Each IMF replaces the signal detail, at a certain scale or frequency band [12]. The EMD picks out the highest frequency oscillation that remains in $s(t)$. An IMF satisfies two conditions:

1) Numbers of extrema, and of zeros crossings, may differ by no more than one;
2) Mean value of the envelope defined by the local maxima, and the envelope defined by the local minima, is zero.

To be successfully decomposed into IMFs, $s(t)$ must have at least two extremas, one minimum and one maximum. The sifting is repeated in order to get a true IMF that fulfills conditions (1) and (2). The result is that $s(t)$ will be decomposed into IMFs and a residual:

$$s(t) = \sum_{j=1}^{N} \text{IMF}_j(t) + r_N(t)$$

HT is applied to each IMF and associated analytic signal, $s_a(t)$ is formed. IF and IA of the IMF are given by

$$f(t) = \frac{1}{2\pi} \frac{df(t)}{dt}, a(t) = \sqrt{R_s(t) + I_s(t)}$$

where $\phi(t) = \arctan \left( \frac{I_s(t)}{R_s(t)} \right)$

$$\psi[s(t)] = s(t) - \hat{s}(t) \hat{s}(t)$$

$$f(t) \approx \frac{1}{2\pi} \sqrt{\psi[s(t)]}, a(t) \approx \frac{s(t)}{\sqrt{\psi[s(t)]}}$$

where $\hat{s}(t) = \frac{\partial}{\partial t}

2.2. Teager Huang Transform:

THT is a combination of the EMD and the Teager operator. This operator is applied to each IMF, $\psi[\text{IMF}(t)]$. To demodulate an IMF, the Energy Separation Algorithm (ESA) based on Teager operator is used [13]. ESA estimates the IF and the IA of $s(t)$ as follows:

$$\psi[s(t)] = s(t) - \hat{s}(t) \hat{s}(t)$$

$$f(t) \approx \frac{1}{2\pi} \frac{df(t)}{dt}, a(t) \approx \frac{s(t)}{\sqrt{\psi[s(t)]}}$$

3. TFRs OF ECHOES FROM CYLINDRICAL SHELLS

TFRs from echoes of cylindrical shells are presented. A tube characterized by a material and radius ratio $b/a$ (a, outer radius; b inner radius) is used. Usually specular echo is a broadband signal with high amplitude compared to Lamb waves. Since associated TFR is not easy to analyze, specular echo is cancelled before analysis. The tube of aluminium is immersed in water filled full of air with $b/a = 0.95, a = 1 \text{cm}$. Sifting results of echo $s_1(t)$ is given in Fig. 1 and the associated TFRs are shown in Figs. 2-5. These TFRs show that circumnavigate waves around and inside the tube are globally detected. However, for Smooth Pseudo WVD (SPWVD), these waves are not evidenced on the TF plan. Indeed, in SPWVD the filtering used for smoothing the representation could badly affect this last [6]. These waves can be recovered using convenient parameters which are in practice difficult to determine. As for SPWVD we can draw the same conclusions for spectrogram. TF features (Fig. 2) are not well evidenced. Except the time localisation, spectrogram features do not give pertinent information. In contrast HHT and THT give accuracy frequency values and represent
correctly the circumnavigate waves in TF plan. More particularly, HHT reveals different laws frequency for each wave packet, corresponding to different IMFs of $s_1(t)$. IMFs are shown in figure 1. The first one is noise dominant mode; its frequency is the highest one. IMFs are presented from the highest frequency mode to the lowest one. As can be seen in Figs. 6 and 7, a zooming region in time interval [1.54ms, 1.57ms] of both HHT and THT show different frequency laws, characterizing the echo. Figures 8 and 9 show spectrogram and SPWVD of signal echo, $s_2(t)$, of steel tube with $a/b = 0.97$, respectively. As for Figs. 2 and 3, these TFRs do not provide interesting features. Thus, good window parameters adapted to the analyzed signal are necessary to improve the resolution of these TFRs to get pertinent information. For HHT and THT this problem is systematically resolved by EMD which decomposes the signal with no a priori. From a zooming on HHT and THT in time interval [1.29ms, 1.37ms] one can observe different frequency laws corresponding to $s_2(t)$.

Fig. 1: Sifting results of signal, $s_1(t)$, of tube of aluminium with $a/b = 0.94$. The signal $s_1(t)$ (left up in blue), IMF1-IMF12 (from the second line at left to the before last line right) and the residue (the last line, right)

Fig. 2: Spectrogram of signal $s_1(t)$ divided into segments equal to 64, with overlap of 32 and nfft of 128.

Fig. 3: SPWVD of $s_1(t)$ (filtering window is ‘Kaiser’, with larger 3 and 127 for time and frequency).

Fig. 4: HHT of scattering signal, $s_1(t)$.

Fig. 5: THT of scattering signal, $s_1(t)$.
Fig. 6: A zooming in 1.54ms to 1.57ms time interval on HHT

Fig. 7: A zooming in 1.54ms to 1.57ms time interval on THT

Fig. 8: Spectrogram of signal $s_2(t)$ divided into segments equal to 64, with overlap of 32 and nfft of 128.

Fig. 9: SP WVD of $s_2(t)$ (filtering window is ‘Kaiser’, with larger 3 and 127 for time and frequency).

Fig. 10: THT of scattering signal, $s_3(t)$.

Fig. 11: THT of scattering signal, $s_2(t)$. 
Fig. 12: A zooming in 1.29ms to 1.37ms time interval on HHT  
Fig. 13: A zooming in 1.29ms to 1.37ms time interval on THT

ACKNOWLEDGEMENTS

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REFERENCES

New Cubic Cross filter detector for multi beam data recorded with DIDSON acoustic camera

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Abstract: Target detection in data from the DIDSON (dual-frequency identification sonar) is a challenge. Due to high frequency and many narrow beams, the system can take series of acoustic pictures of passing fish. The entire body can be seen, but are often blurred and unclear in individual pictures. It is only when series of pictures are played like a film one can see that the target actually is a swimming fish. Hence, a good target detector should not be based on individual frames, but on using series of pictures.

The Cross filter detector was originally designed for single, dual and split beam systems, to detect targets under very low signal to noise ratio conditions. In order to operate on multi beam data, we have added an extra dimension to the detecting filters. This resulted in a detector that takes into account both the time, range, and width dimensions in the data during the detection process. Preliminary tests of cubic detector were performed with various parameter settings.

Keywords: Fish detection, target tracking, signal-to-noise, cross filter, data filters, behaviour, DIDSON
1. INTRODUCTION

The DIDSON (dual-frequency identification sonar) [1] has become popular in many fisheries research applications. The ability to produce pictures from the moving fish, to provide an alternative sizing method to the sensitive TS measure, and the ability to “see” targets between rocks on the bottom in horizontal applications are among the reasons to its raising popularity. There are, however, many challenges related to the use of the device like the lack of calibration and the missing position information in one domain. Another challenge is the problems with the automatic analysis. Manual fish counting by watching through the many hours of recorded data is extremely time consuming. Generating echograms and looking through them manually is faster, but the main hope is always the automatic counting button. Common approaches for automatic counting algorithms in data from DIDSON are based on various image enhancement operators, background subtraction, threshold and tracking of isolated echoes [2].

2. THE ORIGINAL CROSSFILTER DETECTOR

Automatic counting methods for fish in data from split beam echo sounders is commonly based on single echo detection (SED) [3] and multi target tracking [4]. In shallow river applications we experienced that this approach did not work due to problems with the SED [5]. We needed another method, and started by asking why it was possible for human beings to see and count fish traces in the echograms when the SED and tracker could not. Three reasons were noted. a) We look at more than one ping at a time, b) we compared the trace with the surrounding background signal, and c) we evaluate and discard traces not looking like fish traces.

By studying the echo intensity and frequency specter along horizontal and vertical lines through various echo phenomena, we found that different phenomena had different “spectral fingerprints”. A fish trace has low frequency components along a horizontal line and high frequency components along a vertical line through the trace. Other echoes such as boat wakes, bottom, cavitations, electric noise etc. have other spectral fingerprints.

An arrangement of two filters followed by a comparator could thereby be applied to detect or remove various kinds of echoes from the data. The filters and comparator fulfilled requirement (a) and (b) since echo intensity from multiple pings and from the background is involved. An evaluator was added to fulfill requirement (c), testing on size and shape criteria and sorting out unwanted detections. The detector outputs entire traces, thus additional tracking is not necessary. Since the filters tended to form a cross, we named it a Cross filter detector. The detector was patented [6], and published [7].

![Figure 1. Simplified Cross filter detector principle](image)
3. MATERIAL AND METHODS

The new Cross filter detector algorithm was written in Pascal with Borland’s Delphi and linked into the Sonar5 post processing tool for testing [8].

Test data was recorded in the Rimov reservoir (Czech Republic) during summer 2007. Bream was fed every evening at the same time and place during the week before the recording. The bream was also fed during recording. Water depth at the transducer site was about 1 m and the DIDON SR unit set up in HF mode and placed mid water slightly tilted down. Small bream (standard length 25-40 cm) were seen milling around in the beam hunting for the food at depth 1-2 m. The signal to noise ratio was low due to the size of the fish and reflection from bottom. The bottom structure and shadows from the fish on the bottom were clearly seen.

Adapting the Cross filter detector to DIDSON

Data from DIDSON can be used to generate ordinary echograms. A fish passing through the DIDSON sound beam will produce similar echo traces in this echogram as a fish passing through the beam from a split beam echo sounder (Figure 2). We can apply the original Cross filter detector to DIDSON data if we first produce ordinary echograms. However, as soon as there is more than one target in the beam at the same time and range, we run into similar problems as with split beam echo sounders. The targets mix and can not be separated or detected well.

Including information from the individual beam can improve this. More than one echo can be seen at the same range and the shape and size of the echoes provide additional information which can be used to improve detection.

Cubical filters

The filters were extended with one dimension into cubical running window filter operators, running along the frame, beam, and range domains. Mean, median, and dilation operators were implemented. Eq. 1 shows the mean filter operator processing frame f, beam b, and range r for a filter with dimensions F, B and R. The operation has to be repeated for all possible f, b, and r.

\[
Out[f, b, r] = \frac{1}{FBR} \sum_{i=-d(F-1)/2}^{f+(F-1)/2} \sum_{j=-d(B-1)/2}^{b+(B-1)/2} \sum_{k=-d(R-1)/2}^{r+(R-1)/2} In[i, j, k]
\]

Eq. 1
A passing fish may produce a trace lasting for a few frames, cutting a few beams, and covering a few range bins. To reduce variation and noise without suppressing the fish, a low pass filter must be set up with a cutoff frequency higher than the fish specter. Reducing variation in the foreground signal is important to prevent the fish echo from being split into multiple small echoes and to prevent spikes in the background from being detected as fish. A typical foreground filter size can be \( \{F,B,R\} = \{3, 5, 3\} \), where F, B, R indicates frame, beam and range respectively. Frame rate, fish size, swimming speed, and sampling regime will influence on the frequency specter and on the optimal operator size.

**Detecting the background**

To suppress fish and detect background, a low pass filter must cut below the main part of the frequency components from the fish trace. A large filter window e.g., \( \{F,B,R\} = \{51, 21, 51\} \) can do this, but it will be a computation intensive process. Normally it is sufficient to apply the suppressing background filter to the domain where the target has its highest frequency specter, such as the range domain.

**Alternative background filter**

An approximated median filter [9] was implemented as an alternative to the running window filters. Here the intensity in each sample in the current frame is compared with the current state of the background filter matrix. If a sample has higher or lower intensity, the matching sample in the background filter is increased or decreased with a constant value. The value determines how fast the background will approach the current frame intensity.

**The comparator and evaluator**

The comparator compares the output from the two filter systems. Two options were implemented and tested, threshold and difference. For the threshold, a constant offset value is added to the output from the background filter, and the result is tested against the output from the foreground filter. Offset is measured in the same byte unit as the intensity values stored in the DIDSON ddf file. Detections are formed when the foreground filter output is higher than the background intensity. With the difference, the comparator form detections when the differences between the two outputs exceed a user defined offset value. Note that difference combined with the approximated median turn the Cross filter detector into something similar to a traditional motion detector.

The output from the comparator contains echoes consisting of neighbor samples forming 2D clusters in the beam and range domain (Figure 3, left). For larger fish, the clusters take on the shape of the fish. The evaluator extract features like perimeter length, width, height, area and elongation from the clusters and compare these with user defined min and max values to separate fish echoes from noise.

*Figure 3 A slowly passing fish form one single cluster of samples (left). Increased speed break the cluster into individual observations. (right).*
4. RESULTS AND DISCUSSION

Tracking is needed if the fish is too small or moves to fast

The original Cross filter detector obtains the entire fish trace by directly connecting neighbor samples in the output from the comparator. We tested this approach on the DIDSON data as well, tempting to detect the fish as entire 3D clusters. (Figure 3, left). However, since the individual beams forming the DIDSON sound field are thin, fish traces easily split up into individual observations if the targets move too fast relative to the frame rate (Figure 3, right). Hence, some sort of a tracker was needed. We tested the following methods.

Growing echoes: This method nearly resembles the original detector in that an echo is fattened until it starts to overlap some of the samples from the echo from the same fish in the next frame. If two different echoes are overlapped, the closest is selected.

Fish shape: Traditional point tracers have a problem in that they cannot predict the next echo position before the second echo has been found. Here DIDSON data has an advantage. At least for larger fish, the shape of the echo may point in the swimming direction. We used linear regression on the points along the center of gravity line through the fish echoes, and let the algorithm search a user defined distance along this line for the next echo.

Alpha Beta tracker: The alpha beta tracker uses a simplified version of the Kalman filter as a predictor. We estimated center of gravity points for all detected DIDSON echoes passed the points onto Sonar5’s existing Alpha Beta tracker.

Final implementation

Figure 4 show the final detector controller as implemented in Sonar5. Parameters for the individual steps are hidden in separate dialogs to help keeping the overview of the process. When a show box is checked, the film viewer will play the output from the checked element. Show for box 1, 2 and 3 are mutual exclusive, while final detections from step 4 can be overlaid by the output form any of the others. If tracking is checked on, all tracked targets are transferred further to a classification and storage unit. This is not described any further here.

Testing the detector

Targets where tracked manually and compared with the automatic detections. Manual tracking was difficult and time consuming due to the high number and behavior of the fish,
and the low signal to noise ratio. There were many situations where we were uncertain about whether a dot should be counted as fish or not. The detector was trimmed as described below and set to count the fish automatically data.

The results are presented in Table 1 and Figure 5. The table shows that the detector counted slightly more fish than we did. Checking the actual detections showed that there, in most situations simply was a question of interpretation, and that the dots found by the detector very well could be fish. In a few situations, the detector had detected two echoes from the fish. This was most profound for fish passing very close to the transducer. If we look at the result in Figure 5, we see a fairly stable counting situation, except for the last part close to frame nr 3000. Checking the data show that the increase and decrease is correct except for the very high peak. Something shadowed the transducer for a short period of time and when the transducer again starts to see fish, a burst of false detections are generated. This is probably an artifact caused by the approximated mean operator.

<table>
<thead>
<tr>
<th>Frame nr</th>
<th>251</th>
<th>289</th>
<th>354</th>
<th>390</th>
<th>400</th>
<th>517</th>
<th>564</th>
</tr>
</thead>
<tbody>
<tr>
<td>Manuel</td>
<td>18</td>
<td>15</td>
<td>13</td>
<td>26</td>
<td>17</td>
<td>22</td>
<td>16</td>
</tr>
<tr>
<td>Automatic</td>
<td>22</td>
<td>19</td>
<td>19</td>
<td>27</td>
<td>25</td>
<td>27</td>
<td>25</td>
</tr>
</tbody>
</table>

Table 1 Counting result for random selected frames.

Figure 5. Absolute and smoothed number of detections for the file 2007-07-05_205000_HF.ddf.

Trimming the parameters

The foreground filter was trimmed until the fish was most visible relative to the surrounding noise. Since the fish are small and moving relatively fast, only a small filter was expected to work. We found that a median filter with size \( \{F,B,R\} = \{3, 1, 1\} \) worked highlighted the tracks best.

Background and comparator setting: We tested a mean filter, a mean filter combined with approximated median, and approximated median alone. Approximated median was tested with the comparator set up both for threshold and difference. The settings were tuned while the detector was playing and presenting the comparator output. Fine tuning was done by stepping one frame at a time. A mean filter with size \( \{3, 3, 1\} \) followed by the approximated median operator, and the comparator set up with a threshold and an offset of 20 gave the best results.

The evaluator was trimmed with the viewer presenting the original input data overlaid the output form the evaluator. Each evaluator criterion was trimmed individually with the other turned off, until the noise detections were discarded as well as possible. Afterwards, the best working criteria was turned on. This gave a minimum perimeter length of 7 samples and a min echo width of 3 beams.
The three trackers were tested. Due to small fish with relative high speed it was not a surprise that best results where obtained with the alpha beta tracker. Figure 6 show a brief example of the output form the tracker working on detections from the Crossfilter detector.

Figure 6. Left Echogram with tracks detected by the cubic Cross filter detector. Right DIDSON viewer plotting the output from the evaluator.

5. ACKNOWLEDGEMENTS

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REFERENCES

DETECTION PERFORMANCE OF THE FRACTIONAL FOURIER TRANSFORM (CHIRP FFT) FOR FREQUENCY MODULATED SIGNALS

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The Fractional Fourier transform (FrFT) is well suited to the processing of linear frequency modulated signals. Amongst the abilities it potentially offers are: the detection of signals at lower SNRs, better signal separation and the possibility of estimating the chirp rate of a signal. This work concentrates on using such a transform for detecting frequency modulated signals. For the detection problem the magnitude of the FrFT is sufficient and consequently the algorithm can be implemented efficiently by modifying a standard FFT, this implementation we shall refer to as a ‘Chirp-FFT’. This study considers the performance of the Chirp-FFT routine compared to the FFT routine for the detection of quiet underwater signals of both biological and man-made origin whose chirp rates are unknown a priori. Critically, the performance analysis here includes the effect the algorithms have on the background noise – a factor overlooked in other similar studies. The comparison is made on the basis of real data. It is shown that, unsurprisingly, the detection advantage offered by the Chirp-FFT is very much dependent on the character of the signal, but for the examples considered the gain in performance is in the range 0 - 3 dB.

Keywords: Fractional Fourier transform, detection, sonar processing, spectrogram, frequency modulation, chirp transform.
1. THE CHIRP-FFT

The FFT exploits a basis set consisting of complex sinusoids, consequently it efficiently represents signals that are tonal or near tonal. For frequency modulated signals, the FFT is sub-optimal. This is because under such circumstances the energy of the signal is not represented in the FFT by a small number of complex sinusoidal basis functions, but by a combination of several basis functions.

An established alternative to the FFT is the Fractional Fourier Transform (FrFT) [1-3]. The FrFT creates a signal representation in which the basis functions are linearly chirped complex sinusoids and is tailored to the processing of signals containing linear frequency modulations. One can efficiently implement the FrFT in several ways [2]. In this work we shall only consider the problem of signal detection based on the (squared) magnitude of the transform and so we shall use an intuitively appealing and efficient implementation of the FrFT, referred to as the Chirp-FFT [4]. This consists of pre-multiplying the signal by a synthetic linear FM signal prior to performing an FFT; if the sweep rate of the synthetic signal is chosen appropriately then the results of this product is a signal that is nearly tonal and so suitable for analysis with the FFT. In essence, if the signal contains an up-chirp at a particular rate, then by multiplying by a down-chirp at the same rate, the combined signal has a chirp-rate which is zero, i.e. it is a sinusoid. It is important that mean frequency of the multiplying chirp is constructed to be zero to ensure that multiplication by the chirp does not affect the centre frequency of the signal. For this transformation to be effective the analytic form [5] of the input signal must be used.

Equation 1 below, defines the discrete form of the Chirp-FFT. The analytic form, \( \hat{x}(n) \), of the input signal, \( x(n) \), is obtained using the Hilbert transform. This is then multiplied by a chirp \( c(n; \alpha) \), where \( \alpha \) is the chirp-rate, before taking the FFT. Ideally the chirp-rate \( \alpha \) should be selected to coincide with the chirp-rate of the original signal \( x(n) \).

\[
X(k; \alpha) = \sum_{n=0}^{N-1} c(n; \alpha) \hat{x}(n)e^{-2\pi ink/N}
\]

In most cases any chirp-rates present in the signal are not known a priori, in which case one approach is to compute a set of Chirp-FFTs using a range of chirp-rates. The Chirp-FFT which yields the largest peak value is then selected as the output [5,6]. Fig 1. is a representation of the magnitude of the Chirp-FFTs computed using different chirp rates, where horizontal lines correspond to spectra from Chirp-FFTs with different chirp-rates. The Chirp-FFT which produces the largest peak in the plot corresponds to a chirp-rate of 60 Hz/s and a centre frequency 50 Hz. These are the chirp rate and centre frequency of the synthetic input signal. The FFT of the signal is represented in this plot as the line corresponding to a chirp rate of 0 Hz/s. The data in this region are not concentrated in a few frequency bins but spread (smeared) across several bins.
2. THE CHIRP-FFT SPECTROGRAM

The concept of a spectrogram can be extended by replacing the FFT with the Chirp-FFT [6]. This requires one to compute a Chirp-FFT for every time window, with the chirp-rate being recomputed independently for each window. Fig. 2 shows example spectrograms based on the FFT and Chirp-FFT for a synthetic chirp embedded in Gaussian white noise, the chirp has a quadratic instantaneous frequency law. The estimated chirp-rate derived from the Chirp-FFT for this signal is also depicted. From the figure it is evident that the FFT produces its highest output level when the signal is nearly tonal close to \( t = 0.5 \), in this region the signal energy is concentrated into a small number of frequency bins. As the signal’s chirp rate increases the signal energy occupies more frequency bins resulting in a reduction in the peak level. The Chirp-FFT is more robust to these phenomena; it is evident from the central frame in Fig. 2 that signal energy is well concentrated over the duration of the signal irrespective of chirp-rate. It is noteworthy that the Chirp-FFT spectrogram has altered the appearance of the background noise, introducing visual correlations coinciding with the direction of the chirping signal.

3. CHIRP-FFT NORMALISATION

One has to take considerable care when comparing the FFT against the Chirp-FFT using different window sizes. Such a comparison should be based on a metric which relates performance when presented with a signal in noise to performance in a noise only environment. When the input to the algorithms is noise (Gaussian and white) the magnitudes of both transforms are non-Gaussian. The statistics of the FFT in this case are well characterised, in particular, the squared magnitude of the FFT of white Gaussian noise is known to conform to an exponential distribution [7]. The statistical distribution of the Chirp-FFT for a single (known) chirp-rate is the same exponential distribution. However, the statistical character of the output of the Chirp-FFT, in the case where the chirp-rate is unknown and has to be estimated from the data, has not been studied. As we shall...
demonstrate, the squared magnitude of the optimal Chirp-FFT (obtained by considering a set of chirp rates and selecting the best rate) has a statistical distribution which deviates significantly from an exponential distribution. The consequence is that when comparing the two methods extra care is required because of their different behaviours in noise only environments. This is important since the statistics’ of the algorithm output in the noise only case dictates the threshold levels used to generate a specified false alarm rate. One principled approach on which to base a comparison is to measure performance relative to a threshold, where that threshold is set to achieve the same (specified) false alarm rate for both algorithms. The underlying detection process consists of a simple threshold applied to the output of the transform, i.e. a detection is made whenever the algorithm output exceeds the threshold value. The thresholds are applied to each time frequency cell individually. In practice such detections are of limited operational use, since detections in isolated time-frequency cells do not provide sufficient information regarding the character of the signal. In a realistic system an alarm would probably not be triggered until a sequence of such detections had been made and identified as having appropriate characteristics. Consequently the false alarm rates used here may appear to be high when compared to operationally acceptable false alarm rates.

Fig. 2: Comparison of spectrogram (top), Chirp-FFT spectrogram (middle) and estimated chirp-rate (bottom) for a synthetic signal with a quadratic instantaneous frequency law
\[ e^{2 \pi (t - 2048) + 3072r^2} \]

Fig. 3 has been obtained through Montê-Carlo simulation. The plots illustrate the probability of false alarm (the probability that a cell in the Chirp-FFT will exceed the given value) against spectral level. From these plots we can identify the spectral level associated with a specified false alarm rate. The difference between this level and the level associated with the FFT is the correction factor that needs to be applied to equalise the false alarm rates of the two methods and so render the comparison fair. Table 1 summarises these correction factors.

<table>
<thead>
<tr>
<th>( N )</th>
<th>256</th>
<th>512</th>
<th>1024</th>
<th>2048</th>
<th>4096</th>
<th>8192</th>
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<tr>
<td>False Alarm Rate</td>
<td>( 1 \times 10^4 )</td>
<td>1.1</td>
<td>1.4</td>
<td>1.8</td>
<td>2.0</td>
<td>2.2</td>
</tr>
<tr>
<td></td>
<td>( 1 \times 10^5 )</td>
<td>1.0</td>
<td>1.2</td>
<td>1.5</td>
<td>1.7</td>
<td>1.9</td>
</tr>
</tbody>
</table>
Table 1: Correction factors in dB

![Correction factors in dB graph]

Fig. 3: Results of Monte-Carlo simulations of Chirp-FFT probabilities of false alarm

4. RESULTS FOR THE BOTTLENOSE DOLPHIN SIGNAL

Fig. 4 and Fig. 5 illustrate the analysis on an example section of the whistle of a bottlenose dolphin (*Tursiops truncatus*) represented by four FFT spectrograms and four Chirp-FFT spectrograms respectively. The figures show the representations for different window sizes of 256 points to 2048 points; the overlap is fixed at 87.5%. As the noise statistics have been normalised, the peak levels from the Chirp FFT can be directly compared to that of the FFT to give an indication of signal-to-noise performance. The results in Table 2 show that for this signal the Chirp-FFT offers SNR gain over the FFT of 2.2 dB. The chirp-rate is estimated to be 54300 Hz/s.

This methodology has been applied to a wider range of narrow-band marine mammal vocalisations where it generally offers a similar level of SNR gain [8].

<table>
<thead>
<tr>
<th>$f_c = 148100$ Hz</th>
<th>$N$</th>
<th>256</th>
<th>512</th>
<th>1024</th>
<th>2048</th>
<th>4096</th>
<th>8192</th>
</tr>
</thead>
<tbody>
<tr>
<td>Estimated chirp-rate (Hz/s)</td>
<td>3rd International Conference &amp; Exhibition on &quot;Underwater Acoustic Measurements: Technologies &amp; Results&quot;</td>
<td>-1340000</td>
<td>-167000</td>
<td>-126000</td>
<td>-41800</td>
<td>-36600</td>
<td>-41200</td>
</tr>
<tr>
<td>FFT Level (dB)</td>
<td>0.6</td>
<td>2.4</td>
<td>3.0</td>
<td>3.2</td>
<td>2.8</td>
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<tr>
<td>Chirp-FFT Level (dB)</td>
<td>0.1</td>
<td>1.2</td>
<td>2</td>
<td>3.2</td>
<td>5.4</td>
<td>3.8</td>
<td></td>
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<tr>
<td>Chirp-FFT SNR gain (dB)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2.2</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 2: Bottlenose dolphin results
5. REFERENCES


MULTISTATIC PROCESSING AND TRACKING OF UNDERWATER TARGET USING AUTONOMOUS UNDERWATER VEHICLES

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Abstract: This paper discusses the use of distributed autonomous underwater vehicles (AUVs) for multistatic processing and tracking of underwater target. A localized processing chain is implemented on each AUV to generate the contact report of an underwater target that is transmitted to a command center via underwater acoustic communications (UWA). The contact reports from several AUVs are combined together using data fusion to create the fused report for formulating the target track solution. The experimental results obtained from the recent Generic Littoral Interoperable Network Technology (GLINT) 2008 experiment illustrates this concept and provides the discussion for further experimentation.

Keywords: Multistatic processing, multistatic tracking, autonomous underwater vehicle (AUV), data fusion
1. INTRODUCTION

The detection and localization of underwater target using active monostatic system can be difficult in situations where you have very low acoustical backscattering. To mitigate this problem, the active multistatic system is commonly employed as it offers multiple angle observations that enhance the performance. However, operating the multistatic platforms on conventional ships can be costly. More importantly, the exposure of these platforms to possible threats should always be minimized. For these reasons, there is a need to explore unmanned platforms for multistatics, particularly those capable of a certain level of autonomous maneuverability, provide ease of rapid insertion and offer effective sensing. The use of distributed autonomous underwater vehicles (AUVs) for active multistatic system is particularly attractive. However, with a much smaller payload, there is a general restriction on the processing power in each platform. There is also a need to link up these distributed AUVs together with a command centre via underwater acoustic communications (UWA) to provide a multistatic solution. In this paper, we will first start by exploring the feasibility of processing the multistatic data off-line and determine the information exchange between the AUV and the command center. In future work, we will explore the possibility of real-time processing onboard each platform and the usage of UWA to exchange information between the platforms.

2. PROBLEM FORMULATION

We consider the case where there are two underwater sensors, active sonar denoted by $S_1$ and passive receiver denoted by $S_2$, and one underwater target denoted by $T_1$. The objective here is to estimate the target state of $T_1$ at any time measurement number $k$ given the acoustic measurements obtained by $S_1$ and $S_2$,

$$X_{i1}[k] = \begin{bmatrix} x_{i1}[k] & y_{i1}[k] & v_{x_{i1}}[k] & v_{y_{i1}}[k] \end{bmatrix}^T.$$

Here, $x_{i1}[k]$ and $y_{i1}[k]$ are the $x$ and $y$ positions of $T_1$, and $v_{x_{i1}}[k]$ and $v_{y_{i1}}[k]$ are the corresponding speeds.

The target state of $T_1$ can be inferred from the monostatic ranging carried by the active sonar $S_1$ as shown in Fig. 1. Here, the active acoustic source is co-located with the sensor array, which listens to the target echo. The monostatic range $r_{s1t1}[k]$ measured will determine a circular locus of probable $T_1$ positions, called the ambiguity circle. The direction-of-arrival (DOA) $\theta_{s1t1}[k]$ measured will then specify two probable $T_1$ positions on the circle. The velocity of $T_1$ can be deduced from the target Doppler, which is readily available by measuring the target echo frequency $F_{s1t1}[k]$. The details of how these measurements are obtained are provided in [1]. These measurements can be readily compared with the true values computed from the expressions in [2].

The target state of $T_1$ can also be inferred from the bistatic ranging of passive receiver $S_2$ as illustrated in Fig. 1. Here, the sensor array is passively listening to the active acoustic source transmissions either directly from $S_1$, or indirectly reflected from $T_1$. The bistatic range $r_{s2t1}[k]$ measured will determine an elliptical locus of probable $T_1$ positions, called the ambiguity ellipse. The DOA $\theta_{s2t1}[k]$ will then specify two probable $T_1$ positions on the
ellipse. Similarly, the velocity of $T_1$ can be deduced from the target Doppler, which is readily available by measuring the frequency of the indirect active acoustic source transmission $F_{s1t1s2}[k]$. Both [1] and [2] also provide detailed information on the measurements and their true values.

From Fig. 1, each of the sensors is able to estimate the target state of $T_1$, but with an undesirable ghost. The estimated state will form the contact report transmitted to a command centre. The combined reports can be used to remove the ghosts by simple cross-fixing [2]. More importantly, the likelihood of the estimated target state can be ascertained taking into account the detection statistics of both sensors [2][3]. This will help in creating the fused report that will be subsequently used to formulate the target track solution.

The definition of multistatics simply extends the above problem to more complex cases. Examples include cases with (i) one active acoustic source and several passive receivers, (ii) few active acoustic sources with several passive receivers, and (iii) many other variants. In this paper, we are particularly interested in multistatics using distributed AUVs with passive receivers that are linked together via UWA.

$$\begin{align*}
    r_{s1t1}[k] & : \text{monostatic range, } S_1 \text{ to } T_1 \\
    r_{t1s2}[k] & : \text{range, } T_1 \text{ to } S_2 \\
    r_{s1s2}[k] & : \text{range, } S_1 \text{ to } S_2 \\
    r_{s1t1s2}[k] & : \text{bistatic range, } r_{s1t1}[k] + r_{t1s2}[k] \\
    \theta_{s1t1}[k] & : \text{DOA of } T_1 \text{ at } S_1 \\
    \theta_{s2t1}[k] & : \text{DOA of } T_1 \text{ at } S_2 \\
    \theta_{s2s1}[k] & : \text{DOA of } S_1 \text{ at } S_2 \\
    \gamma[k] & : \text{separation angle, } \angle S_1 S_2 T_1 \\
    F_{s1t1s1}[k] & : \text{echo frequency over path } S_1 T_1 S_1 \\
    F_{s1t1s2}[k] & : \text{echo frequency over path } S_1 T_1 S_2 
\end{align*}$$

Fig. 1: Illustration of monostatic and bistatic ranging by $S_1$ and $S_2$ respectively

### 3. EXPERIMENTAL SETUP

To evaluate the feasibility of multistatic processing and tracking using distributed AUVs, the Generic Littoral Interoperable Network Technology (GLINT) 2008 experiment was conducted at Pianosa, Italy from Jul to Aug 2008. The experimental assets involved the (i) NRV Alliance that functioned as a command centre and deployed with an active acoustic source (active source ($S_1$)), (ii) Unicorn AUV that towed the DURIP sensor array (passive receiver 1 ($S_2$)), (iii) OEX AUV that towed the SLITA sensor array (passive receiver 2 ($S_3$)), and (iv) CRV Leonardo that deployed an echo-repeater to simulate active insonification of a target (target ($T_1$)). Data acquisitions were carried out on the AUVs, but the processing was done off-line.

Hyperbolic frequency modulated (HFM) pulse and continuous wave (CW) active sonar pulse signals were transmitted from $S_1$. The HFM pulse signal was selected because of the pulse compression property that provides good range resolution, and the lower coupling effect of the errors in the range and Doppler estimations compared with the standard linear frequency modulated (LFM) pulse signal [4]. Although the CW pulse signal offers poor range resolution, the absence of the abovementioned coupling effect makes it preferable for Doppler estimation [4].
4. PROCESSING CHAINS

Two processing chains, namely the localized chain at each AUV ($S_2$ and $S_3$) and the centralized chain at command center ($S_1$), were set up off-line. The beamformers and the matched filters in the localized chain, as shown in Fig. 2, were implemented as described in [5] and [4] respectively.

The contact reports from both sensors ($S_2$ and $S_3$) were generated in a format suitable for UWA. These reports were combined at the centralized chain and assigned on a discretized four-dimensional contact grid of $x$ and $y$ cells, and $vx$ and $vy$ cells [2]. Cells with contacts from both sensors were flagged out, while those with contact from one sensor were assumed ghosts [2]. Thereafter, the fusion grid was created in the process of spatial fusion where cells with two contacts were assigned with higher likelihood, taking into account the detection statistics of both sensors [2][3]. The fusion grid assumed the same dimensional structure as the contact grid but with calculated likelihood values [2]. Target track solution can then be formulated using the fusion grid. However, the target tracking algorithm is still undergoing development at the time of submitting this paper and the corresponding results will not be discussed here. The reference in [3] provides an extensive selection of possible tracking algorithms.

![Diagram of processing chains](image)

*Fig. 2: Localized processing chain at each AUV [processing chain at $S_2$ illustrated here]*

5. DATA ANALYSIS

For purpose of illustration, a particular experimental run on 07 Aug 2008 is used. The ground truth for this run is given in Fig. 3. Using the localized processing chain in Fig. 2, the contact reports from the DURIP ($S_2$) and SLITA ($S_3$) sensor arrays were obtained. The results for the DURIP array are depicted in Fig. 4 where the measurements and calculated values are plotted. The calculated values are computed using the ground truth in Fig. 3.
Clearly, we see close matches between the measurements and calculated values for the bistatic ranges, DOAs and echo frequencies. The only exception is the bistatic range for time before 1300 seconds. This actually corresponds to the case when Leonardo ($T_1$) was stationary, where the GPS log was unfortunately corrupted.

The contact grid generated by the centralized processing chain is shown in Fig. 5 for a selected time measurement number $k$. Here, $k$ increments in a time block of pulse repetition interval (PRI). Cells with contacts from both DURIP and SLITA arrays are flagged out as “0” on the contact grid of $x$ and $y$ cells, and $vx$ and $vy$ cells. These cells are then assigned with higher likelihood values shown in the corresponding fusion grid in Fig. 6. From the fusion grid, we also see that the ghost contacts are assigned with much lower likelihood values. The ground truth of Leonardo ($T_1$) is plotted blue in both Fig. 5 and Fig. 6. Clearly, we see that the $x$ and $y$ positions and the $vx$ and $vy$ velocities of $T_1$ are rather accurately estimated. Ghost and/or false contacts in both the $x$ and $y$ positions and the $vx$ and $vy$ velocities can be easily removed using target tracking algorithm that associates the fusion grids across time measurement number $k$ [3]. By combining the estimated $x$ and $y$ positions and $vx$ and $vy$ velocities of $T_1$, the target state of $T_1$ in (1) is approximated.

6. RECOMMENDATIONS

Following the successful implementation and experimental testing of the off-line processing chains, the next task is to develop the localized processing chain in Fig. 2 to real-time. Work has already embarked to develop each AUV for the GLINT 2009 experiment, planned at Pianosa, Italy from Jun to Jul 2009. More multistatic runs, particularly those with moving target, will be planned to ascertain the Doppler performance and establish better detection statistics for each sensor array. We also hope to demonstrate the generation of bistatic contact reports to activate simple AUV maneuvering behaviours that adaptively optimize the multistatic performance, including synchronized swimming of two or more AUVs. The communication requirements for the contact reports between the AUVs and the command center will be examined as well.

At the same time, the target tracking algorithm of the fusion grids across time measurement number $k$ will be studied and implemented. With more data, rigorous experimental testing will be carried out to evaluate the multistatic processing and tracking performance quantitatively.

7. CONCLUSION

In this paper, we have explored the feasibility of distributed AUVs for multistatic processing and tracking of underwater target. Specifically, the GLINT 2008 experimental setup was used for illustration. Both the off-line localized and centralized processing chains on each AUV and command centre, respectively, were presented. A particular experimental run here clearly illustrated close matches between the measurements and calculated values for the contact reports in terms of bistatic ranges, DOAs and echo frequencies. By combining these reports from both AUVs, the ghost contacts were removed and the target state was estimated. Following this paper, the GLINT 2009 experiment has been planned to explore real-time localized processing chain and to provide an opportunity to collect more data, particularly those with moving target, to ascertain the Doppler performance and establish better detection statistics for each sensor array.
8. ACKNOWLEDGEMENTS

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REFERENCES


*Fig. 3: Ground truth (x and y grid) of assets for experimental run on 07 Aug 2008*
Fig. 4: Measurements (plotted in magenta markers) from DURIP sensor array ($S_2$) compared with calculated values (plotted in orange lines).

Fig. 5: Contact grid obtained at $k = 78$ by combining contact reports from $S_2$ and $S_3$.

Fig. 6: Fusion grid obtained at $k = 78$ by fusing contact grid.

Darker shading implies higher likelihood values.
AMPLITUD MODULATION OF UNDERWATER NOISE PRODUCED BY SEAGOING VESSELS

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Abstract. The amplitude modulation of underwater noise produced by seagoing vessels is
classified for the classification of ships. Due to the analyses of amplitude mod-
ulation of generating noise the following parameters of the ships can be obtained: the type of
ship, her displacement, the type of engine, the amount of paddling grosses, the amount of
screw propeller blades, the velocity of the ship, the moment of course changing or velocity
changing.

The purpose of the report is to describe the results of the experimental study of the basic cha-
racteristics of the amplitude-modulated noise produced by a ship in application to two types
of modulation:
– the modulation caused by the rotation of the propeller shaft and the screw-propeller;
– the modulation caused by ship roll and pitch in waves.

The list of investigated characteristics of the amplitude-modulated noise included:
– the distribution of the magnitude and phase of the modulation coefficient over the carrier
frequency spectrum;
– the width of the frequency band of the modulation process and its dependence on state of
the sea;
– the value of the modulation coefficient.

Keywords: underwater acoustics, signal processing, underwater noise modulation.
The amplitude modulation of underwater noise produced by seagoing vessels is widely used in practice for the classification of ships [1–5]. The purpose of this paper is to describe the results of the experimental study of a model of the amplitude-modulated underwater noise produced by a ship in application to two types of modulation:

1) modulation caused by the rotation of the propeller shaft (shafts) and the screw-propeller (propellers) (in what follows, we will call it shaft–blade modulation; the origin of this term will be explained below);

2) modulation caused by ship roll and pitch in a seaway (we will call it roll–pitch modulation).

Since the model is intended for solving the problem of target classification, we do not consider here other problems connected with these types of amplitude modulation, specifically, the mechanism of their generation. In the most general form, the model of instant spectra of amplitude modulated noise emitted by a ship can be presented as follows:

\[ A_{AMN}(f,t) = [1 + U_{SBM}(f,t) + U_{RPM}(f,t)] \cdot A_N(f) \]  

where \( A_{AMN}(f,t) \) is the instant amplitude spectrum [6] of the amplitude-modulated noise of a ship, \( f \) is the frequency, \( t \) is the running time, \( A_N(f) \) is the instant amplitude spectrum of the nonmodulated quasi-stationary carrier of the noise, \( U_{SBM}(f,t) \) is the normalized modulation process characterizing the shaft–blade modulation of the noise carrier in a narrow frequency band with the central frequency \( f \) and \( U_{RPM}(f,t) \) is the normalized modulation process characterizing the roll–pitch modulation of the noise carrier in a narrow frequency band with the central frequency \( f \).

It is necessary to note that, in the calculation of the spectrum given by Eq. (1), the frequency band of the spectral analysis must be much greater than the frequencies of the shaft–blade and roll–pitch modulations.

Thus, the above-stated purpose of this paper is specified to consist in the refinement of the form and parameters of the functions \( U_{SBM}(f,t) \) and \( U_{RPM}(f,t) \) representing the models of the shaft–blade and roll–pitch modulations, respectively.

We start with the refinement of the model of the shaft–blade modulation. From papers [1–5] concerned with this type of modulation, the following data are known.

1) Shaft–blade modulation is a consequence of the rotation of the propeller shaft and the screw-propeller. The corresponding modulation coefficient strongly depends on the rotation rate. In the case of a small (precavitation) rate of rotation, the modulation coefficient is fairly small, which usually prevents the detection of the shaft–blade modulation in the received signals of noise emission from ships at an insufficiently high signal-to-noise ratio. A sharp growth of the modulation coefficient begins from the rate corresponding to the appearance of cavitation at the screw-propeller. This rate of shaft rotation (as well as the ship speed corresponding to it) is called the critical rotation rate (the critical speed).

2) In the course of the rotation of the propeller shaft and the screw-propeller, the modulation of noise occurs at two discrete frequencies, namely, at the frequency of the shaft rotation (the shaft frequency) and at the frequency equal to the product of the frequency of shaft rotation and the number of blades of the screw-propeller (the blade frequency). This is the reason why this type of modulation is called shaft–blade modulation. The shaft frequencies of seagoing vessels lie within the interval 2–6 Hz, and the blade frequencies, within 6–24 Hz. Due to the nonlinear effects accompanying the radiation, the discrete spectrum of shaft–blade modulation usually contains the harmonics of the shaft and blade frequencies, which form a combined shaft–blade sound series with the fundamental equal to the shaft frequency. The shaft–blade sound series may extend to 100 Hz or more, but in most cases, it does not go beyond 30 Hz. In some cases, the shaft frequency and its harmonics cannot manifest themselves and the spectrum of the shaft–blade modulation contains only the blade sound series.
Taking this into account, the model of the shaft–blade modulation can be represented in the form [2]

\[ U_{SBM}(f, t) = \sum_{i=1}^{N_{SBSS}} m_{SBM}(f) \cdot \cos \left[ 2 \cdot \pi \cdot \left( F_{SBM} \cdot t + \delta F_{SBM}(t) / F_{SBM} \right) + \varphi_{SBM}(f) \right] \]  

(2)

where \( N_{SBSS} \) is the number of harmonics in the shaft–blade sound series, \( F_{SBM} \) is the middle frequency of the \( i \)th harmonic in Hertz, \( \delta F_{SBM}(t) \) is the centered normal process describing the frequency fluctuations of the \( i \)th harmonic, \( m_{SBM}(f) \) is the frequency dependence of the modulation coefficient of the modulated process for the \( i \)th harmonic of the shaft–blade sound series, and \( \varphi_{SBM}(f) \) is the frequency dependence of the initial phase of modulation of the modulated process for the \( i \)th harmonic of the shaft–blade sound series.

3) A device whose flowchart is given in Fig. 1 is used to separate shaft–blade modulation [2, 7]. An amplitude-modulated signal is fed to the input of this device. A band-pass filter provides an opportunity to select the carrier frequency band, where the amplitude envelope is separated. An automatic gain control provides a constant level of amplitude-modulated signal at the filter output. The envelope is separated by using an amplitude detector. As a rule, the latter is represented by a full-wave linear detector realized by ignoring the sign of the output process. This detector provides a certain advantage over other detectors [7]. A low-pass filter limits the frequency range of the envelope by the band containing a discrete spectrum of the shaft–blade modulation. A narrow-band spectrum analyzer calculates the accumulated narrow-band energy spectrum of the envelope of the shaft–blade modulation, in which it manifests itself as a shaft–blade sound series, as has been mentioned above. A computing device connected to the output of the spectrum analyzer serves to separate this shaft–blade sound series.

![Flowchart of the device for the separation of the amplitude modulation of broadband noise produced by marine objects.](image)

**Fig. 1.** Flow chart of the device for the separation of the amplitude modulation of broadband noise produced by marine objects (BPF is a band-pass filter, AGC is a device of automatic gain control, AD is an amplitude detector, LPF is a low-pass filter, SA is a narrowband spectrum analyzer, and CD is a computing device).

The noise immunity of the device pictured in Fig. 1 is described by the formula [8]

\[ Q_{out,i} = \sqrt{N_{SA} \cdot \frac{\Delta f_{BF}}{\max(\Delta F_{SA}, \Delta F_{SBM})} \cdot m_{SBM}^2 \cdot \frac{q_{in}^2}{(q_{in} + i)^2}} \]  

(3)

where \( Q_{out,i} \) is the output signal-to-noise ratio equal to the ratio of the excess of the \( i \)th harmonic of the shaft–blade sound series observed in the energy spectrum of the envelope over the average level of the background in its vicinity to the value of the rms fluctuation of the background in the same vicinity; \( \Delta F_{SBM} \) is the bandwidth of the frequencies of the modulating process corresponding to the \( i \)th harmonic of the shaft–blade sound series in hertz; \( m_{SBM} \) is the modulation coefficient corresponding to the \( i \)th harmonic of the shaft–blade series, which is equal to the ratio of the modulating process amplitude to the carrier amplitude; \( \Delta f_{BF} \) is the filter band in hertz; \( q_{in} \) is the input ratio of the power of the amplitude-modulated signal to the power of noise at the filter output; \( \Delta F_{SA} \) is the band of the spectral analysis in hertz; and \( N_{SA} \) is the number of the acts of energy spectrum storage.

To refine the model of shaft–blade modulation, a processing of the noise records obtained from the output of an omnidirectional hydrophone with a transmission band from 10 Hz to 10 kHz was carried out. The hydrophone was cast from a ship drifting near navigation...
routes (in various regions of the ocean) to a depth of 50–200 m, which provided the best conditions for the reception of underwater noise from a ship passing by.

The noise was recorded by a measuring tape recorder with an operational frequency band of 5 Hz to 10 kHz. Each record began when a ship was visually detected, on the condition that its noise level exceeded the level of ambient sea noise by no less than 10 dB. The length of records was from 5 to 30 min, depending on the observation conditions. Each record was complemented by characteristic information containing the type and name of the ship, its speed, the time dependence of the record of its relative bearing (aspect angle), the records of marine noise before and after the ship’s passage, and the degree of sea roughness.

In total, 189 noise records were obtained, including 67 records from tankers, 39 records from dry cargo carriers, 31 records from trawlers, 23 records from passenger vessels, and 29 records of noise from ships of other types.

The distribution of the width of the frequency band of the modulating process was determined first of all. The experimental technique proceeded from the fact [see Eq. (3)] that if

\[ \Delta F_{SA} > \Delta F_{SBM} \]

is satisfied (i.e., the band of analysis exceeds the width of the frequency band of the modulating process), a decrease in the band of analysis \( \Delta F_{SA} \) will be accompanied by a proportional increase in the output signal-to-noise ratio \( Q_{out} \). This growth stops only when Eq. (4) becomes invalid. Taking this fact into account, the technique used for the record processing was chosen to be as follows. Each record of underwater noise, being multiply reproduced by the tape recorder, was fed to the input of the processing device whose flowchart is given in Fig. 1. The processing of each record was performed twice. At first, the one-third-octave band of the filter was determined within the range 50 Hz–10 kHz, at the output of which the output value of the signal-to-noise ratio \( Q_{out} \) was maximal. All other parameters of the device were fixed: the cutoff frequency of the lowpass filter was 25 Hz, the band of analysis was \( \Delta F_{SA} = 0.05 \) Hz, and \( N_{SA} = 4 \). Then, the selected one-third-octave band corresponding to the maximum of the signal-to-noise ratio \( Q_{out} \) was set as the filter band, and only the band \( \Delta F_{SA} \) was reduced from 0.1 Hz down to the value at which the growth of the output signal-to-noise ratio \( Q_{out} \) terminated for all harmonics of the shaft–blade sound series. In this process, for each harmonic of the shaft–blade sound series, the band \( \Delta F_{SA} \) was fixed, at which the growth of the output signal-to-noise ratio for this harmonic terminated. This band was taken as the width of the frequency band of the modulating process \( \Delta F_{SBM} \) for this harmonic. The results obtained after processing all the records were analyzed, and this analysis led to the following conclusions. The bandwidth of the modulating process in the case of shaft–blade modulation depends most strongly on the number of harmonic and on the degree of sea roughness. The dependence on the harmonic number is a directly proportional one, and, therefore, the relative (with respect to the middle frequency of the harmonic) band of the modulating process essentially depends on only the degree of sea roughness according to the empirical formulas

\[
M \left\{ \frac{\Delta F_{SBM}}{F_{SBM}} \right\} = \left( 3 + 0.25 \cdot W^3 \right) \cdot 10^{-3}
\]

\[
\sigma \left\{ \frac{\Delta F_{SBM}}{F_{SBM}} \right\} = \max[2.5; 2.5 + 5 \cdot (W - 2)] \cdot 10^{-3}
\]

where \( M\{X\} \) and \( \sigma\{X\} \) are the sample mathematical expectation and the rms deviation of the random quantity \( X \), and \( W \) is the degree of sea roughness (number on the Beaufort scale).
From Eqs. (5), it follows in particular that, in the case of small sea roughness (up to Beaufort 3 inclusive), the relative bandwidth of the modulating process of the shaft–blade modulation is no greater than 1%.

The distribution of the shaft–blade modulation coefficient and the dependence of the modulation coefficient on the frequency of the modulated carrier were studied using the same device (Fig. 1). The technique used for the record processing was as follows. Of all the noise records available, 149 were chosen to correspond to the sea roughness up to Beaufort 3 inclusive. All the device parameters, except for the filter band, were fixed (the cutoff frequency of the low-pass filter was 25 Hz, $\Delta F_{SA} = 0.05$ Hz, and $N_{SA} = 8$). Each selected noise record was multiply reproduced by the tape recorder and fed to the device input, each time with a new filter band value. Simultaneously, both the central frequency and the filter bandwidth were changed: the central frequency was tuned from 50 Hz to 10 kHz and the relative filter width was 22% for the central frequencies within the range 50–500 Hz, 10% within the range 0.5–1 kHz, 5% within the range 1–2 kHz, 3% within the range 2–4 kHz, and 1.5% within the range 4–10 kHz. The input signal-to-noise ratio $q_{in}$ was calculated for each of these bands (as the ratio of the power of the amplitude-modulated signal to the noise power at the filter output). Then, the harmonics of the shaft–blade sound series were separated from the envelope spectrum, and the output signal-to-noise ratio was calculated for each of them (as the ratio of the excess of the $i$th harmonic of the shaft–blade sound series over the average background level in its vicinity to the value of the rms fluctuation of the background in the same vicinity). After that, the modulation coefficient was calculated for each harmonic of the shaft–blade sound series according to the formula

$$m_{SBM_i} = \frac{Q_{out_i}}{\sqrt{N_{SA} \cdot \Delta f_{PF}} \cdot \frac{q_{in}}{\Delta F_{SA} (q_{in} + 1)}} \quad (6)$$

which was obtained from Eq. (3) with condition (4) (according to the results of processing, this condition is satisfied in the case of small sea roughness with the selected band of analysis of 0.05 Hz).

The analysis of the processing results obtained for all records led to the following conclusions.

1) In the case of ship movement with over-critical speed, all studied carrier frequencies (50 Hz–10 kHz) are subjected to shaft–blade modulation.

2) The dependence of the modulation coefficient on the carrier frequency for each noise record has a complex form characterized as a rule by the presence of so called cavitation peaks. The form of this dependence is repeated for all harmonics of the shaft–blade series. It was impossible to determine the connection of the peak positions with the type, speed, and aspect angle of ships. The result of averaging the dependences of the modulation coefficient on the carrier frequency over all processed noise records is an almost straight line parallel to the frequency axis. In view of this result, when synthesizing an algorithm for target classification, it is expedient to proceed from a uniform distribution of the coefficient of shaft–blade modulation over the modulated carrier frequencies. The shaft–blade modulation coefficient that corresponds to the maximal harmonic of the shaft–blade sound series with respect to the output signal-to-noise ratio lies within the interval 0–0.15 with a probability of 0.95, and single peaks reach a value of 0.4. The distribution of this shaft–blade modulation coefficient obeys a limited ($m_{SBM} \geq 0$) normal law with a mathematical expectation of 0.06 and an rms deviation of 0.035.

To study the distribution of the initial phase of the shaft–blade modulation process over the modulated carrier frequencies of noise, the device shown in Fig. 1 was transformed into the device shown in Fig. 2. The device became a two-channel one, and the spectrum ana-
lyzer began to operate in the mode of calculation of the amplitude and phase cross-spectra. The values of the phase cross-spectrum at the frequency of each harmonic of the shaft–blade sound series were determined by the computing device, together with the separation of the harmonics of the shaft–blade sound series in the amplitude spectrum. The technique used for the record processing was as follows. The band-pass filters BPF-1 and BPF-2 were tuned to the nonoverlapping frequency bands from the band set realized in the preceding experiment. Other parameters of the device were also taken from the preceding experiment. The noise record played back by the tape recorder was fed simultaneously to both inputs of the processor. The phase differences of the modulating processes corresponding to each harmonic of the shaft–blade sound series that was separated from the cross-spectrum of the amplitude envelopes detected in the nonoverlapping bands of the carrier were determined.

Fig. 2. Flow chart of the device for the determination of the distribution of the initial phase of a modulating process over the carrier frequency spectrum. (Notation is the same as in Fig. 1.)

An analysis of the results of processing all the records by this technique when different nonoverlapping bands of the carrier are chosen led to an expected conclusion: the phase difference of like processes (i.e., corresponding to the same harmonics of the shaft–blade sound series) modulating different bands of the carrier is equal to zero in all cases correct to the measurement error of the phase spectrum; i.e., the modulation of all carrier frequencies is performed in phase. This fact could be predicted proceeding from the unified mechanism of formation of the shaft–blade modulation. Our experiments confirm this hypothesis. Moreover, the conclusion that the modulation of all carrier frequencies is in phase is very important, since it provides an opportunity to substantiate such an important parameter of the device separating the shaft–blade modulation as the preselector band.

The initial phase of the modulating process \( \phi_{SBM}(f) \) in Eq. (2) does not depend on frequency due to the in-phase character of the modulation of all carrier frequencies of the ship noise, and, furthermore, it can be assumed to be equal to zero for all harmonics of the shaft–blade sound series without any loss for the problem.

Now, let us proceed to the refinement of the model of noise modulation by the roll and pitch of the ship.

The following data are known from earlier papers.
1) The spectrum of ship noise can be modulated in amplitude by ship roll or pitch (and in rare cases, by both of them together).
2) The frequency of ship roll lies within the interval 0.03–0.2 Hz and the frequency of ship pitch, within the interval 0.12–0.4 Hz, depending on the type and tonnage of a ship.
3) The coefficient of modulation by roll and pitch increases monotonically with the growth of the degree of sea roughness.

Taking these facts into account, the noise modulation by roll and pitch can be represented in a form analogous to the model of the shaft–blade modulation [Eq. (2)]:

\[
U_{RPM}(f,t) = m_{RPM}(f) \cdot \cos \left[ 2 \cdot \pi \cdot \left( F_{RPM} \cdot t + \delta F_{RPM}(t) / F_{RPM} \right) + \phi_{RPM}(f) \right]
\]

(7)
where $F_{RPM}$ is the average frequency of the ship roll or pitch in hertz, $\delta F_{RPM}(t)$ is the centered normal process describing the frequency fluctuations of roll or pitch, $m_{RPM}(f)$ is the frequency dependence of the coefficient of modulation by roll or pitch, and $\varphi_{RPM}(f)$ is the frequency dependence of the initial phase of modulation by roll or pitch.

To separate this type of modulation, the same device was used as for the separation of shaft–blade modulation (Fig. 1). The processing of magnetic records of ship noise for studying the parameters of modulation by roll and pitch was conducted using the same technique (with the corresponding changes in the frequency range).

The analysis of the processing results led to the following conclusions.

1) The frequency bandwidth of the modulating process in the case of modulation by roll and pitch lies within the interval 0.01–0.12 Hz. It was impossible to determine the dependence of the width of the modulating process on the sea roughness and the ship type.

2) All frequencies of the studied frequency range 50 Hz–10 kHz are subjected to modulation by roll and pitch. The coefficient of modulation by roll and pitch strongly depends on the sea roughness and the carrier frequency.

3) In the case of sea roughness within Beaufort 3–6, the mathematical expectation of the frequency dependences of the coefficient of modulation by roll and pitch and its rms deviation is well approximated by the dependences

$$M \left\{ m_{RPM} (f) \right\} = \left[ 9.5 + 15.5 \cdot \lg \left( f_{\text{MHz}} \right) \right] \cdot 10^{-2}$$

$$M \left\{ \sigma_{RPM} (f) \right\} = 0.05 \ldots 0.07$$

4) In the carrier frequency band 0.5–8 kHz, the mathematical expectation of the coefficient of modulation by roll and pitch and its rms deviation depending on sea roughness have the form

$$M \left\{ m_{RPM} (W) \right\} = 0.043 \cdot W$$

$$M \left\{ \sigma_{RPM} (W) \right\} = 0.04 \ldots 0.06$$

5) The modulation of all carrier frequencies by roll and pitch occurs in phase, which allows one to set the term $\varphi_{RPM}(f)$ in Eq. (7) equal to zero.

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MULTI-ASPECT FEATURE EXTRACTION AND FEATURE INTEGRATION WITH APPLICATION TO UNDERWATER ORDNANCE RECOGNITION

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Abstract: This paper examines the acoustic signatures of three types of underwater ordnances (MK6 Hedgehog, M47A2 100lbs Chemical Bomb, and 75mm MKII Projectile) using their 360-degree free-space scattering echoes. A novel feature extraction approach has been developed to obtain the multi-aspect acoustic signatures. The acoustic signals are examined on two-dimensional Time-Scale (TS) space using a Continuous Wavelet Transform (CWT). A modified Gustafson-Kessel (GK) clustering algorithm is applied to create a clustering distribution which represents the scattering features in the TS space. For each target, the scattering echoes from the 360-degree-range aspects are divided into several classes according to their correlations with the incident pulse. A feature template is generated for each class by fusing the corresponding clustering distributions. A target recognition scheme is developed by comparing the clustering distribution of test signals with the available feature templates. Efficiency of the multi-aspect feature templates has been evaluated with the backscattered data collected in a tank experiment. A digitized dolphin-click with a center frequency of 120 kHz is used as the incident pulse. The target is placed on a rotor which rotates contraclockwise at a constant speed, while the backscattered echoes are collected from the range of 0-360 degree. The resulting recognition rate increases from 65% to 91% when the number of observation angles used for the recognition varies from 1 to 12.

Keywords: Underwater ordnance recognition, wavelet analysis, fuzzy clustering analysis, multi-aspect feature extraction, dolphin SONAR.
1. INTRODUCTION

In an active acoustic scattering experiment, a small variation of the incident and/or observation angle may yield significant difference in the returned echoes even for the same target. A multi-aspect feature extraction scheme provides a feature template which characterizes a target’s scatterings of 0-360 degree view. Traditionally, it is conducted by searching for the common features, known as the aspect-independent features, in the scattering echoes received at different angles. In an early study [1], Principal Component Analysis (PCA) is applied on sampled frequency responses of different aspect angles, to extract the features that vary the least as the aspect changes. A more sophisticated method is introduced in [2], which examines the cross-correlations between the incident pulse and the scattering responses of different aspects, then constructs a feature template based on the coefficients of wavelet package decomposition and a linear predictive coding model. Note that the selection of aspect-independent features merely preserves the aspect-invariant characteristics such as the target’s size and material content. Therefore these methods are less applicable when the targets to be discriminated are similar with respect to their sizes and material structure, which is a common situation for underwater ordnances.

We introduce a new multi-aspect feature extraction approach where the feature templates are created by fusing the scattering responses received at different angles. This method is applied to examine the acoustic signatures of three underwater ordnance models, with their 360-degree scattering responses collected in a confined tank experiment. Section 2 details the scattering measurement, including the tank experiment setup and the physical properties of modelled ordnances. Section 3 introduces the procedures of multi-aspect feature extraction, featuring on Continuous Wavelet Transforms (CWT) and Gustafson-Kessel (GK) fuzzy clustering analysis. The multi-aspect feature templates are used to discriminate the modelled ordnances, with the results and discussions given in Section 4 and 5, respectively.

2. TANK EXPERIMENT

The scattering experiment was conducted in a controlled environment, namely, a water tank which is approximately 2.4 m of diameter and 1.8 m of depth. The experimental setup is illustrated in Fig. 1 (a). A disk transducer was placed in the water to transmit incident pulses towards the center of a target, and to collect the backscattered echoes. The target was attached to a rotor and it was suspended horizontally into the tank. As viewed from the top, it rotated counter-clockwise about the vertical axis. Let \( \theta \) denote the aspect of observation, which is defined as the angle between the major axis of the target and the vertical alignment of the transducer. Each target was insonified at 550-650 aspect angles, with \( \theta \) ranging from 0 to 360 degrees.

As shown in Fig. 1 (b), the targets considered in this experiment are models of the following military ordnances: (1) MK6 Hedgehog, scaled 5:1 from the actual size; (2) M47A2 100lbs Chemical Bomb, scaled 5:1 from the actual size; (3) 75mm MKII Projectile, machined according to the actual size. The targets are made of a combination of A36 and 1018 mild steel, with their cavities filled by air, water, or sand in the experiments. For convenience, we label the targets according to their type and filling material, e.g. the MK6 Hedgehog target filled with the three materials are referred to as 1A, 1W, and 1S, respectively.
A digitized dolphin-emitted sound was used as the incident pulse. The center frequency of the transmitted dolphin sound was 120 kHz. The incident pulse and its frequency spectrum can be found in [3]. The sampling frequency was set to 1 MHz at the receiver, and a signal consisting of 2048 samples was collected at each aspect angle. A window function was applied to eliminate the scattering echoes from the back wall of the tank.

Fig. 2 displays the backscattered signals measured at 45° and 90°, respectively. Note that the signals received at 45° (shown in the left column) have much different scattering patterns
comparing to the signals received at 90° (shown in the right column). When θ=45°, the scattering responses exhibit “flat” temporal features; whereas the features are compactly supported in a short time-duration for θ=90°. Hence the traditional aspect-independent feature extraction would lead to a loss of features which only appear at certain angles. Another observation is that the scattering features from different targets may be extremely similar at some angles, but the discrimination can be made using the scattering responses received at a different angle. For example, the backscattered signals from targets 2W and 3W exhibit similar temporal features at 90°, but their differences are distinguishable at 45°.

3. MULTI-ASPECT FEATURE EXTRACTION AND FEATURE FUSION

The backscattered signals are examined in the two-dimensional Time-Scale (TS) space using a Continuous Wavelet Transform (CWT). Let f(t) denote an acoustic backscattered signal, its CWT is given by [4]:

$$Wf(u,s) = \int_{-\infty}^{\infty} f(t) \frac{1}{\sqrt{s}} \psi^*(\frac{t-u}{s})dt,$$  

(1)

where \(\psi(t)\) denotes the mother wavelet, \(u\) and \(s\) are the translation and dilation parameters, respectively.

To prepare the CWT results for fuzzy clustering, we calculate the absolute values of the computed CWT and normalize the results by setting the maximum amplitude equal to one. The values below a pre-selected threshold \(\gamma\) (with \(0<\gamma<1\)) are set to zero, so that the influence of signal clutter is suppressed. Fuzzy clustering analysis is an unsupervised learning process which partitions an input data set into a number of clusters according to the data distribution. When it is applied on the normalized and thresholded CWT coefficients, it creates a clustering distribution where the TS features are represented by a number of clusters. In our previous work [3], [5], a modified Gustafson-Kessel (GK) fuzzy clustering algorithm was introduced to generate elliptical-shaped clusters with auto-adjustable sizes. The clustering algorithm runs in iterations with the objective of minimizing the following cost function:

$$J_{\xi} = \sum_{i=1}^{N} \sum_{j=1}^{c} u_{ij}^m \cdot \zeta_i \cdot d^2(x_i, \upsilon_j),$$  

(2)

where \(c\) denotes the number of clusters, \(x_i\) and \(\zeta_i\) are the coordinate and amplitude of the \(i\)-th non-zero pre-processed wavelet coefficient, \(u_{ij}\) denotes the degree of association between \(x_i\) and the \(j\)-th cluster center \(\upsilon_j\), and \(d(x_i, \upsilon_j)\) is the elliptical distance between \(x_i\) and \(\upsilon_j\) [5].

In order to maintain the distinctive features at different observation angles, we introduce a pre-grouping step prior to the feature fusion. We calculate the cross-correlations between the scattered responses and the incident pulse, and divide the 360-degree scattering responses into a number of classes according to the resulting cross-correlation.
Fig. 3: Cross-correlation versus \( \theta \), for the scattering responses from target 1A.

Fig. 3 displays the normalized cross-correlations between the incident dolphin-click and the scattering responses from target 1A. Signals received around 90°, 180° and 270° closely resemble the incident pulse; while around 45° and 315°, the scattering responses exhibit flat temporal features and yield small cross-correlations. Let \( C_{fg} \) denote the cross-correlation between a scattering response \( f(t) \) and the incident pulse \( g(t) \), the backscattered echoes are divided into three classes according to the values of \( C_{fg} \): (I) \( 0 < C_{fg} \leq 0.10 \); (II) \( 0.10 < C_{fg} \leq 0.17 \); and (III) \( 0.17 < C_{fg} \leq 1 \).

A multi-aspect feature template is then generated for each class, by fusing the clustering distributions of the scattering responses of the same class. The cluster centres of different clustering distributions are combined, while the overlaps are eliminated. The feature templates created using the scattering responses from target 1A are displayed in Fig. 4, where the graphs (a)-(c) each represent the feature fusion result of class I-III, respectively. The mother wavelet \( \psi(t) \) used in CWT is the Morlet wavelet function \([4]\), with the maximum level of decomposition set to 16. The threshold applied on the computed CWT coefficients is \( \gamma = 0.05 \). The number of clusters in GK clustering is set to \( c = 32 \), which is selected using the partitioning measurement introduced in \([3]\).

Fig. 4: Multi-aspect feature templates of target 1A.

4. TAGRET RECOGNITION RESULTS

To prove the efficiency of the multi-aspect feature extraction method, the multi-aspect feature templates are used to discriminate targets 1A, 1W, 1S, 2W, and 3W. Scattering responses received at 0° to 357° with 3° increments are used to generate the feature templates, and the rest of the signals are used for testing. This yields 120 training samples and 240 testing samples for each target. The recognition is conducted by comparing the clustering distributions of test signals with the available feature templates, and classification result is given by the target which yields the closest match \([3], [5]\). The resulting correct recognition
percentage is 65%. To improve the recognition rate, we introduced a decision-level fusion which utilizes the classification results obtained from multiple scattering responses received at a sequence of concessive observation angles. Fig. 5 gives the recognition rate between the five targets, when the number of observation angles varies from 1 to 12. The recognition rate reaches above 85% when more than five observation angles are used, and it is above 90% when the number is increased to ten.

![Recognition rate graph](image)

**Fig. 5**: Recognition results with different number of observation angles.

5. CONCLUSION

A novel multi-aspect feature extraction and feature fusion approach is developed based on CWT and fuzzy GK clustering analysis. Its efficiency is tested on the free-space acoustic scattering responses from three underwater ordnances. A target recognition test is conducted using the multi-aspect feature templates. With the number of observation angles ranging from 1 to 12, the recognition rate increases from 65% to 91%.

6. ACKNOWLEDGEMENTS

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THEORETICAL OPTICAL FLOW FOR TARGET POSITION
PREDICTION ON FLS IMAGES

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Abstract: This study is related to the obstacle avoidance issue for an Autonomous
Underwater Vehicle (AUV). In addition to its original mission, the vehicle must ensure its
own survival and therefore understand the environment in safety. The use of a Forward
Looking Sonar (FLS) on AUV is one of the most efficient solutions to detect unexpected and
potentially dangerous changes of the environment, like the presence of obstacles or seabed
slope. Like this, a FLS can prevent the vehicle from obstacles or terrain that may endanger
the underwater vehicle. Besides navigation is performed knowing data from a Doppler
Velocity Log (DVL) and a Motion Reference Unit (MRU). From these last data a process
model has been derived in order to predict the motion of a ground target detected in a sonar
image. The state vector is composed of the coordinates of the target in the image. Motion
prediction leads to a theoretical optical flow that we can compare to an observed optical flow
based on images correlation.

This study is of high interest for the French organism GESMA which is involved in the
development of decisional autonomy for AUV for several years. Results will be given on real
data recorded in April 2006 during sea trials organized by GESMA and should be extended
on data recorded during future campaigns involving the Rapid Environment Assessment
(REA) AUV “Daurade”.

Keywords: Optical flow - Obstacle Detection and Avoidance – AUV – Forward Looking
Sonar
1. INTRODUCTION

The development of AUVs is a real challenge for researchers that have to supply the vehicle with a total autonomy in terms of energy, data processing, navigation in order to enable it to bring off its mission. This leads to related issues such as survival in a hostile context. A Forward Looking Sonar (FLS) can prevent the vehicle from obstacles or terrain that may endanger the underwater vehicle [1]. Indeed by combining imaging data with navigation data, still ground targets can be detected and tracked. The first step of tracking consists in deriving a process model in order to predict the motion of ground targets on the sonar screen. This model will be used in the prediction step of a Kalman filter that will be implemented later to track a still target through the following frames. A theoretical optical flow based on the process model can be visualised on the sonar images sequence and compared to an observed optical flow based on images correlation.

This study is of high interest for GESMA involved in the development of experimental AUVs such as the Redermor for several years [2] [3] and more recently in the Rapid Environment Assessment (REA) AUV “Daurade” [4].

This paper has been divided into three main parts. Part number 2 gives a quick description of data at our disposal in this study. The development of part 3 ends with the process model able to predict the motion of a target detected on the sonar screen by taking into account navigation data. The fourth part gives some results on real data.

2. DATA DESCRIPTION

2.1. Sea trials

The Redermor is the experimental platform deployed from the French Navy ship BEGM Thetis. It is a heavy and large AUV (3.8 tons x 6 m).

In order to test the capability of the Redermor vehicle to react when obstacles are encountered on its way, GESMA organized an experimental trial in April 2006, named DEVITOBS’06 “DETecion et EVITement d’OBStacles”. The aim of this campaign was to record sonar data in several modes with various obstacles. In that way, it has been possible to test, qualify and upgrade the sensors suite, to initiate an obstacle database, to start algorithm development on those obstacles. The goal was to prepare avoidance tactics and strategies to be given to robot mission supervisor.

The experiment has been conducted in the Douarnenez Bay, near Brest. Several objects have been laid: a moored mine like object, a net and plastic chains. Other objects have been investigated like the shipwreck “Meuse” in the same bay and schools of fish. Up to now analysis has been mainly focussed on the Reson 8101 data.

2.2. Avoidance sonar data

In its last release, avoidance means of Redermor consist of a network of 10 Tritech echosounders and a Reson Seabat 8101 Forward Looking Sonar. The 240 kHz Reson Seabat 8101 FLS is derived from a multibeam echosounder and can operate in bathymetric mode or a sector scan mode. The system integrated in the Redermor can play a beamformed image over a 15° (vertical) x 60° (horizontal) sector with a 1.5° azimuth resolution and a 5 cm range resolution. The sonar has been oriented 15° from the horizontal plane.
In this article sonar data used only come from the Reson Seabat 8101 FLS. The Reson Seabat 8101 FLS provides data that are processed and plotted on a sector Plan Position Indicator (PPI) display. This image is characterized by low contrast. The receive beam of 1.5° (H) by 15° (V) does not provide precise details and precise localization of the obstacle. For a range from 20 to 100m, the resolution cell increases from about 50cm to 2.5m in length according to the sector formed.

2.3. Navigation data

Navigation is performed knowing data from a Doppler Velocity Log (DVL) and a Motion Reference Unit (MRU). The DVL gives the vehicle speed in relation to the seafloor. The MRU gives the vehicle orientation and its acceleration in relation to the earth (or absolute) reference frame (X : geographical North, Y : East, Z : gravity direction).

In the AUV environment, one can consider the absolute reference frame \( R_a(O_a,i_a,j_a,k_a) \) where the \( i_a \) vector stands for geographical North, \( j_a \) stands for East and \( k_a \) is oriented towards the centre of the Earth, \( O_a \) is a point at the ocean surface. Besides, let us consider the AUV (or mobile) reference frame \( R_r(O_r,i_r,j_r,k_r) \) where \( O_r \) is merged with the inertia centre of the vehicle, \( i_r \) is oriented towards the front of the vehicle, \( k_r \) towards the seafloor and \( j_r \) direction is deduced from \( i_r \) and \( k_r \) to provide an orthonormal reference frame (see Fig. 1).

To convert the coordinates from the mobile reference frame to the absolute reference frame we use a rotation matrix \( R_{Euler}(\phi, \theta, \psi) \) where \( (\phi, \theta, \psi) \) are the Euler angles. The angles provided by the MRU are sufficient to describe any vehicle motion.

In the absolute reference frame, the following data are available :

- \( p_a = (p_a^x, p_a^y, p_a^z) \) stands for the coordinates of the AUV (we supposed its location merged with all the other sensors)
- \( m_a = (m_a^x, m_a^y, m_a^z) \) stands for the coordinates of an object laying on the seafloor
- \( v_a = (v_a^x, v_a^y, v_a^z) \) stands for the speed of the AUV

In the mobile reference frame, the following data are available :

- \( v_r = (v_r^x, v_r^y, v_r^z) \) stands for the speed of the AUV (measured by the DVL)
- \( m_r = (m_r^x, m_r^y, m_r^z) \) stands for the coordinates of an object laying on the seafloor

3. PROCESS MODEL [5][6]

3.1. Introduction

The researched process model must describe the motion of a target whose coordinates are \((x_e, y_e)\) on the sonar screen given the navigation data. This model is obtained in two steps :

- The first one consists in modelling the coordinate conversion from the mobile reference frame to the screen image reference frame. In other words, it consists in finding the relation between the sonar coordinates \((d, \delta)\) of the target in the mobile reference frame and the screen coordinates \((x_e, y_e)\). Let us call \( g \) the corresponding function.
- The second one consists in modelling the motion of the target on the screen by taking into account the navigation data. Let us call \( f \) the corresponding function.
We have $\dot{p}_e = (\dot{x}_e, \dot{y}_e) = f(x_e, y_e, u) = f(p_e, u)$, where $u$ stands for a vector of navigation data.

Fig. 1 gives the operational configuration with the position of variables used in the following. Seafloor is supposed to be flat.

**3.2. First step**

In order to visualise a sector image, we have to convert polar data to Cartesian data given the following equations:

$$\begin{align*}
\begin{cases}
x = r \times \sin \theta & \text{along the columns} \\
y = r \times \cos \theta & \text{along the rows}
\end{cases}
\end{align*}$$

with

$$\begin{align*}
\begin{cases}
r = d / \Delta d \\
\theta = \delta / \Delta \theta
\end{cases}
\end{align*}$$

where $(r, \theta)$ stand for polar coordinates

$(x, y)$ stand for Cartesian coordinates

$(d, \delta)$ stand for sonar coordinates (distance, azimuth)

$(\Delta d, \Delta \theta)$ stand for sampling rates in range and in azimuth

Finally, to follow the image representation norm of MatLab (image whose size is M rows by N columns), a reference frame rotation is operated to convert Cartesian sonar coordinates $(x_e, y_e)$ into Cartesian screen coordinates $(x_e', y_e')$ such as:

$$\begin{align*}
\begin{cases}
\delta = \Delta \theta \times \arctan \left( \frac{-x_e + \frac{N}{2}}{-y_e + \frac{M}{2}} \right) \\
d = \Delta d \times \arctan \left( \frac{-x_e + \frac{N}{2}}{-y_e + \frac{M}{2}} \right)
\end{cases}
\end{align*}$$

(2)
Or,
\[
\begin{align*}
    x_e &= \frac{N}{2} - \frac{d}{ddist} \sin \left( \frac{\delta}{d\theta} \right) \\
    y_e &= M - \frac{d}{ddist} \cos \left( \frac{\delta}{d\theta} \right)
\end{align*}
\]
i.e. \((x_e, y_e) = g(d, \delta)\) (3)

This first step allows conversion from the mobile reference frame to the image reference frame.

### 3.3. Second step

The model of the vehicle is given by:
\[
\dot{p}_a = v_a = R_{\text{euler}}(\varphi, \theta, \psi)v_r
\]
(4)

We have as well
\[
p_a - m_a = -R_{\text{euler}}(\varphi, \theta, \psi) \cdot m_r
\]
(5)

By derivating the last equation, we get
\[
v_a = -\hat{R}_{\text{euler}} \cdot \begin{pmatrix} m_r^x \\ m_r^y \\ m_r^z \end{pmatrix} - \hat{R}_{\text{euler}} \cdot \begin{pmatrix} \dot{m}_r^x \\ \dot{m}_r^y \\ \dot{m}_r^z \end{pmatrix}
\]
(6)
i.e.
\[
\begin{pmatrix} \dot{m}_r^x \\ \dot{m}_r^y \\ \dot{m}_r^z \end{pmatrix} = -R^T_{\text{euler}} \hat{R}_{\text{euler}} \cdot \begin{pmatrix} m_r^x \\ m_r^y \\ m_r^z \end{pmatrix} - v_r = f_h(m_r^x, m_r^y, m_r^z, v_x, v_y, v_z, \varphi, \theta, \psi, \varphi, \dot{\varphi}, \dot{\theta}, \dot{\psi})
\]
(7)

Moreover in the mobile reference frame the target is located by means of the following equations (see Fig. 1):
\[
\begin{align*}
    (m_r^x)^2 + (m_r^y)^2 + (m_r^z)^2 &= d^2 \\
    m_r^x &= \tan \delta \cdot m_r^x = \tan \delta \cdot d \\
    -\sin \theta \cdot m_r^x + \cos \theta \cdot \sin \varphi \cdot m_r^y + \cos \theta \cdot \cos \varphi \cdot m_r^z &= h
\end{align*}
\]
(8)

We can show that:
\[
a_1 \left( m_r^x \right)^2 + b_1 m_r^x + c_1 = 0
\]
(9)

With:
\[
\begin{align*}
    a_1 &= 1 + \left( \frac{\sin \theta}{\cos \theta \cdot \cos \varphi} \right)^2 \\
    b_1 &= 2 \frac{\sin \theta}{\cos \theta \cdot \cos \varphi} \left( h - \cos \theta \cdot \sin \varphi \cdot \tan \delta \cdot d \right) \\
    c_1 &= \left( \tan^2 \delta - 1 \right) \cdot d^2 + \left( \frac{h - \cos \theta \cdot \sin \varphi \cdot \tan \delta \cdot d}{\cos \theta \cdot \cos \varphi} \right)^2
\end{align*}
\]
(10)

and deduce that
By deriving (3), we obtain:

\[
\begin{pmatrix}
\dot{x}_e \\
\dot{y}_e
\end{pmatrix} =
\begin{pmatrix}
\frac{1}{ddist} \sin \left( \frac{\delta}{d\theta} \right) & \frac{d}{ddist} \frac{\cos \left( \frac{\delta}{d\theta} \right)}{d\theta} \\
\frac{1}{ddist} \cos \left( \frac{\delta}{d\theta} \right) & \frac{d}{ddist} \frac{\sin \left( \frac{\delta}{d\theta} \right)}{d\theta}
\end{pmatrix}
\begin{pmatrix}
d \\
\delta
\end{pmatrix}
\]  

i.e. \((\dot{x}_e, \dot{y}_e) = f_d(d, \delta, \dot{d}, \dot{\delta})\) (13)

where \(\dot{d}, \dot{\delta}\) can be derived from (8):

\[
\begin{pmatrix}
\dot{d} \\
\dot{\delta}
\end{pmatrix} =
\begin{pmatrix}
\dot{m}_r^x m_r^x + \dot{m}_r^y m_r^y + \dot{m}_r^z m_r^z \\
\dot{m}_r^y - \frac{\tan \delta}{d} \left( \dot{m}_r^x m_r^x + \dot{m}_r^y m_r^y + \dot{m}_r^z m_r^z \right)
\end{pmatrix}
\]  

i.e. \((\dot{d}, \dot{\delta}) = f_r(\dot{m}_r^x, \dot{m}_r^y, \dot{m}_r^z, d, \delta)\) (14)

Given (13), (3), (14), (7) and (11), we can now give the expression of the process model on which our theoretical optical flow is based:

\[
(\dot{x}_e, \dot{y}_e) = f(\hat{x}_e, \hat{y}_e, v_r, \phi, \psi, \dot{\phi}, \dot{\psi})
\]

where \(f = f_d \circ f_r \circ f_o\)

4. RESULTS

4.1. Detection step (Initialization step of a future Kalman filtering)

A degrading physical effect that appears on sonar images is the speckle noise. This noise is multiplicative if we consider the image as the visualisation of the amplitude modulus of the reflected wave which follows a Rayleigh law [7]. Under this hypothesis, we have derived a simple adjustment test that only consists in verifying the relation of proportionality that exists between the mean and the standard deviation of pixels [8]. In practice, we divide the sonar image into smaller images and test for each of them the value of the ratio between the standard deviation value and mean value of pixels levels. If this ration is too far from the expected value (about 0.52) we consider that a target is in sight. A detected point is the centre of inertia of a set of connected small images whose pixels do not follow a Rayleigh distribution.

4.2. Optical flows

We give in this section a comparison between theoretical flow and observed flow. On the one hand, theoretical flow that we will use later in the prediction step of a Kalman filtering is...
based on the previous developed process model. On the other hand, the observed flow is based on the correlation of small images that surround the detected object echo on two successive pings. An example of correlation is given Fig. 2 on the sequence of FLS images where we can see a moored mine in sight.

In order to quantitatively compare the two optical flows, we plot the difference between the two related estimated motions versus each ping. Fig. 3 shows on the left a sector image extracted from the sequence with a moored mine in sight (echoes are surrounded in yellow). The two flows (green: observed, red: theoretical) are plotted (with a larger displacement to be visible). On the right we can see the small difference between the estimated motions and that, in spite of the fact that a moored mine is considered. Actually here the tether is too short in relation to the AUV altitude to make the algorithm fail. Another example is given Fig. 4 for the shipwreck. Despite these small differences, the observed flow is not reliable because of the poor contrast of these images that leads to a great risk of mistakes in terms of correlation. Lastly theoretical flow is finer than the observed one, i.e. displacement estimation can be finer than the pixel resolution.

**Fig. 2 : results of the correlation of two small images of the detected moored mine**

**Fig. 3 : Comparison of the two optical flows in the case of moored mine detection**
5. CONCLUSION

In this article a process model has been developed in order to start with a robust target tracking in an underwater environment. This model is derived from the vehicle process model based on navigation data. It has been tested on real data supplied by the French organism GESMA. First results are encouraging and sufficient to carry on with the development of a Kalman filtering, in progress today.

REFERENCES

ATTITUDE AND ROTATION ESTIMATION USING HOUGH TRANSFORMATION FOR NOISY SONAR IMAGES

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Abstract: Several researches estimate locations and attitudes of underwater vehicles through seafloor patterns in onboard video or sonar images. However, seafloor maps should be prepared in advance and the images are usually noisy and obscure. In practical images only the sea surface and the seafloor are able to be identified. A new attitude estimation technique based on acoustical image processing has been developed using imaging sonars with Hough transformation for noisy sonar images in our research. The method estimates onboard sonar attitude parameters from the location of the reference plane like the sea surface in the sonar image. In this paper an additional approach of rotation parameter estimation is introduced using these attitude parameter time variations. It does not require the Hough transformation, hence it is useful for smaller and lighter ocean observing platforms. After summarizing the attitude estimation based on the Hough transformation, the new procedure is explained referring its coordinate system which is one of the most significant features of the method.

Keywords: Attitude estimation, AUV positioning, Imaging sonar, Image processing, Hough transformation
1. INTRODUCTION

For wide range ocean surveys, long time availability is required for observing systems including a variety of types of autonomous vehicles. One of the most traditional approaches is saving energy for long lifetime operations. The engines and the fuels/batteries are usually focused for improvements, however, sensors should be taken into account because of their relatively large power consumption. It is possible to abridge sensors through a sensor aggregation.

On the other hand, imaging sensors show their usefulness in many underwater observations recently. They can survey wide areas quickly and effectively generating multi-dimensional data sets. Among them imaging sonars are powerful for surveys of long distance and muddy water. They can obtain three dimensional environments. Therefore they can derive sensor attitudes relative to the terrains if any types of seafloor maps are acquired in advance using a shape matching as in [1], which is similar to video image based methods as in [2] and [3]. If the attitudes derived from the imaging sonars show enough accuracies to utilize for a vehicle control and other services, it can be said that they have the potential to serve for attitude sensors. Removing sensors also enhance the possibility of less complex vehicle designs and leads to lower frequencies of the vehicle maintenance.

Though it is possible to realize sensor aggregations for some high-accuracy imaging sonars like side-scan sonars, images are sometimes too noisy to recognize seafloor terrains. As a noise robust method a new vehicle attitude estimation was proposed using imaging sonars with Hough transformation in the previous paper [4]. The method estimates onboard sonar attitude parameters from the location of the reference plane in the sonar image. The sea surface is used as the reference plane in the paper. The derived accuracies are within three degrees in rolls and pitches, 5 meters in depths excluding some configurations. The method is still on a development stage considering these accuracies, however, it shows a potential to replace attitude sensors. Although it cannot figure yaws, it is useful for the minimum survival control for vehicles.

After the publication a new advance had been developed for further safer vehicle navigations. The new progress extracts rotation parameters from rolls and pitches. In the new method vehicle attitude changes are described by two different coordinate transformation matrices. One of these matrices provides a new aspect for coordinate descriptions.

Rotations are essential elements not only for underwater vehicles. For example, underwater instruments tethered with floats sometimes drift from the moored points and roll around themselves or the dangling wires because of currents or tides, which leads to loss of the instrument attitudes. It is important to figure their sensing directions from the rotations.

In this paper the Hough transformation attitude estimation is summarized at first. Then these coordinate systems are explained. And the new method is described based on these coordinate systems for estimating the rotation parameters include the rotation velocity.

2. USUAL TWO DIMENSIONAL HOUGH TRANSFORMATION

The Hough transformation is widely applied in line detections on two dimensional noisy images, and has been mainly studied on the ground using camera images as in [5]. It is robust not only for noise but for line disconnections. It needs heavy computation except for straight lines. The line detection procedure is as follows.

At first, the line equation is defined as

\[ \rho = x \cos \theta + y \sin \theta. \] (1)
The $\rho$ is the line distance from the origin and the $\theta$ is the angle between the $x$ axis and the perpendicular to the line (Fig.1), and both of them are referred to as the line parameters. Next the image is binarized by a threshold to determine line area candidates. And all parameter combinations are calculated for lines passing through each pixel on line area candidates. After that, all parameter combinations are plotted as a parameter curve on the parameter space for each pixel on line area candidates. The parameter space axes are line equation parameters. For example, if the points A, B, C are detected in Fig.1, the parameter curves A, B, C are plotted as in Fig.2. These curves intersect each other. And the most frequently intersected point on the parameter space shows the most possible parameter combination. So, from Fig.2, the line equation is determined as

$$\rho_0 = x \cos \theta_0 + y \sin \theta_0.$$  

3. ATTITUDE ESTIMATION USING HOUGH TRANSFORMATION

As in the previous paper, the sea surface is regarded as a reference plane for estimating vehicle attitudes. Identifying the plane location has the same meaning of estimating attitude parameters. Most of three dimensional extensions of Hough transformation formulate that the plane equation is an analogy from two dimensional cases. They use $\rho$ which is the distance from the origin to the plane as in Fig.3, and the equation becomes like

$$\rho = x \cos \theta \cos \phi + y \cos \theta \sin \phi + z \cos \phi.$$  

Although this extension looks natural and easy to treat, parameter conversions are needed for attitude estimations.

Instead of these extensions, attitude parameters are directly chosen as the plane equation parameters as in Fig.4. The plane equation is expressed by roll $\alpha$, pitch $\beta$, and depth $d$ as

$$d = -x' \sin \alpha + y' \cos \alpha \sin \beta + z' \cos \alpha \cos \beta.$$  

$(x',y',z')$ means sonar fixed coordination and it is equivalent to ground fixed coordination $(x,y,z)$ for $\alpha=0$, $\beta=0$.

The first step in the Hough transformation attitude estimation is binarizing the three dimensional image by a threshold to determine plane candidates. Then all parameter combinations are calculated for planes passing through each pixel on plane candidates. After that, all parameter combinations are plotted as a parameter plane in the parameter space for each pixel in plane candidates. The parameter space axes are plane equation parameters. Parameter planes intersect each other, and the most frequently intersected point on the parameter space is the most probable plane parameter combination in the real space.
4. ROTATION ESTIMATION

The new extension utilizes time variation of rolls and pitches to extract rotation parameters. The most important idea is describing vehicle attitudes by two different coordinate systems.

**A. Coordinate Systems**

In the new method vehicle attitude changes are described by two different coordinate transformation matrices. One is the matrix $U$ based on a usual attitude parameter the roll, the pitch and the yaw. It is usual attitude description and it is the coordinate system used in the Hough transformation attitude estimation. Another is the matrix $V$ based on a rotation plane based coordinate system which is newly introduced.

The former matrix $U$ is based on the roll $\alpha$, the pitch $\beta$ and the yaw $\gamma$ as in Fig.5. These attitude parameters are able to be determined by the Hough transformation procedures excluding the yaw. The yaw $\gamma$ was not explicitly displayed in the previous paper, since it was supposed to be zero.

The matrix elements $U_{ij} (i, j = 1, 2, 3)$ are described as follows.

$$
egin{align*}
U_{11} &= \cos \alpha \cos \gamma + \sin \alpha \sin \beta \sin \gamma \\
U_{12} &= \cos \beta \sin \gamma \\
U_{13} &= -\sin \alpha \cos \gamma + \cos \alpha \sin \beta \sin \gamma \\
U_{21} &= -\cos \alpha \sin \gamma + \sin \alpha \sin \beta \cos \gamma \\
U_{22} &= \cos \beta \cos \gamma \\
U_{23} &= \sin \alpha \sin \gamma + \cos \alpha \sin \beta \cos \gamma \\
U_{31} &= \sin \alpha \cos \beta \\
U_{32} &= -\sin \beta \\
U_{33} &= \cos \alpha \cos \beta
\end{align*}
$$

$$
\begin{pmatrix}
U_{11} & U_{12} & U_{13} \\
U_{21} & U_{22} & U_{23} \\
U_{31} & U_{32} & U_{33}
\end{pmatrix}
$$

Another is the matrix $V$ based on a rotation plane based coordinate as in Fig.6. In this coordinate there are four attitude parameters $A$, $B$, $C$ and $D$. The $A$ is the angle between the
rotation plane and the sonar line of sight. The $B$ is the rotated angle from the supposed basic direction. The $C$ is the angle between the rotation plane and the reference plane, which is the sea surface for example. The $D$ is the rotated angle of the rotation axis around the $z$ axis which is the normal of the reference plane from the supposed basic direction. The largest difference from usual coordinate systems is expression by these four parameters not three parameters.

![Rotation parameter configurations in the proposed method.](image)

Using these new parameters the matrix elements $V_{ij}$ ($i,j = 1,2,3$) are described as follows.

\[
\begin{align*}
V_{11} &= \cos B \cos D - \sin B \cos C \sin D \\
V_{12} &= \cos A \sin B \cos D + (\cos A \cos B \cos C - \sin A \sin C) \sin D \\
V_{13} &= \sin A \sin B \cos D + (\sin A \cos B \cos C \cos A \sin C) \sin D \\
V_{21} &= - \cos B \sin D - \sin B \cos C \cos D \\
V_{22} &= - \cos A \sin B \sin D + (\cos A \cos B \cos C \sin A \sin C) \cos D \\
V_{23} &= - \sin A \sin B \sin D + (\sin A \cos B \cos C \cos A \sin C) \cos D \\
V_{31} &= \sin B \sin C \\
V_{32} &= - \cos A \cos B \sin C \sin A \cos C \\
V_{33} &= - \sin A \cos B \sin C + \cos A \cos C 
\end{align*}
\]

(6)

**B. Time Variations**

These two matrices should be equal because they express the same attitude transforms. If the $A$ and the $C$ are constant and the $C$ is known, the $B$’s time variation – the rotation velocity – is determined using the attitude parameter $\alpha$ and $\beta$ from $U=V$. If the roll, the pitch and the rotated angle are $\alpha_1$, $\beta_1$ and $B_1$ for each at time $t_1$, and they turn to $\alpha_2$, $\beta_2$ and $B_2$ at time $t_2$, the rotation velocity $(B_2 - B_1) / (t_2 - t_1)$ is obtained easily from the following equations.

\[
\begin{align*}
\sin B_1 &= \frac{\sin \alpha_1 \cos \beta_1}{\sin C} \quad (7) \\
\sin B_2 &= \frac{\sin \alpha_2 \cos \beta_2}{\sin C} \quad (8)
\end{align*}
\]

Even if the $C$ is not known in advance, it is possible to calculate the rotation velocity since the parameter $A$ and $C$ are derived by the following equations (9) and (10) which are results of $U=V$.

\[
\begin{align*}
\cos \alpha_1 \cos \beta_1 - \cos \alpha_2 \cos \beta_2 &= \tan A (\sin \beta_2 - \sin \beta_1) \quad (9) \\
\cos A \cos \alpha \cos \beta + \sin A \sin \beta &= \cos C \quad (10)
\end{align*}
\]
Moreover the parameter \( D \) is also obtained by eliminating the yaw \( \gamma \) using following equations (11) and (12).

\[
\sin \gamma = \frac{1}{\cos \beta} \left\{ \cos A \sin B \cos D + (\cos A \cos B \cos C - \sin A \sin C) \sin D \right\}
\]

\[
\cos \gamma = \frac{1}{\cos \beta} \left\{ -\cos A \sin B \cos D + (\cos A \cos B \cos C - \sin A \sin C) \cos D \right\}
\]

From these equations, the yaw \( \gamma \) is derived relative to a reference direction. These relationships are also applicable to slightly time varying cases.

The important point is the new method does not necessarily require the Hough transformation attitude estimation since it only needs the attitude parameters rolls and pitches. Therefore it is open for lighter attitude estimation algorithms for example a least square fitting of the reference plane considering lower computing power platforms for long lifetime vehicles.

5. SUMMARY
A new rotation estimation method is presented for underwater vehicles utilizing the newly introduced coordinate system. If the rotation axis is constant it is easy to derive the rotation velocity by this method. It is effective for any vehicle attitude estimation methods, although it originally based on the Hough transformation attitude estimation. It is also applicable to video image vehicle positionings. Moreover three distortion corrections are explained for the Hough transformation attitude estimation. These are essential procedures toward practical use in the real ocean.

The Hough transformation based attitude and rotation estimation is applicable not only to the AUVs but also to tethered instruments like buoys with imaging sonars. In spite of its restrictions and calculating powers demanded, the Hough transformation method will be widely applied to ocean investigation instruments as robots on the ground.

REFERENCES
Regular Session VII

Sound propagation in shallow and deep water
Temporal variability of mid-frequency sound propagation in a shallow water area (Adventure Bank) of the Mediterranean Sea

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Abstract: One of the goals in shallow water acoustics is to understand the temporal variability of transmission loss (TL) and acoustic multi-path in response to fluctuating environmental parameters. To this end we conducted experiments with both a slowly drifting and a fixed acoustic source on the Adventure Bank south of the Island of Sicily. Whereas the drifting source was lowered from a ship, the fixed acoustic transmitter was installed on a sturdy tripod 5 m above the sea-floor. At a distance of 6.7 km a vertical nested array of hydrophones was moored. Various acoustic signals in the frequency range between 2.1 and 5.6 kHz were transmitted. Narrow band sweeps with 10 different center-frequencies were used to study the frequency dependence of transmission loss and acoustic multipath and their variation on time scales from minutes to hours. BPSK (Binary Phase Shift Keying) communication signals were used to look at the variability on time scales of seconds. With the acoustic systems being slowly drifting or stationary the only cause for acoustic variability are variations of oceanographic conditions. These were monitored by moored and ship-borne instruments. As the sound-speed profile was downward refracting the TL was smaller for the lower than for the upper hydrophones. The absolute variation of TL was largest for the lower hydrophones. Depending on frequency and hydrophone position the TL fluctuated by up to 10 dB over only a few minutes. The multipath structure shows that the primary reason were variations of the strength of the main arrival(s). Our description and analysis of the acoustic and environmental observations is aided by numerical simulations.
A FINITE ELEMENT SOLUTION OF THE STANDARD PARABOLIC EQUATION IN CURVILINEAR COORDINATES APPLIED TO UNDERWATER SOUND PROPAGATION

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Abstract: The standard parabolic equation expressed in curvilinear coordinates is considered in a cylindrically symmetric, semi infinite inhomogeneous waveguide, bounded by an irregular lower boundary. The initial-boundary value problem considered, treats the irregular boundary as an interface and considers an artificial layer extending beyond it, with an artificial horizontal boundary along which a nonlocal, transparent boundary applies. The artificial boundary condition used is an integral representation of the exterior wave field mapped on the artificial boundary; the cases of a constant index of refraction and a linear, when squared (as function of depth) one, are considered. This complex physical waveguide is reduced to an orthogonal computational domain by the means of a numerical transformation to curvilinear coordinates fitting the irregular interface. This technique is of practical importance since a mesh generator, using bathometric data, may be used to compute numerically the transformation of coordinates. This technique is straightforward and effectively extended to 3D geometries. The transformed initial--boundary value problem (on the orthogonal computational domain) is discretized by a Crank-Nicolson scheme which advances in range a continuous, piecewise linear finite element solution. The resulting computational model is applied to the simulation of the propagation of underwater, cylindrically symmetric sound waves, emitted by a harmonic source. Numerical simulations have been performed for several characteristic test cases. They demonstrate the effectiveness of the overall method through the proper handling of complex interface topographies.
THE INVESTIGATIONS OF WAYS OF ACOUSTICS ENERGY TRANSMISSION THROUGH HULL OF SHIPS INTO THE WATER

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Abstract: The paper present results of investigation of spatial distribute of ship’s noise. The comparison of physical dimensions of researched ship with spatial distribution of measured under hull sound allows to describe the areas of the maximum transmissivity of acoustics energy. Acquirement knowledge is necessary during project phase of ships construction especially military applications. For the naval ship this results of research permit skilfully operate in underwater mine zone.

Keywords: acoustics ship’s signature, underwater noise, hydroacoustics
1. INTRODUCTION

The Hydroacoustics Institute of the Polish Naval Academy more than thirty years leads investigation related with experimental research the transmission of the acoustic energy generated by moving objects into the sea water. The research are provided for the following aims:

— Passive ship defense,
— Monitoring self underwater noise,
— Classification and identification objects,
— Reconnaissance of sea object,
— Description of the technical state of ship,
— Improve acoustics research method and skills,
— Development of hydroacoustics systems.

This paper presents chosen and results of investigations lead in the Polish Naval Academy for The Polish Navy and used to determine the position of ships noises’ sources and areas of the highest noise. The main aim of investigations were:

1. To determine of locations of underwater noise’s sources,
2. To localization the areas of the highest noise levels of hull of ship,
3. To expand the methods of hydroacoustics investigation about longitudinal changes of hydroacoustic noise chosen ships.

2. METHOD OF RESEARCH

The basic method of investigations is dynamic method of measurement underwater noises. The sensors there are directly on the permanent special construction situated on the bottom of sea. The directional buoys lead ship straight above underwater section of hydrophones. This method allows calculate longitudinal and transverse (spatial) distribute of underwater sound generated by hull of moving ship. The measurement transverse and spatial distribute of underwater noises required more than one sensor, but investigation longitudinal distribute acoustic energy emitted by hull of ship don’t require more than one sensor.

The appearance of range where were made investigations is presented in Fig.1. In dynamic method of hydroacoustic measurements two groups of researchers take part in investigations. First group on the acoustics range measure underwater noises and second group on the board of investigated ship measure vibration the main sources of sound. This method allows to check correctness measured records of vibration and underwater noise [1].
3. RESULTS OF RESEARCH

The previous investigations show that knowledge about position of harmonics of generating set in spectrum of ship’s underwater noise is very important [2,3]. Harmonic, which a generating set is a source almost always appears in the spectrum of underwater sounds, because a movement of the ship is impossible without supplying the current to power electrical system [4].

In this examinations the harmonics related with the generating sets were a base of localization another ship’s mechanisms. Idea of location the main sources of underwater noise is presented in Fig. 2.

The method based on characteristics harmonics tracking in $k$ records, where $k$ depend on length of measurement data and length of ship.

The dozen ships were being analyzed, every at different parameters of the operating of ship's mechanisms. It was noticed that every ship have an individual image, which distinguish it from other ships. Some results for four chosen ships is presented in Fig. 2 and 3. In this article four ships of different types and sizes moving with the comparable speed from 9 to 11 knots were committed comparing, it is shown on Table 1. We can see that the distances between main sources underwater noises aren’t comparable. It is allowed in the further investigations to test all sources witch appear in the tested files and to describe its shape. The calculated distances are similar to really length of investigation ships.
Fig. 2. The dynamic measurement of ship’s noise – the method of measurement and analyze of longitudinal distribution of sources of underwater noise ship no 1, speed 10 knots.
Fig. 3. the analyze of longitudinal distribution of sources of underwater noise: Upper diagram: Ship 2 – Speed 9.6 kts, Generating set marks doted line, the accentuation line - engines. Middle diagram: Ship 3 – speed 11 kts, Generating set marks doted line, the higher lines - engines. Lower diagram: Ship 4 – speed 9 kts, Generating set - accentuation line, the higher lines - engines.
<table>
<thead>
<tr>
<th></th>
<th>Gensets [m]</th>
<th>Engines [m]</th>
<th>Propellers [m]</th>
<th>Difference G-E [m]</th>
<th>Difference E-P [m]</th>
<th>Difference G-P [m]</th>
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<tbody>
<tr>
<td>Ship 1</td>
<td>52,47</td>
<td>34,19</td>
<td>-</td>
<td>18,28</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>10 kts</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ship 2</td>
<td>49,05</td>
<td>58,97</td>
<td>-</td>
<td>-9,92</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>9,6 kts</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ship 3</td>
<td>51,8</td>
<td>38,88</td>
<td>69,36</td>
<td>12,92</td>
<td>-30,48</td>
<td>-17,56</td>
</tr>
<tr>
<td>11 kts</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Ship 4</td>
<td>26,79</td>
<td>36,46</td>
<td>49,83</td>
<td>-9,67</td>
<td>-13,37</td>
<td>-23,04</td>
</tr>
<tr>
<td>9 kts</td>
<td></td>
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</table>

Table 1. The position of ship’s various machinery in relation to position of sources generating sets.

4. SUMMARY

The preliminary research of position of the areas of the highest noise levels of hull of ship allows to conclude:

1. Based on the measured underwater noise generated by ship it is possible to get conclusion about distances between main sources of underwater noises generated throw hull of investigation ship.

2. The experience gathered during research localization the areas of the highest noise levels of hull of ship could help minimalisation acoustics signature of ships.

3. The results of research will be considered during configuration of measurements equipment with taking into account the parameters of working main mechanisms (included heavy current engineering system).

4. The results show that it is necessary to have detailed base of location parts of hull and ships’ mechanisms, which will let probably precise classification being based on a location along of objects.

REFERENCES


ROBUST UNDERWATER LOCALISATION OF ULTRA LOW FREQUENCY SOURCES IN OPERATIONAL CONTEXT

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Abstract: This work focuses on acoustic passive localisation (in depth and in range) of Ultra-Low Frequency sources (ULF) in shallow water (50-400m). In this environment acoustic signal can be described by propagation modes with amplitudes depending on the source depth. The range information is carried by mode phases. For recordings of the acoustic field generated by an impulsive ULF (1-100 Hz) source we use a horizontal array of hydrophones at monitored depth. Our approach is based on the combination of Matched Field Processing (MFP) and Matched Mode Processing (MMP). The dispersive modes are separable in the frequency - wavenumber plane (f-k plane) which is the representation space of the signal. MFP requires the simulated acoustic field to be compared with the real acoustic field. For the reference data we use realistic simulations generated with Moctesuma underwater acoustic propagation simulator. Modes extracted from the f-k plane by adapted masks are analysed for localisation purposes. The estimated source depth is found for the best matching between real and reference modes amplitudes. The access to the mode phases allows estimation of source range and mode signs (which improves significantly the localisation performance). We validated the feasibility of the method in shallow water for different operational configurations with known and unknown sources. The performance and robustness of the localisation is very satisfactory even for low Signal to Noise Ratio.

Keywords: underwater localisation, matched mode processing, f-k transform, sign of mode, source spectrum estimation, horizontal line array
1. INTRODUCTION

Passive source localisation in shallow water environment has been studied for many years in underwater acoustics. The acoustic propagation in shallow water (modeled as an oceanic waveguide) is based on normal mode theory. Matched Field Processing (MFP) [1] and Matched-Mode Processing (MMP) [2, 3] are widely used in this context. As MMP is less sensitive for environmental mismatches and requires much less time processing, this technique is more interesting for practical applications and is used in our approach to estimate the source depth. MMP is based on decomposition of the acoustic pressure field in a series of depth-dependent modes. A depth localisation is performed by correlating the real mode amplitudes with the reference (simulated) mode amplitudes for different source depths. The best matching indicates the estimated source depth. Classically MMP was applied on Vertical Line Arrays (VLA) for mode amplitude (excitation factor) extraction. As proposed in [4] we apply MMP on single Horizontal Line Array (HLA) as it is more practical in real applications (towing possibility, simpler deployment, and stability). In this configuration, modes can be accessed in the frequency-wavenumber plane (f-k) which is a 2D Fourier transform of time – radial distance section. In the f-k domain, modes are separable and can be extracted by 2D mask filtering. The localisation method we propose here is suited for ultra-low frequency (ULF, 1-100Hz) sources as in this band information about propagation properties are very relevant. As underwater propagation of ULF signals are complex, a good modeling of the environment is needed. Classically the shallow water environment is modeled as the Pekeris waveguide. As this model is very simple and does not represent well real environments, we propose a new realistic modeling of underwater acoustic propagation called Moctesuma simulator and developed by Thales Underwater Systems. For the source range estimation, we use a method based on the mode phase. The method discussed in this paper makes also possible the geoacoustical parameters estimation [5].

We propose two localisation methods for source depth and range estimations. First, we provide a brief presentation of normal mode theory and give short description of Moctesuma underwater acoustic propagation modeling. After giving details about localisation configuration, we describe depth and range localisation algorithms and give simulations results. As in MMP the reference acoustic field should be generated with a source very similar to the real source, we propose a simple method of source spectrum estimation from recorded data. We also demonstrate that by adding the sign information estimated by the algorithm presented in [6] to find modes amplitudes, the depth localisation performance drastically improves. All simulations presented in this paper are done with Moctesuma realistic underwater acoustic propagation simulator.

2. NORMAL MODES AND MOCTESUMA SIMULATOR

The ULF propagation in shallow water is usually modeled by normal mode theory. We introduce the simplest model of oceanic environment: the perfect waveguide (Fig. 1).

![Perfect waveguide](image)

Fig.1: Perfect waveguide.
This model is made of a homogeneous layer of water between perfectly reflecting boundaries at surface and sea bottom (at depth \(D\)). The water layer is characterized by a velocity \(V_1\) and a density \(\rho_1\). Here, the study is presented for an omnidirectional harmonic point source, with a frequency \(f\) located at depth \(z_s\), and at range \(0\). However, the results are similar for a broadband source. Acoustic pressure \(P(r,z,t)\) received at reception-point of coordinates \((r,z)\) can be expressed by \(P(r,z,t) = p(r,z)\exp(2i\pi ft)\), where \(p(r,z)\) satisfies the Helmholtz equation and is, at long range, a sum of modes:

\[
p(r,z) = \sum_{m=1}^{\infty} \psi_m(z_s)\psi_m(z) \frac{\exp(-2i\pi k_m r)}{\sqrt{k_m r}}
\]

with \(A\) a constant, \(\psi_m\) a modal function of order \(m\) and \(k_m\) a horizontal wavenumber of \(m\)'th mode. By homogeneity with the temporal frequency \(f\), \(k\) is defined as a spatial frequency \(k = f/V_1\). The mode wavenumber spectrum is discrete and each mode is associated with a unique wavenumber. The mode amplitude \(\psi_m(z_s)\) is a function of the source depth \(z_s\):

\[
\psi_m(z_s) = \sqrt{\frac{2}{D}} \sin(2\pi k_m z_s)
\]

with \(k_m = (2m - 1)/4D\). This short theoretical introduction of normal mode theory on the perfect waveguide example shows that the mode excitation factor is function of the source depth \(z_s\). This property is at the basis of MMP.

In MMP to perform localisation, one needs to simulate the acoustic reference field for a set of different source depths. The quality of localisation depends on the matching accuracy between the real acoustic field and the simulated acoustic field (replica). Therefore, a good choice of the propagation model is required. As Pekeris waveguide considers just a few propagation phenomenon (reflection, phase shift, attenuation etc.), it is not suited as a model. Many simplifications such as constant sound speed profile or two-layers bottom imply that the simulated propagation does not correspond correctly to real propagation and cannot be efficient in real environments.

In our study, we propose to use Moctesuma as a realistic underwater propagation simulator provided by Thales Underwater System. With Moctesuma one can simulate an underwater acoustic propagation for range-dependent environment providing all information such as sea state, sound speed profile, seabed type, etc.... Moctesuma is based on mode theory and gives us a complete acoustic field in time for each sensor. In our method, we use Moctesuma to simulate the reference acoustic field and to access the environmental parameters such us horizontal wavenumbers and modes excitation factors. The horizontal wavenumbers are used to generate 2D adapted masks in f-k plane for matched mode filtering. The mode excitation factors allow a direct extraction of simulated mode signs from the simulations (no estimation needed).

We simulated the environment studied in this paper (described in the next paragraph) with Pekeris model and Moctesuma modeling. The differences between two models are remarkable (Fig. 2). This is due to a strong gradient in sound speed profile and type of sea bed that can not be modeled by Pekeris model.
3. EXPERIMENT CONFIGURATION

The data analysed in this paper are done in a shallow water environment located in the Mediterranean Sea during summer. The seabed is at 130 m. Sound speed profile in water (with a strong negative gradient (approx. 25 m/s) and in seabed are presented in Fig.3. The source is the transitory signal (Fig.4).
At the reception, we use a horizontal line antenna (HLA) of length 800 m with 240 omnidirectional hydrophones. Frequency sampling is 1 kHz and for the reason of ULF band, an undersampling (with factor 4) is applied before processing. The distance between source and the antenna is 10 km. Source depth is 80 m and antenna depth is 130 m (seabed - minimum of sound speed profile). Both of them are motionless. The presented analyses are done for 5dB signal-to-noise ratio.

4. LOCALISATION IN DEPTH AND RANGE

The depth and range estimations are based respectively on information given by modes amplitudes and modes phases.

Source depth estimation

The localisation algorithm based on MMP consists in finding the best matching between a set of modes amplitudes extracted from the real data (see Fig.5a) and a series of sets of modes amplitudes extracted from the reference data simulated for different source depths.

To improve the dynamics of modal representation and survey the spatial aliasing, before calculating f-k representation, we apply on time-radial distance section a velocity correction [4]. The modes are filtered in the f-k plane using adapted binary masks which are built at basis of modes characteristics provided by Moctesuma modeling simulator (see Fig.5b). The filtering is the same for the real and the reference data which makes the method less sensitive to the mismatch between real and simulated environment. A trace of mask for the mode is given by its wavenumbers. The mask is constructed by dilatation, which is done independently in both domains (temporal and spatial). This allows a better matching of masks to modes in f-k plane [4]. The masks of different modes can not overlap. Once the modes are extracted, the mode excitation factors \( c_m(z) \) are calculated as a mean over frequency domain. Next, a normalisation using the closure relationship between modes is applied [4] such as: \( \sum_m c_m^2(z) = 1 \). The excitation factors due to the processing are all positive, which is not natural (modal function is a real function). The absence of information about modes signs causes deterioration of localisation performance. For this reason, we add modes signs (estimated as proposed in [6]) to excitation factors to improve localisation performances (see Fig.6). The best matching between the real and the simulated data is found by a contrast function \( G \) maximisation:
where \( N_{\text{mod}} \) is number of modes, \( c_m^{\text{real}} \) and \( c_m^{\text{simul}} \) are respectively mode excitation factors for the real and the simulated data. Classically, for ULF sources in shallow water environments, there is a maximum of ten propagating modes due to cut frequencies existence. In the example of localisation presented in this paper we used first 5 modes for processing purposes. In MMP the knowledge of several parameters is crucial: geoacoustical characteristics of the environment and spectral properties of the source. Actually, the knowledge (at least approximate) of the environment characteristics is not a problem. The second issue is more problematic. We propose a simple method of source spectrum estimation based on first mode analysis:

\[
S'(\omega) = A(\omega) \cdot \frac{1}{N_s} \sum_{n=1}^{N_s} X_n^{\text{mod1}}(\omega)
\]

where \( S'(\omega) \) is an estimated source spectrum, \( A(\omega) \) is a spectral factor correcting propagation attenuation over frequency, \( N_s \) is a number of sensors and \( X_n^{\text{mod1}}(\omega) \) is a spectrum of the first mode on the \( n^{\text{th}} \) sensor. The estimation can be decomposed in few steps:

- filtering of the first mode (horizontal mode, always excited) in f-k plane,
- calculation of inverse 2D Fourier transform on the filtered mode,
- calculation of the 1st mode spectrum for each sensor,
- calculation of the mean spectrum over all sensors,
- correction of the mean spectrum for spectral acoustical propagation attenuation.

Let us consider an example of source depth-localisation with configuration given in section 3. Results of this analysis are given in Fig.6.
We consider 3 different cases:

- spectral characteristics of the source are known and these are used for simulation of the reference data (Fig.6a);
- spectral characteristics of the source are not known; we use a transient broadband ULF signal with flat spectrum to simulate the reference data (Fig.6b);
- spectral characteristics of the source are not known, but we estimate them by the method described above (see eq.5); the reference data is simulated with the source estimated directly from the recorded data (Fig.6c).

The results presented in Fig.6 demonstrate how it is important to know spectral properties of the source. If the source used for simulation of the reference data is different from the real source, then the excitation of mode excitation factors will not be correct and thus the localisation performance will decrease. The method of source spectrum estimation proposed in this paper improves significantly the performance of localisation in case when the source spectrum is not known. The contrast functions presented in Fig.6a, b, c (in black for the method without mode signs and in red for the method with mode signs) show also the importance of the estimation of the mode signs. The mode signs allow cancelling false “mirror” solutions (due to the oscillating character of the modal function) and thus minimize the value of the contrast function for these error solutions.
Source distance estimation

The estimation of the source range (from the antenna) is combined with the mode sign estimation. The method proposed in [7] is based on the analysis of modes phases, and recently was discussed in [6]. The estimator calculates a cost function based on two modes phases (not necessarily consecutive as long as their numbers are known) and their wavenumbers (given by Moctesuma). For the purpose of range localisation we used 4 couples of modes extracted from f-k representation of the data: 1&2, 2&3, 3&4 and 4&5. We have so 240 estimations of range (for each sensor in the antenna). The obtained results are very satisfactory (see Fig.6d). The mean error is between 5 and 25 m which is a very good result for the range of 10 km. The range estimation algorithm is efficient if the modes are filtered correctly and if the model of the environment (modes wavenumbers) matches well with the real environment.

5. CONCLUSION

In this paper we presented a robust passive localisation method for ULF source depth and range estimation. The method is based on matched mode processing which especially in shallow water environment demonstrates its performance. The extraction and analysis of the mode amplitudes (which carry information about source depth) and the mode phases (which carry information about source range) make possible depth and range localisation. The algorithm presented in this paper demonstrates that the method is ready to be used in operational context. The important point is to simulate the reference data with the source very similar to the real one. Therefore, we propose an efficient estimation of spectral characteristics of the source. The results of depth and range localisation obtained on signals simulated with Moctesuma in realistic geophysical conditions are
very satisfactory and demonstrate the performance of the proposed method. Currently, we are working on adaptation of the method for deep water localisation.

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REFERENCES

SIMPLE APPROXIMATION SCHEME FOR DISPERSIVE PULSE PROPAGATION, WITH COMPARISON TO THE STATIONARY PHASE APPROXIMATION

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\textbf{Abstract:} A standard approximation for dispersive pulse propagation is the method of stationary phase. A new phase space approximation based on the joint position-wavenumber Wigner distribution has been developed, which is very simple to apply and moreover is more accurate than the stationary phase approximation, especially for small time. In addition, a modification to the stationary phase approximation has been developed that improves the accuracy of the approximation for small time. We compare the accuracy of these three approximations, by computing the spatial moments of the resulting pulse over time. In the large time limit, all three approximations approach the same answer. For small time, the Wigner approximation is the most accurate of the three, followed by the modified stationary phase approximation. One of the reasons why the Wigner approximation is so accurate is that it exactly preserves low-order spatial moments of the pulse.

\textbf{Keywords:} dispersion, propagation, stationary phase, moments, Wigner distribution
1. Introduction

Recently, we have proposed a new approximation for wave motion, based on the Wigner phase space distribution, that appears to be more accurate and insightful than the standard stationary phase approximation [3, 7]. In our efforts to better understand the Wigner approximation and assess its accuracy in a general way, we found a way to improve the stationary phase approximation, which we call the modified stationary phase method [4]. In this paper, we compare these three approximations by calculating the spatial moments for each one. In the large time limit, all three approximations approach the same answer. For small time, the modified stationary phase approximation is more accurate than the standard approximation. However, the Wigner approximation is the most accurate (and simplest to apply) of the three approximations, and gives the exact low-order moments for all time.

2. Wave Propagation in Phase Space

We use the notation \( u(x,t) \) for the wave at time \( t \), with spatial Fourier transform \( S(k,t) \),

\[
\begin{aligned}
S(k,t) &= \frac{1}{\sqrt{2\pi}} \int u(x,t) e^{-ikx} dx \\
u(x,t) &= \frac{1}{\sqrt{2\pi}} \int S(k,t) e^{ikx} dk
\end{aligned}
\]

For linear wave equations with constant coefficients, the solution for one mode is given by [6, 8]

\[
u(x,t) = \frac{1}{\sqrt{2\pi}} \int S(k,0) e^{ikx - i\omega(k)t} dk
\]

where \( S(k,0) \) is the spatial spectrum of the initial wave

\[
S(k,0) = \frac{1}{\sqrt{2\pi}} \int u(x,0) e^{-ikx} dx
\]

and \( \omega(k) \) is the dispersion relation for that mode. The relation between the time dependent spatial spectrum, \( S(k,t) \), and its value at time zero is

\[
S(k,t) = S(k,0) e^{-i\omega(k)t}
\]

We consider real dispersion relations \( \omega(k) \) (no damping), by which it follows that

\[
|S(k,t)| = |S(k,0)|
\]

Also, as standard, we define the group velocity, \( v(k) \), by

\[
v(k) = \omega'(k)
\]

We use the phase space of the spatial variable \( x \) and the wave number, \( k \), and for the phase space distribution we use the Wigner distribution,

\[
W(x,k,t) = \frac{1}{2\pi} \int [u(x+\lambda/2,t)u^*(x-\lambda/2,t)] e^{-i\lambda k} d\lambda
\]

If we integrate the Wigner distribution over \( x \) or \( k \), we obtain the "marginals,"

\[
\int W(x,k,t) dx = |S(k,t)|^2 ; \int W(x,k,t) dk = |u(x,t)|^2
\]

We have previously shown that the Wigner distribution at time \( t \) can be obtained from the Wigner distribution at \( t = 0 \), \( W(x,k,0) \), by the approximation [3, 7]

\[
W(x,k,t) \approx W(x-v(k)t,k,0) = W_a(x,k,t)
\]

where we use \( W_a(x,k,t) \), to denote the Wigner approximation. Note that this approximation is very easy to apply, as one just substitutes \( x-v(k)t \) for \( x \) in the initial Wigner distribution, \( W(x,k,0) \). It also gives considerable insight into the nature of dispersive pulse propagation because it shows that each point in phase space travels (approximately) at a constant velocity,
given by the group velocity at that point. We also point out that the Wigner approximation gives the correct frequency marginal \( \langle |S(k,t)|^2 \rangle \), and hence the spatial spectral moments \( \langle k^n \rangle \), of the Wigner approximation are exact. However, it does not give the correct spatial marginal \( \langle |u(x,t)|^2 \rangle \), and accordingly the spatial moments \( \langle x^n \rangle \), are in general not exact, although as shown below, the low-order moments are exact.

3. Stationary Phase Approximation and Modification

The standard stationary phase approximation is given by [6, 8],

\[
\psi_{SP}(x,t) \sim S(k_s,0) e^{i(k_s x - \omega(k_s) t - \frac{\pi}{4} \text{sgn}(\omega(k_s)))} \tag{10}
\]

for each stationary point \( k_s \), which are obtained by solving

\[
\omega'(k_s) = \frac{x}{t} \tag{11}
\]

In the standard derivation of the stationary phase approximation, the spatial phase of the initial pulse is not included in the calculation of the stationary points. In our modification, we do so [4]. We express the initial spatial spectrum in terms of its amplitude and phase as \( \psi(k_s,0) \) by which Eq. (2) becomes

\[
u(x,t) = \frac{1}{\sqrt{2\pi}} \int |S(k,t)|^2 e^{i(k_s x - \omega(k_s) t + \psi(k_s,0))} \, dk \tag{12}
\]

The stationary points are then obtained by requiring that

\[
\frac{\partial}{\partial k_s} (\psi(k_s) + k_s x - \omega(k_s) t) = 0 \tag{13}
\]

which gives

\[
\omega'(k_s) = \frac{\psi'(k_s,0) + \frac{x}{t}}{t} \tag{14}
\]

The resulting modified stationary phase approximation is [4]

\[
u_{MSP}(x,t) \sim \frac{|S(k_s,0)|}{\sqrt{t \omega'(k_s) - \psi'(k_s,0)}} e^{i(\psi(k_s,0) + k_s x - \omega(k_s) t - \frac{\pi}{4} \text{sgn}(\omega(k_s)))} \tag{15}
\]

4. Exact Moments

We assume that the initial pulse is normalized to unit-energy. Given that there is no damping (the dispersion relation is purely real), it follows that (see Eq. (5))

\[
\int |S(k,t)|^2 \, dk = \int |\psi(x,t)|^2 \, dx = 1 \tag{16}
\]

Accordingly, treating \( |\psi(x,t)|^2 \) as a density, the spatial moments of the pulse are defined by

\[
\langle x^n \rangle = \int x^n |\psi(x,t)|^2 \, dx \tag{17}
\]

which can be equivalently calculated by way of

\[
\langle x^n \rangle = \int S^*(k,t) x^n S(k,t) dk \tag{18}
\]

where \( \chi = i \frac{\partial}{\partial k} \). For the first two moments, one obtains [2]

\[
\langle x \rangle = \langle x \rangle_0 + Vt \tag{19}
\]

\[
\langle x^2 \rangle = \langle x^2 \rangle_0 + t(\nu \chi + \chi \nu)_0 + t^2 \langle \nu^2 \rangle_0 \tag{20}
\]
\[ \sigma_{v_r}^2 = \langle x^2 \rangle_t - \langle x \rangle_t^2 = \sigma_{v_0}^2 + 2t \text{Cov}_{x_v} + t^2 \sigma_v^2 \]  

where the notation \( \langle \cdot \rangle_0 \) denotes the value at \( t = 0 \). In the above equations

\[ V = \int v(k) |S(k,0)|^2 dk \quad ; \quad \sigma_v^2 = \int (v(k) - V)^2 |S(k,0)|^2 dk \]  

\[ \text{Cov}_{x_v} = \frac{1}{2} \langle v \chi + \chi v \rangle_0 - \langle v \rangle_0 \langle x \rangle_0 \]  

5. Approximate moments

5.1 Moments of the Wigner approximation

For the approximate Wigner distribution, \( W_a(x,k,t) \), the moments are obtained by \([5]\)

\[ \langle x^n \rangle_t = \int \int x^n W(x-v(k)t,k,0) dx dk = \int \int (x + v(k)t)^n W(x,k,0) dx dk \]  

For the first two moments, we obtain the exact answer, Eqs. (19), (20). In general, for large time \( (v(k)t >> x) \), the moments are

\[ \langle x^n \rangle_t \sim t^n \langle v^n(k) \rangle \quad [\text{for large } t] \]

and for small time, they are

\[ \langle x^n \rangle_t \sim \langle x^n \rangle_0 + n \langle x^{n-1} v(k) \rangle_0 t \quad [\text{for small } t] \]

5.2 Stationary phase moments

Using Eq. (10), and noting by Eq. (11) that \( x u''(k) dk_x = dx \) by Eq. (11), it follows that the moments for the stationary phase approximation are given by

\[ \langle x^n \rangle_t = \int x^n |u_{sp}(x,t)|^2 dx = \int x^n \frac{|S(k_x)|^2}{t\omega(k)} dx = \int \left( t\omega(k) \right)^n |S(k_x)|^2 dk_x \]  

by which we see that the variance is \( \sigma_{v_p}^2 = t^2 \sigma_v^2 \). Comparing with the exact answers (Eqs. (19)-(21)) we see that the stationary phase moments approach the exact answer for large \( t \).

5.3 Modified stationary phase moments

Using Eq. (15), and noting by Eq. (14) that \( t\omega''(k_x) dk_x = \psi''(k_x) dk_x + dx \), we have that the moments of the modified stationary phase approximation are

\[ \langle x^n \rangle_t = \int x^n |u_MSP(x,t)|^2 dx = \int x^n \frac{|S(k_x,0)|^2}{t\omega(k) - \psi(k_x)} dx \]  

\[ = \int \left( t\omega(k) - \psi(k_x) \right)^n |S(k_x,0)|^2 dk_x \]

For the first moment we obtain \( \langle x \rangle_t = -\psi(k_x,0) + t\langle v(k) \rangle \). However \([1, 2]\),

\[ \langle -\psi(k_x,0) \rangle = \int -\psi(k_x,0) |S(k_x,0)|^2 dk_x = \int x |u(x,0)|^2 dx = \langle x \rangle_0 \]

and therefore, we see that the first moment is exact,

\[ \langle x \rangle_t = \langle x \rangle_0 + t\langle v(k) \rangle \]

\[ ^1\text{In reference [5], due to a typographical error, } \langle x^{n-1} v(k) \rangle_0 \text{ appears as } \langle x^{n-1} \rangle_0 \langle v(k) \rangle. \]
For the second moment we have
\[ \langle x^2 \rangle = \langle t^2 \psi^2(k) - 2t \psi(k) \nu(k) + \nu^2(k) \rangle \]
\[ = t^2 \langle \psi^2(k) \rangle + 2t \langle -\psi(k) \nu(k) \rangle + \langle (\psi(k))^2 \rangle \]
by which it follows that the variance is
\[ \sigma_{sl}^2 = \langle x^2 \rangle - \langle x \rangle^2 = \sigma^2 + 2t \langle (\psi(k)) \nu(k) \rangle + t^2 \sigma^2_{\nu} \]
To simplify these expressions, we note that [1, 2]
\[ \langle x^2 \rangle_0 = \left( \frac{B_k(0)}{B(k,0)} \right)^2 + \langle (\psi(k))^2 \rangle \quad \text{and} \quad \sigma_{sl}^2 = \left( \frac{B_k(0)}{B(k,0)} \right)^2 + \sigma^2_{\psi} \]
and
\[ \langle \nu \chi + \chi \nu \rangle_0 = -2 \langle \psi(k) \nu(k) \rangle \]
where \( B(k,0) \) is the amplitude of the initial spatial spectrum. Therefore, we have
\[ \langle x^2 \rangle_t = \langle x^2 \rangle_0 - \left( \frac{B_k(0)}{B(k,0)} \right)^2 + \langle (\nu \chi + \chi \nu) \rangle_0 + 2t \nu \chi + t^2 \sigma^2_\nu \]
where \( \text{Cov}_{\nu} \) is defined in Eq. (23). Comparing Eqs. (36) and (37) with Eqs. (20) and (21) we see that the only difference is the term \( \left( \frac{B_k(0)}{B(k,0)} \right)^2 \). Thus, if the initial spatial spectrum is flat, then the second order moments of the modified stationary phase approximation are exact. Otherwise, the modified stationary phase approximation underestimates the variance in \( x \). Even then, the second order moments of the modified stationary phase approximation are more accurate for small-to-moderate values of \( t \) than are the moments of the standard stationary phase approximation; for large \( t \), both approximations approach the same answer, \( \sigma_{sl}^2 \rightarrow t^2 \sigma^2_\nu \).

In general, for large \( t \) the moments go as
\[ \langle x^n \rangle_t \sim t^n \langle \psi^n(k) \rangle \quad \text{[large } t \text{]} \]
which is the same as the ordinary stationary phase moments. For small \( t \) we obtain
\[ \langle x^n \rangle_t \sim \left( \frac{1}{n!} \left( -\psi(k,0) \right)^n \right) + t \left( \nu(k) - \psi(k,0) \right)^{n-1} \quad \text{[small } t \text{]} \]
which are similar in form, but not identical, to the moments of the Wigner approximation.

6. Conclusion

We compared the accuracy of the stationary phase approximation for dispersive pulse propagation to two recently proposed approximations: a modified stationary phase method [4], which takes into account the spectral phase of the initial pulse, and a phase space approximation based on the Wigner distribution of the pulse [7]. To compare the approximations, we calculated spatial moments of the pulse resulting from each approximation. The modified stationary phase approximation is more accurate than the standard stationary phase approach for small-to-moderate times, and all three approximations approach the same results for large \( t \). The Wigner approximation is the simplest to apply, and also the most accurate, in that it preserves the exact spatial mean and variance of the pulse for all time, for real dispersion relations (no damping). The case of dispersive propagation with damping (complex dispersion relation) will be considered in a future
paper, although we note that taking damping into account is very simple to do with the Wigner approximation [7].

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References

SOUND PROPAGATION MODELING ON INTELLIGENT GIS BASIS

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Abstract: Certain theoretical results and existing experience of sound propagation modeling (SPM) based upon Intelligent Geographic Information System (IGIS) is presented in this paper. IGIS technologies have revealed a wide spectrum of SPM capacities, including an easy access to a huge volume of input data, intuitive, understandable and flexible visual interface and open intelligent subsystem for real time modeling.

The core of SPM forms two models of sound propagation: ray and wave models. Ray model is a typical underwater ray acoustics and is used when the difference between results of two models is not relevant as well as when the time for SPM by wave model is not real for selected conditions. The basis of the wave model is formed by parabolic (pseudo differential) equations.

Modeling process of sound propagation is running under intelligent subsystem control. The proposed approach is flexible enough to embed the expert’s intelligence into a real-work algorithm.

At the modeling optimization two variants of SPM development are considered: parallel computing, immunocomputing embedding and combined approach.

The SPM case study is also described in the paper. Some results of sound propagation modeling for different regions from underwater acoustics’ point of view are discussed.
1. INTRODUCTION

Computer model for acoustic field calculation constitutes an original software product incorporating a proper user’s interface, calculating module and data bases that provide the modeling process with source data. Such computer model structure sufficiently well supports realizing the comparatively simple procedures for hydro acoustic calculations accounting, at that, for a limited number of environment factors. The procedures complication, their combined use aimed at increase of accuracy and field’s assessment efficiency, leads to the necessity of searching for techniques ensuring an access to different geospatial data as well as to realizing the methods managing an interaction between various computation modules within a common model. The IGIS capacities deploying furnishes with some solutions for the above problems.

An aggregate of models for acoustic field calculations and range assessment of hydro acoustic means integrated into IGIS constitutes a system of IGIS based hydro acoustic calculations.

The system of hydro acoustic calculations (SHC) core is the complex of hydro acoustic calculation models realizing different procedures assessing the hydro acoustic field parameters. It is worth noting, that by now the theoretic basics of the underwater acoustics are sufficiently developed. Due to the above fact the common understanding of a phenomenon of sound propagation in the ocean as well as of mechanisms affecting it is arrived at. The developed theory can be used in SHC at numerical calculations of the sound field created by a definite source as well as at calculations of the hydro acoustic means’ range. However, certain difficulties limiting the accuracy of such calculations are encountered.

The first one is an absence of adequate information about the sound speed in the ocean regarded as a function of spatial coordinates and time. This problem solution for SHC lies in the plane of developing and realizing the methods for environmental parameters forecast based on the analysis of currently available information and analysis of many years observations.

The second problem consists in analytical and computational complexity of acoustic fields calculations according to the already known ocean characteristics. The approaches to this problem solving consist in arranging for a rational choice of a mathematical model category that would support the calculations’ sufficient accuracy balance and efficiency as well as in realizing a complex of measures aimed at deploying the maximal computational capacities of SHC technical means.

The choice of the acoustic field calculation model category is determined by physical premises with regard to parameters of environment where different acoustic fields are being formed, namely, homogeneous ocean of a constant depth; heterogeneous stratificated ocean of a constant depth; quasi stratificated ocean (with slowly altering properties along the horizontal coordinates) of a constant depth; homogeneous ocean with a free horizontal
boundary and hard bottom having a permanent slight tilt; 2D heterogeneous ocean; randomly heterogeneous ocean.

It should be noted that the rules of the model choice are assumed to account for such factors as the signal source and receiver characteristics, distance between them, the signal frequency, ocean depth.

Not dwelling on the methods’ details note the following:

- acoustic field model in a form of normal waves’ sum is used for long distances;

- at distances from the source comparable with the water area depth the attenuating wave modes cannot be neglected, and many of them should be taken into account at calculating the sound field. At that, the models based upon Hankel transform are used [2].

- method of multiple dissipation is universal and applicable when the sound pressure in a given point can possibly be represented as a sum of a direct wave and the environmentally dissipated waves;

- the models can be simplified by transforming to asymptotic form valid on condition that an acoustic wave length is short against a distance the sound speed undergoes significant alterations along. The asymptotic representations are mutually transformed as well.

- the asymptotic representations of a sound field by the multiple dissipation method could be interpreted via rays used in geometric acoustics. So, this representation is called the ray theory.

In practice the sound speed along the acoustic energy propagation distance is subjected to alterations in horizontal as well as in vertical dimension, i.e. it is heterogeneous by two coordinates. Moreover, the horizontal dimension as a rule affects the sea depth and rough sea elevations. This is why the described models incompletely satisfy the adequacy requirements of the acoustic field calculation. Specifically the acoustic field parameters calculation errors manifest at long distances of low frequency signals propagation.

2. WAVE MODEL ON IGIS BASIS

For the above listed conditions a high accuracy and efficiency of the field calculations is provided by the method of pseudodifferential parabolic equations (PDPE) that is adequately described by [1-2] and consists in a specific algorithmic realization of the cross-sections unilateral method. The given method can be considered as a universal SHC method, particularly allowing solving problems of acoustic field calculation for the heterogeneous stratificated constant depth environment.

Problem of calculating by PDPE method a sound pressure field in 2D heterogeneous waveguide with uneven surface $P(x,y)$ of the lumped on axis $z$ point source $4\pi\delta(0,z-z_s)$, that in homogeneous environment provides for a unit pressure at the unit distance, could be reduced to solving the system of acoustics equations:
\[
\begin{align*}
\frac{dP}{dx} - i \omega \rho V_x &= 0 \\
\frac{dP}{dz} - i \omega \rho V_z &= 0 \\
- i \omega / \rho C^2 + \frac{dV_z}{dz} + \frac{dV_x}{dx} &= 4 \pi \delta (0, z-z_x),
\end{align*}
\]

in a band

\[(x, z) \in (-\infty, \infty) \times [0, H],\]

with boundary conditions

\[P(x,0) = 0, P(x,H) = 0\]

\[\left\{ \text{Im} k^2 (x, z) > 0, (x,z) \in (-\infty, \infty) \times [0,H] \right\} \Rightarrow |P(x,z)| < M,\]

being sought as:

\[P(x,z) = \sum_k C_k(x) \phi_k(x,z), k \in \mathbb{N}\]

amplitudes \(C_k\) of local normal waves \(\phi_k(x,z)\) as a matter of fact present the Cauchy problem solution:

\[\frac{dC_k(x)}{dx} - i \xi_k(x) C_k(x) = - \sum_i \gamma_{kl} (x) C_i(x) \frac{\xi_i + \xi_k}{2 \sqrt{\xi_i \xi_k}},\]

\[C_k(0) = \frac{1}{2 \sqrt{\xi_k(0)}} \phi_k(0,z),\]

and local normal waves \(\phi_k(x,z)\) and their local wave numbers \(\xi_k(x)\) as a matter of fact present problem solution on proper values:

\[\begin{align*}
\rho(x,z) \frac{d}{dz} \frac{1}{\rho(x,z)} \frac{d\phi_k(x,z)}{dz} + k^2 (x,z) \phi_k(x,z) &= \xi_k^2 (x) \phi_k(x,z) \\
\phi_k(x,0) &= 0, \quad \phi_k(x,H) = 0
\end{align*}\]

Where: \(\gamma_{kl}\) - coefficients of local normal waves interaction.

PDPE method uses the basis invariant method formulation and build upon it special grid calculation pattern. PDPE method computer realization performs the necessary at each distance’s step interpolation of environment properties, building matrices \(\hat{A}\) and \(\hat{A}\) in each cross-section, calculations bases on definite pattern and the calculated field derivation. Where \(\hat{A}\) and \(\hat{A}\) are tridiagonal matrices such that:
\[
\hat{A}^{-1}\hat{A} \approx \rho(x,z) \frac{1}{dz} \frac{d}{d\rho(x,z)} \frac{d}{dz} + k^2(x,z).
\] (8)

Results of calculating the acoustic field parameters by PDPE method in sea environment layered models completely coincide with the results of calculating by normal waves’ method in the same models. In 2D heterogeneous environments the considered PDPE method realization neglects the waves dissipated from the environment horizontal heterogeneities in the direction opposite to the waves propagation and, thus, somewhat simplifies interaction between local normal waves moving in the propagation direction. The upper bound of the frequency range, where PDPE method is applicable, is only limited by the calculation speed and memory size available at the used computer system.

The lower bound of the frequency range of the PDPE method applicability is the frequency, under which the sound propagation acoustogravitational effects should be accounted for (to consider conjointly the inner and acoustic waves propagation), and whose value is measured in units of hertz [1-2].

Major disadvantage of the wave methods at their implementation in software products is the algorithms’ complexity and recurrence, and consequently relatively low efficiency at receiving results. PDPE method allows enhancing the calculations’ efficiency. Fig.1 shows the results of the initial acoustic field modeling by the model realizing the algorithm of pseudodifferential equations solving.

![Fig.1 Results of the primary acoustic field modeling (wave method)](image)

As follows from the theory and confirmed by computer modeling at high frequencies the wave methods support low efficiency of acoustic field calculations and that would negatively affect the efficiency of GIS using the results of acoustic fields’ calculations in dynamically changing environment. In the considered case the methods based upon the geometrical optics approximations could be recommended for the applied problems solving in GIS.
3. RAY MODEL ON IGIS BASIS

In ray methods for the field calculations the acoustic wave over a limited segment is treated as a flat one and on the above basis introduced the notion of a ray as a line whose tangent at each point coincides with the wave propagation direction. Such approach allows speaking about sound propagation along the rays, thus, diverting from its wave nature.

Substitution of the wave equation by a ray approximation significantly simplifies the field description task because the wave equation contains no time derivatives and consequently is frequency independent. Moreover, it allows developing a system of ordinary differential equations describing an individual ray trajectory. At that it should be taken into account that the ray pictures development is quite a different task when the sound speed changes in directions of all three coordinates.

The acoustic field calculation by the ray approximation method is based on calculating the propagation anomaly. For an incoherent summation the anomaly is calculated by the following formula:

\[ A(r) = \sum_{j=1}^{N} F_j k_{nj}^n k_{Dj}^m R_j(\theta_{np}), \]  

(9)

where \( F_j \) – the \( j \)-th ray focusing factor, determined by the acoustic rays’ refraction degree and calculated at \( r \) distance from the source at the angle of departure \( \theta_{ui} \) by formula:

\[ F_j = \frac{\left[ r^2 + (h_u - h_{np})^2 \right] \cos \theta_{ui}}{r \left( \frac{\partial r}{\partial \theta} \right) \sin \theta_{np}}. \]  

(10)

where \( h_u \) – depth of the source submergence; \( h_{np} \) – depth of the receiver submergence; \( \theta_{np} \) – ray glancing angle on the receiver on the receiver sea line.

The reflecting factor against intensity from the surface of the \( j \)-th ray falling under the glancing angle \( \theta_{nj} \) is calculated as:

\[ K_{nj} = \exp \left[ 2 \left( -0.3kH \sin \theta_{nj} \right) \right], \]  

(11)

where \( k = 2\pi\lambda^{-1} \) – wave number; \( H \) – wave height depending on wind-induced wave force.

The reflecting factor against intensity from the bottom of the \( j \)-th ray at the glancing angle \( \theta_{Dy} \) for the flat bottom areas is evaluated by the formula:

\[ K_{Dy} = \frac{\left( \frac{\rho_D}{\rho} \sin \theta_{Dy} - \sqrt{\left( \frac{c}{c_D} \right)^2 - \cos^2 \theta_{Dy}} \right)^2}{\left( \frac{\rho_D}{\rho} \sin \theta_{Dy} + \sqrt{\left( \frac{c}{c_D} \right)^2 - \cos^2 \theta_{Dy}} \right)^2}, \]  

(12)

where \( \rho \) and \( c \) – sound density and speed in the near-bottom water layer; \( \rho_D \) and \( c_D \) – sound density and speed in the upper bottom soil, \( R_j(\theta_{np}) \) – characteristics of the vertical
plane aerial directivity; \( m \) and \( n \) – number of the \( j \)-th ray reflections from the surface and the bottom correspondingly. \( N \) – number of rays, arriving upon the point of observation.

The following relation is used to approximately calculate the acoustic field based on the formulae within the hydroacoustic calculations system:

\[
N_{np}(r) = S + N_0(r) + A'(r),
\]

where \( N_{io}(r) = 2 \log \left[ \frac{p_{io}(r)}{p_0} \right] \) – level of acoustic pressure in a real pelagic medium; \( S = 20 \log \left( \frac{p_r}{p_0} \right) \) – radiation level at a unit distance from the source; \( N_0(r) = (-20 \log r - \beta \cdot r) \) – propagation loss, accounting for a spherical wave front divergence and acoustic signals kilometric attenuation; \( A'(r) = 10 \log A(r) \) – propagation anomaly, dB.

\[
Q(r) = N_0(r) - A'(r) = (-2 \log r - \beta \cdot r) - 10 \log A(r),
\]

where \( Q \) – signal energy loss in a media.

In respect to the applicability of the methods based on geometric optics approximations the following should be noted:

- methods do not provide for the field calculation at the points of complete internal reflection;
- methods are not applicable in vicinity of caustic and focal points;
- ray acoustic approximation is applicable when a relative sound speed change on a wavelength is small:

\[
\frac{\lambda}{c} \left| \frac{dc}{dz} \right| << 1
\]

The above is equivalent to the requirement that the sound speed gradient should be less than the frequency value:

\[
dc / dz < f
\]

- the ray theory does ensure a correct solution when the rays’ curvature radius or waves’ amplitude change significantly over a distance equal to the wavelength.

Fig. 2 gives an example of the acoustic field calculations based on ray methods.
Fig.2 Results of the source acoustic field calculations (ray method)

The computer modeling results analysis shows that efficiency of calculations using ray methods has a tendency to marginally increase with the medium frequency increase. It should be specially noted that at high frequencies the calculations’ efficiency significantly grows in comparison with the PDPE method efficiency.

4. IGIS AS A PLATFORM FOR SPM

Any hydroacoustic calculations are determined by the signal’s source and receiver coordinates tie. The above could be explained by the unique values of acoustic field parameters in each point of the field forming. At that the field uniqueness is determined by the complex laws of the medium properties influencing upon the signal propagation as well as by the high changeability of the medium properties in space. The hydroacoustic calculations, accounting for the above specifics, assume running a complex of computing operations over geospatial data in the given points of aquatic environment, located along the acoustic energy propagation path.

Geographic information systems (GIS) to the greatest extent meet the requirements of convenient access to geospatial data, at that, arranging for their processing in the interests of hydroacoustic calculations and the results output. GIS could be considered [3] as a certain platform for the system of hydroacoustic calculations that provides for the following tasks solving:
- input of initial modeling data including the acoustic energy source and receiver geographical tie to the given space zone;
- arrange for an access to medium parameters’ geospatial data bases that specify the acoustic field parameters;
- provide for a modeling control interface;
- output of modeling results in a user-friendly form.

Using GIS capacities as a platform of the hydroacoustic calculations system is a necessary though insufficient condition for a successful solving the system inherent problems. Moreover, additionally to GIS the being developed SHC should incorporate the constituents ensuring:
- common ontology based interaction between all components of the system;
- run calculations ensuring forming prediction estimations for the media specifying source (initial) data;
- implementing various acoustic field calculations methods and algorithms;
- intelligent support of calculations’ control that includes identification of the calculations’ initial conditions, selection of the rational method for acoustic field calculation, selection of the methods increasing the calculations’ efficiency and accuracy, selection of the output data approximation methods, etc.

5. INTELLIGENT SUBSYSTEM OF SPM

An expert system is used to solve the problem of intelligent support for calculations’ control. The above system is a rules oriented system aimed at knowledge stored in the database processing. The rules descriptions can also be stored in the database as a part of a subject area description.

The intelligent subsystem consists of two major parts; the first one is a scenarios system, the second one is an inference machine realizing the first-order predicates logic. By the scenarios’ system based on expert knowledge or premodeling the logic of the field calculation is formed. The inference machine interprets and realizes the modeling logic developed as a scenario. Depending on the initial data set and the results of the current modeling step the calculation technique for the next step is developed. The proposed approach supports solving two problems: the computation speed is decreasing due to omission of insignificant calculation points and the computation accuracy is increasing owing to correct use of mathematic algorithms. In addition to the stepwise optimization along the given path, the optimization of adjacent paths or spatial channels is ensured.

6. PARALLEL COMPUTING FOR SPM

The choice of parallel computing realization was stipulated by the following conditions:

- system of hydroacoustic calculations should support a concurrent availability for several users;
- realization of a paralleling to ensure its application effectiveness should provide for the time of one parallel process execution excess over the time spend for ensuring the exchange between two parallel processes.

In other words, the concurrent realization is expected for two or more processes that would be used by the third object-client in an arbitrary order.

This is why for the sake of the system’s performance enhancing it is desirable to have a mechanism allowing for:

- executing calculations asynchronously to other flows that use the result;
- asynchronous calculation notifying an object that uses it;
- encapsulating calculation in a way that its clients would not know whether the calculation is synchronous or asynchronous;

Based on the above requirements the cluster configuration of Application Server JBoss was implemented as a medium for parallel computations.

With due account for the mentioned conditions the paralleling mechanism in the system of hydroacoustic calculations is realized as follows:

- system of hydroacoustic calculations is realized in the technology of Enterprise JavaBeans (EJB) based applications server JBoss, having a cluster configuration;
• part of calculation (attenuation calculation for one direction) is executed as Session Stateless Bean, being a cluster shared resource;
• Session Stateless Bean asynchronous call, containing a calculation, is executed via the AS JBoss extension aimed at executing asynchronous calls Asynchronous proxy;

Thus, in the presence of a cluster system with the prevailing architecture (Massive Parallel Processing) and having distributed memory practically in its any realization the cluster version of application server JBoss with asynchronous calculations’ execution ensures a uniformed load on each cluster node and allows for performing hydroacoustic calculations in parallel with the total amount of flows being concurrently executed and uniformly distributed over a number of the cluster available nodes.

The given computational scheme is independent of a number of the cluster nodes and is capable of working as applied to a unit server. It also should be noted, that calculation along each direction is executed by the same software code, receiving different source data at the input, and this corresponds to the programming model Single Programm Multiple Datas. At that, time differences of calculation execution along each direction do not exceed 30%. So, the use of a multiprocessors cluster system correspondingly assumes that the time for calculating the whole detection zone would be commensurable with the time for calculating one direction at a standard PC.

7. IMMUNOCOMPUTING FOR SPM

The calculations of sound energy propagation losses in aquatic environment and based on these calculations’ results calculations of the detection zones for active and passive hydrolocation in any case require high computational power and, consequently, are time consuming. To solve the given problem when computational power is limited the acceptable calculation time in SHC could be met by implementing the immunocomputing method [8, 9].

For instance, at calculating the detection zone the arriving at the acceptable result can require a calculation step no less than one degree. Thus, to receive a circular detection zone it is necessary to calculate the losses at hydroacoustic energy propagation 360 times. Consequently, in case of computational power insufficiency the time of the above calculation can be unacceptable. The above situation possible resolution lies in performing calculations at the bigger step and accompanying the calculation by the received results interpolation for the omitted calculation steps, and SHC includes the realization of immunocomputing method as a particular interpolation technique. The choice of the given method is justified by the evidence of its accuracy and performance in comparison with the known least-squares method and artificial neural nets.

The usage of immunocomputing method in the SHC applications allows, based upon available statistic information, for predicting the medium parameters in any ocean point for any preset instant of time. Fig. 3 presents an example of predicting various parameters for different areas of the Black Sea.
Fig. 3 Examples of parameters generated by the predicting component for the Black Sea

a) Surface temperature (January 1); b) Surface temperature (October 17);

The initial data for the component operation are received from [7] and constitute monthly average HPP (sea water temperature, salinity and density) at different depths (0, 5, 10, 20, 30, 50, 75, 100, 125, 150, 200, 250, 300, 450, 500, 600, 800, 1000, 1200, 1500, 2000 meters) in the nodes of regular grid with one degree discontinuity over latitude and longitude.

The interpolation capacities consisting in immunocomputing method allow forming the components of hydroacoustic means’ detection zones on vertical as well as on horizontal plane in the points for which the results of acoustic field calculation do not exist. This in turn allows for increasing the hydroacoustic calculations accuracy.

8. CASE STUDY

Open code geoinformation system Open Map is used as a basis for the geoinformation interface. The user’s interface units provide for the intuitively understandable process of interaction between the system and the user at all calculations’ stages. The user, in particular, can choose for calculations any region of the World Ocean and easily via the given geographic coordinates or by a cursor plot the acoustic energy source, receiver locations, bottom topography (Fig.4) and a vertical sound distribution along the canal (Fig.5). At that, the acoustic energy propagation path is displayed as a segment, connecting the source and the receiver.
The calculations’ results are output by the geoinformation interface in a user-friendly form (Fig. 6-9).

When calculating the hydroacoustic means’ range it is possible to output a detector integral characteristic – its detection zone in the horizontal plane, constituting a space area, where the definite type object could be detected with a detection probability value no less than a given one (Fig. 9).

9. CONCLUSION

The paper presents the version of SHC developed based on intelligent geoinformation system; the considered SHC due to its modular construction, capacity for flexible change of calculations’ control algorithms, using open code software products allows for a
comparatively simple system upgrading and updating, at that, arriving at specified requirements to efficiency and results’ accuracy. The developed systems can be easily adapted to a definite sea or ocean area with due account for an analysis of the full-scale measurements data. The systems of the kind are expected to be widely applied to monitoring water medium of various functions.

The further development of the proposed system assumes through building the following subsystems:
- 3D representation of modeling results;
- solving the filed calculation problem in close to real-time scale;
- modeling the tasks of multistatic location;
- developing field calculation models for the systems adaptive to definite waveguide.

The paper authors are open to continue and initiate cooperative research activities with different organizations and interested individuals in solving the modeling problems stated by this paper.

REFERENCES


NEURAL CLASSIFIER OF SHIPS' HYDROACOUSTICS SIGNATURES

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Abstract: The paper presents the technique of artificial neural networks used as classifier of hydroacoustics signatures generated by moving ship. The main objectives of paper is to propose solution which allows to classify the objects basing on generated by them underwater noises. Firstly, the transmission of acoustic energy via the hull into water and origination of underwater noise in vibrations of ship's mechanisms was discussed. Basing on this considerations, it was shown how to create the hydroacoustics signature or so called "acoustic portrait" of moving ship. Next the classifier of acoustic signatures using artificial neural network was proposed. From the technique of artificial neural networks the Kohonen networks which belongs to group of self organizing networks were chosen to solve the research problem of classification. To check the correctness of classifier work there were made complex ships' measurements on Polish Navy Test and Evaluation Acoustic Range here hydroacoustics noises generated by moving ship were recorded. Basing on this measurements there were made researches in which the number of right classification of objects basing on hydroacoustics signatures was rated. Chosen results of research were presented on this paper. Described method actually is extended and its application is provided as assistant subsystem for hydrolocations systems of Polish Naval ships.
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Underwater acoustic modeling
STATISTICAL ANALYSIS OF ACOUSTICAL SCATTERING BY ROUGH SANDY MARINE SEDIMENTS

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Abstract: The problem of acoustical scattering by rough (rippled) sediments is important for many applications, such as buried objects detection, seafloor characterization, AUV navigation, etc. The presented numerical analysis concerns a general case of an arbitrary grazing angle that can be as low as 5°-10° and a realistic ratio between the height and the quasi-period of roughness. We use the Boundary Element Method (BEM) in 2D geometry to obtain the scattered pressure field in water and in sediment and compare these results to the well-known Helmholtz-Kirchhoff (HK) approximation. Using a Monte-Carlo technique, we evaluate the distribution of the pressure field together with its essential characteristics, such as average and standard deviation. A regime in which the average equals two standard deviations is found. Another observation is depth independence of the averaged penetrated field that appears below some minimum depth in lossless sediment, whereas this phenomenon is not observed for a single roughness realization. Interpretation of these results could help building up a theoretical description for penetration at low grazing angles and high frequency.

Keywords: acoustical scattering, roughness, sediment, boundary elements
1. INTRODUCTION

A problem of wave scattering by rough surfaces is widely addressed in literature [1-9]. There exist several classes of methods that enable to obtain an acoustic field distribution for a single roughness realization or to get a statistics of scattering. The simplest analytical approach considers the roughness height as a small parameter [1]; another important group of methods use the smallness of the roughness slope (the small slope approximation [2,3]). Alternatively, for high frequencies, when locally, at the scale of a ripple, the wave pattern is close to the plane-wave reflection, the Helmholtz-Kirchhoff (HK) method [1,4,9] is frequently used. Combinations [5] of these solutions are possible when roughness exhibits two characteristic scales: global waviness and small-scale ripples.

At the same time, especially for a long-range detection of objects when the grazing angles are small (subcritical), numerical approaches such as the Boundary Element Method (BEM) [4,6,9] are more reliable. In that case, the effects of shadowing and of multiple scattering are automatically taken into account. The integral equations of this method describe the mutual influence of different surface regions. The interaction between distant regions is much weaker than between neighboring ones that gives rise to a family of acceleration algorithms (see [7] and references herein).

In this paper, we apply the 2D BEM [4,6,9] to the problem of acoustic penetration from sediment with roughness having a Gaussian spectrum [4,8,9]. The sediment is modeled as a homogeneous viscous fluid. The developed code enables to calculate the acoustic pressure for the grazing angles as low as 5-10° and observe the huge gain in penetration due to roughness at high frequencies [8,9]. The comparison of the “exact” BEM solution to the HK approximation is performed for a permeable medium similarly to [5] where the validity of the HK approximation is discussed for the stress release boundary conditions. Statistical characteristics of pressure in sediment are calculated in the regime when the small slope and the HK approximations are both inapplicable.

2. EQUATIONS OF THE BOUNDARY ELEMENT METHOD

The geometry of the problem is illustrated in Fig. 1. The acoustical pressures \( p_w \) and \( p \) satisfying the Helmholtz equation in water and in sediment, respectively, are expressed via the following integral representation:

\[
p_w(x,y) = p_{inc}(x,y) + \int_{-\infty}^{\infty} dx' \sqrt{1 + \left(\frac{dx'}{dx}\right)^2} \left( \frac{\partial G_{xx}}{\partial n'} p_w(x',y') - \frac{\partial p_w(x',y')}{\partial n'} G_{xx} \right)_{y' = \xi(x')}
\]

\[
p(x,y) = \int_{-\infty}^{\infty} dx' \sqrt{1 + \left(\frac{dx'}{dx}\right)^2} \left( \frac{\partial G}{\partial n'} p(x',y') - \frac{\partial p(x',y')}{\partial n'} G \right)_{y' = \xi(x')}
\]

where \( G \) is Green’s function in 2D, \( G(x,y,x',y') = \frac{1}{i \omega} H_0^{(1)} \left( k \sqrt{(x-x')^2 + (y-y')^2} \right) \), in which \( H_0^{(1)} \) is the Hankel function of the first kind, and the complex wave number in the sediment \( k = (\alpha c)(1+ia/54.58) \), with \( a \), the attenuation coefficient of the sediment, expressed in dB/\( \lambda \).
Green’s function $G_w$ in water can be obtained by replacing $k$ by $k_w = \omega/c_w$. Here $\omega$ is frequency, $c$ and $c_w$ are the sound velocities of sediment and of water, respectively.

Equations (1) must be supplemented with the boundary conditions

$$
|_{y=\xi(x)} p_w(x,y) = |_{y=\xi(x)} \frac{1}{\rho_w} \hat{\partial} p_w(x,y) / \hat{n} = |_{y=\xi(x)} \frac{1}{\rho} \hat{\partial} p(x,y) / \hat{n}
$$

(2)

valid for a fluid-fluid interface of an arbitrary profile $\xi(x)$, with $\rho_w$ and $\rho$, densities of water and of sediment, respectively. Here $\hat{\partial}/\hat{n}$ denotes the derivative taken along the upward directed normal vector, $\partial / \partial n = \left( -\frac{\xi'}{\omega} \partial / \partial x - \frac{\xi'}{\omega} \partial / \partial y \right) \left( 1 + \left( \frac{x'}{w} \right)^2 \right)^{-1/2}$. The incident field $p_{\text{inc}}(x,y)$ can be specified as the Gaussian beam

$$
p_{\text{inc}}(x,y) = \frac{1}{\pi \sigma_w} k_w W \int_{-\pi/2}^{\pi/2} d\theta ' e^{-(k_{w}W/2)^2(\theta'-\theta)^2} e^{i(k_w(x-x_b)\sin \theta + (y-y_b)\cos \theta ')}
$$

(3)

represented as the plane-wave decomposition, with $w$ having the sense of the beam width. The incident angle $\theta$ is shown in Fig. 1.

![Image](image)

**Fig.1:** The 2D geometry of the problem.

The solution according to Eq. (1) can be calculated provided the surface values of the pressure and its normal derivative are known. They can be found from the same Eq. (1) by letting $(x,y)$ be at the surface $y=\xi(x)$ and adding the factor 1/2 in the left-hand side of Eq. (1). The system (1) is then discretized as in [6] and solved numerically.

### 3. SIMULATION RESULTS FOR A SINGLE REALISATION

The pressure fields for the “exact” BEM solution as well as the HK approximation are given in Fig. 2 in which two cases are illustrated: very smooth and weak roughness with the r.m.s. 3 mm and the correlation length $L=3.3$ m (at the left) and a realistic case with $L=33$ cm and the 1.5 cm r.m.s. (at the right). Other parameters for this simulation were: the wave frequency $\omega/2\pi=60$ kHz, the grazing angle $\alpha=20^\circ$, the beamwidth $w=0.6$ m at focus $x_b=y_b=0$, and the physical constants of sediment $c=1650$ m/s, $\rho=1920$ kg/cm$^3$ and of water $c_w=1515$ m/s, $\rho_w=1024$ kg/m$^3$. These two examples show that, at least at frequencies comparable to 60 kHz, the HK method is hardly usable for analysis of wave penetration at subcritical angles; much more “gently undulating” surfaces are required. In general, in the considered case the
conditions of applicability of the HK method were found to be stricter than for the fully reflective surface [4].

An apparent reason for this effect is that, in the subcritical regime that does not exist for the fully reflective surface, roughness greatly enhances the penetrated acoustic energy, which is in agreement with known experimental [8] and theoretical investigations [8,9]. At high frequencies, a local grazing angle at a particular section of the surface can exceed the nominal critical angle that results in a drastic increase in pressure just below that section. A slight mismatch in boundary conditions leads to a significant difference in the wave pattern created by these “incorrectly” radiating secondary sources. In addition, comparison of the exact surface values for the pressure and its normal derivative to their HK approximations showed that the mismatch in the derivative is much higher than in the pressure itself, while for the reflective surface this factor disappears. As a result, the HK approximation should be used with great caution for analysis of acoustical penetration for small grazing angles. In fact, even for the mean Rayleigh parameter $2kR_c \sin \alpha^*$ ($R_c$ is the curvature radius, $\alpha^*$ is the local grazing angle) comparable to 120, as in the second case of the realistic roughness, the resulting pressure field is completely different from the exact solution.

Other numerical experiments made it possible to find the following tendencies in the HK method accuracy. Firstly, the error in the pressure field in water is considerably smaller than in sediment, since the main contribution in water is provided by the precisely known incident field. Secondly, the considered high-frequency HK asymptote provides a better agreement far from the surface, similarly to the far-field approximation for diffraction at high frequencies. This means that the HK method is more suitable for analysis of reflection than of subcritical penetration. Finally, increase in the grazing angle improves the accuracy. This effect in our situation should be attributed rather to a weaker sensitivity to the boundary conditions mismatch than to shadowing which was found negligible in the discussed cases when the...
grazing angle higher than the mean slope of the profile. For higher slopes or lower grazing angles, shadowing effect would produce further distortions.

Note in conclusion that the case of high frequencies and low grazing angles, in which the effect of roughness on penetration is very high and can not be considered as a perturbation, apparently prohibits the use of the small slope approximation as well. In that way, the statistical results of the next section concern the situation when no existing analytical methods are applicable and the only relevant tool we have is the numerical simulation based on the integral equations.

4. STATISTICAL ANALYSIS OF ACOUSTICAL PENETRATION

The objective of this section is to provide statistical data on the penetrated and reflected acoustical fields and describe some “empiric” properties of the averaged curves. Here our interest is to the roughness-induced component of the solution i.e. the difference $\Delta p$ between the pressure in the presence and in the absence of roughness. In Fig. 3 the real part of the pressure excess $\Delta p$ is illustrated. It is seen that in water a deterministic reflected pressure component is still present (gray curve at the left oscillating with the spatial period of the incident field), while in the sediment this contribution does not appear. In the case when the incident field is close to a plane wave, for lossless sediment, the statistical properties of the penetrated field do not depend on depth below some level exceeding the roughness r.m.s. In particular, both real and imaginary parts of the field have the Gaussian distribution with the zero average and the standard deviation independent of depth.

![Fig. 3: The average and the standard deviation of the real part of the pressure difference $\Delta p$ at the depth -40 cm in water (left) and 40 cm in sediment (right).](image)

The statistical representation (Fig. 4) of the pressure excess amplitude shows again the independence of averaged (1000 realizations) $|\Delta p|$ on depth, while at the level of a single realization this property was not observed. Moreover, in sediment, we found an empiric rule $|\Delta p| = 2\sigma_{\Delta p}$ valid for the plane-wave excitation at depths exceeding the roughness r.m.s. Numerous numerical experiments confirm that the revealed features are quite universal in the regime of high frequency and small grazing angle where the penetrated field is dominated by the incoherent roughness-induced component.

Interpretation of these results is the subject of our future work. We hope that an explanation for the rule “the average equals two standard deviations” can help building up a theoretical description for weakly determined penetrated fields. Ideally, we aim for a procedure that would enable to calculate the statistical characteristics of the field directly
from the statistical characteristics of roughness together with other physical and geometrical parameters of the model.

![Graph showing pressure excess amplitude at different depths](image)

**Fig.4:** The average and the standard deviation of the pressure excess amplitude at the depths ±40 cm (black lines) and ±25 cm (gray lines).

5. ACKNOWLEDGEMENTS

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REFERENCES

SONAR PROCESSING PERFORMANCES IN RANDOM ENVIRONMENTS

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\textbf{Abstract:} We present a numerical model for stochastic propagation of acoustic waves in fluctuating marine environment. This model predicts the de-coherence effects of the propagated acoustic signals (space decorrelation, and time-distortions of transmitted waveform) which degrade the performances of sonar processing and induce further limitations of the detection capabilities of sonar systems.

We investigate the effects of random sound speed fluctuations on the performance of linear horizontal arrays from numerical runs of the stochastic propagation model.

As an illustration, we present quantified results for the performance degradation of conventional beamforming for spatial processing, associated to a towed linear array, in presence of realistic random fluctuations of the medium: waves in a deep ocean environment.

We observe trends in the behavior of the degradation of the array gain which are summarized within an approximate closed analytical formula. This formula will be helpful for quickly correcting the Array Gain term in the sonar equation and for evaluating in a more realistic way the detection range in a fluctuating oceanic environment.

\textbf{Keywords:} Sonar performance, Array gain, Random environment, Acoustic propagation
1. INTRODUCTION

Sonar performance assessment requires a model of acoustic propagation in the marine environment and a description of the sonar system. In fluctuating environments (particularly in coastal and littoral waters), de-coherence effects of the propagated acoustic signals (space decorrelation, and time-distortions of transmitted waveform) degrade the performances of sonar processing and thus induce further limitations of the detection capabilities of sonar systems.

We first present the main features of the acoustic modelling developed in order to give a realistic evaluation of sonar performances in harsh environments. This modelling, based on a stochastic approach for solving the wave equation, provides, on one hand, spatial and temporal statistical moments for the propagated acoustic field and, on another hand, realizations for the impulse response of the random medium. In the modelling we consider the stochastic fluctuations of sound speed in the water column due to internal waves and turbulence.

As an illustration, we present quantified results for the performance degradation of conventional beamforming for spatial processing, associated to a towed linear array, in presence of realistic random fluctuations of the medium: internal waves in a deep ocean environment.

2. AN EQUATION FOR THE CROSS-CORRELATION OF ACOUSTIC PRESSURE IN A RANDOM ENVIRONMENT

Our approach is an extension to 3D configurations of the 2D procedure of Wilson and Tappert ([3]): starting from the Standard Parabolic Equation for governing the propagation of sound through random fluctuations δc:

\[ 2i k_0 \frac{\partial \psi}{\partial x} + \frac{\partial^2 \psi}{\partial y^2} + \frac{\partial^2 \psi}{\partial z^2} + k_0^2 \left( n_0^2 - 1 - 2 \frac{\delta c}{c_0} \right) \psi = 0 \]

the so-called “delta-correlation approximation” (Chapter 4 in Tatarskii, [2]) allows to derive equations for the statistical moments of the acoustic pressure \( \psi \), like for instance its second moment (spatial correlation):

\[ \left\{ 2i k_0 \frac{\partial}{\partial x} + \frac{\partial^2}{\partial y \partial y} + \frac{\partial^2}{\partial z \partial \zeta} \right\} \left\{ \left( \frac{\psi(x,y,0,0,0,0)}{c_0^2} \right) * \left( \frac{\psi(x,y,0,0,0,0)}{c_0^2} \right) \right\} = 0 \]

(1)

where is involved the function \( \phi \), depending on the correlation function of the random sound speed fluctuations:

\[ \phi(\nu, \zeta) = \frac{4}{c_0} \int \left( \delta c \left( \frac{1}{2} x, \frac{1}{2} y, \frac{1}{2} z + \frac{1}{2} \zeta \right) \delta c \left( \frac{1}{2} x, \frac{1}{2} y, \frac{1}{2} z - \frac{1}{2} \zeta \right) \right) d\zeta \]

A Fourier transform along the vertical and departures \( \zeta \) and \( \nu \) gives a new unknown function: the mean Wigner transform \( \Gamma \) of acoustic pressure:

\[ \Gamma(x, y, z, \varphi, \theta) = \frac{k_0^2}{4\pi^2} \int \int \psi(x, y, z + \zeta + \nu, \varphi) \right\} \left( \psi(x, y, z + \zeta + \nu, \theta) \right) d\nu d\zeta \]

which may be understood as a mean angular density of squared pressure:

\[ \left\langle \psi(x, y, z) \right\rangle = \int \int \Gamma(x, y, z, \varphi, \theta) d\nu d\zeta \]
From (1), $\Gamma$ is solution to a diffusive transport equation:

$$
\frac{\partial \Gamma}{\partial t} + \nabla \cdot (\Gamma \nabla \phi) + \frac{1}{2} \frac{\partial \Gamma}{\partial z} \frac{\partial^2 \phi}{\partial z^2} \approx -\sigma \Gamma + \int \int W(\phi' - \phi, \theta' - \theta) \Gamma(x, y, z, \phi', \theta') \, d\phi' \, d\theta' \quad (2)
$$

Featuring transport terms (left member), i.e. conservation along paths, along with an attenuation term $-\sigma \Gamma$ involving the scattering cross-section $\sigma$, and an integral term describing the redistribution of the attenuated power flux:

$$
W(\phi' - \phi, \theta' - \theta) = \frac{k_0^4}{16 \pi^2} \int e^{ik_0(\phi,0,0) \phi(\theta,0,0) \zeta} d\phi d\zeta \quad \sigma = \frac{k_0^2}{4} \phi(0,0) = \int \int W(\phi, \theta) \, d\phi \, d\theta
$$

We have implemented a Monte-Carlo solution to equation (2), derived from Wilson and Tappert’s technique ([3]), and making use of the properties of the diffusive transport equation (see e.g. Dautray [1], Chap. 3): conservation of $\Gamma$ along paths (left term), which undergo random deviations adjusted so that the right member is reproduced when playing the procedure many times. Using this technique, we have developed the program DEMOCRITe (Diffusion Effects by a Monte-Carlo Resolution of an Intensity Transport Equation) and we evaluate numerically the cross-correlation of pressure $\langle \psi(x,y,z)\psi^*(x',y',z') \rangle$.

### 3. Degradation of Array Gain: Definition

We consider the acoustic pressures $m_n$ measured by the N transducers of a line array, which combine signals $\psi_n$ and random white noise $b_n$:

$$
\langle b_n \rangle = 0 \quad \langle |b_n|^2 \rangle = \sigma_n^2
$$

The Signal to Noise Ratio $\text{SNR}_0$ is:

$$
\text{SNR}_0 = 10 \log_{10} \frac{\langle |\psi_0|^2 \rangle}{\sigma_N^2}
$$

The output of a beamforming along the broadside direction combines contributions from the signal $\psi$ and from the noise: $B = \Sigma + B$ with $\Sigma = \sum_{n=1}^{N} \psi_n$ and $B = \sum_{n=1}^{N} b_n$

If the signal is perfectly correlated, the standard deviation of the signal’s contribution is $\Sigma^2 = N^2 \langle |\psi_0|^2 \rangle$; divided by the standard deviation of noise contribution $\langle |B|^2 \rangle = N \sigma_N^2$, it gives the Signal to Noise Ratio $\text{SNR}_1$ after processing:

$$
\text{SNR}_1 = \text{SNR}_0 + 10 \log_{10} N
$$

If decorrelation occurs along the array: $\langle \psi_n \psi_n^* \rangle = |\psi_0|^2 A_{mn}$, the Signal to Noise Ratio $\text{SNR}'_1$ after processing is different:

$$
\text{SNR}'_1 = \text{SNR}_0 + 10 \log_{10} \left( \frac{1}{N^2} \sum_{n} \sum_{n'} A_{nn'} \right)
$$

The array gains in both configurations are: $AG = \text{SNR}_1 - \text{SNR}_0$ and $AG' = \text{SNR}'_1 - \text{SNR}_0$

Their difference gives the degradation of the array gain due to decorrelation of sound:

$$
\delta AG = AG - AG' = -10 \log_{10} \left( \frac{1}{N^2} \sum_{n} \sum_{n'} A_{nn'} \right)
$$

Since $A_{nn'} \leq 1$, $\delta AG$ is always positive, and is 0 when the $A_{nn'}$’s are 1 (perfect correlation).

### 4. Close Formulas

We have run the program DEMOCRITe in a deep Mediterranean environment, where direct propagation from source to array always occurs: various configurations were investigated including different source-array ranges, parameters of random sound speed fluctuations and frequencies, as listed in the table A below. The array is otherwise considered as perfect (no distortion …) and motionless.
Table A

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>300, 1000, 5000</th>
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<tr>
<td>Source-array range, x (km)</td>
<td>1, 2, 3, 4, 5, 7.5, 10, 12.5, 15, 20, 30</td>
</tr>
<tr>
<td>Amplitude of Sound-Speed fluctuations, ( \delta c ) (m/s)</td>
<td>1, 0.5, 2</td>
</tr>
<tr>
<td>Horizontal correlation length of SSp fluct. ( L_x ) (km)</td>
<td>0.2, 0.1, 0.4</td>
</tr>
<tr>
<td>Vertical correlation length of SSp fluct. ( L_z ) (m)</td>
<td>10, 20</td>
</tr>
<tr>
<td>Correlation time of SSp fluct. ( L_T ) (min)</td>
<td>15</td>
</tr>
<tr>
<td>Source depth (m)</td>
<td>150</td>
</tr>
<tr>
<td>Array depth (m)</td>
<td>150</td>
</tr>
<tr>
<td>Sea State</td>
<td>2</td>
</tr>
<tr>
<td>bottom porosity</td>
<td>56 %</td>
</tr>
<tr>
<td>bearing angle ( \theta )</td>
<td>90°</td>
</tr>
</tbody>
</table>

**Decorrelation of acoustic field:** a linear regression in the log-log plane (see Figure 2) gives a power law for the correlation length \( L_{\text{correlation}} \) of the acoustic field, as a function of horizontal range and of the characteristic features of the random sound-speed fluctuations:

\[
\frac{L_{\text{correlation}}}{\lambda} = \frac{\delta c}{c} \left( \frac{x}{L_x} \right)^\beta = \alpha \quad \text{where} \quad \alpha = 0.061\ldots \quad \beta = 0.4849\ldots
\]  

A more complex regression gives the following expression:

\[
L_{\text{correlation}} \approx \frac{c^2 \sqrt{L_x (\text{km})}}{2.2 (2\pi)^{3/4} \frac{0.955}{(\text{kHz})} \delta c (\text{m/s}) x \times 0.025 \log_{10}(f) + 0.41}
\]  

**Degradation of array gain:** a regression gives the expression below for the degradation of the array gain as a function of the ratio array length / correlation length of acoustic field (see Figure 3).

\[
\delta AG \approx 5 \log_{10} \left(1 + \frac{2}{3} w^2\right) \quad \text{where} \quad w = \frac{L_{\text{array}}}{L_{\text{correlation}}}
\]

\[
\delta AG \approx \begin{cases} 
10\log_{10} \left( \frac{L_{\text{array}}}{L_{\text{correlation}}} \right)^{-4} & \text{if} \quad L_{\text{array}} \geq 5L_{\text{correlation}} \\
0.9 \frac{L_{\text{array}}}{L_{\text{correlation}}} - 0.6 & \text{if} \quad L_{\text{correlation}} \leq L_{\text{array}} \leq 5L_{\text{correlation}} \\
\frac{1}{3} \left( \frac{L_{\text{array}}}{L_{\text{correlation}}} \right)^{1.8} & \text{if} \quad L_{\text{array}} \leq L_{\text{correlation}}
\end{cases}
\]
Figure 2 - Horizontal correlation length of the acoustic field as a function of distance and of the characteristics of the sound speed fluctuations

Figure 3 – Degradation of the array gain as a function of the ratio array length / horizontal correlation length of acoustic field
5. CONCLUSION

We have investigated the effects of random sound speed fluctuations on the performance of linear horizontal arrays from numerical runs of a stochastic propagation model. We have identified trends in the behavior of the degradation of the array gain which are summarized within an approximate closed analytical formula. This formula will be helpful for quickly correcting the Array Gain term in the sonar equation and for evaluating in a more realistic way the detection range in a fluctuating oceanic environment. It may also be useful when designing an array by indicating an upper limit to the array gain (see Figure 4).

![Figure 4 – Array gain as a function of array length for various values of $L_{\text{correlation}}$ and of frequency](image)

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SOUND SCATTERING BY ANISOTROPIC ROUGH SURFACES WITH THE SMALL SLOPE APPROXIMATION

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Abstract: Ocean scattering can be driven by acoustic interactions with the seafloor. Far from being smooth and perfectly rigid, the ocean floor is a key component in the analysis of sound scattering. At frequencies of order of 10 kHz to hundreds of kHz, hence for those wavelengths, the roughness of a seabed is considered at different scales and this feature has to be taken into account in the analysis of the roughness sound scattering. The first-order Small Slope Approximation (SSA) is considered to predict the roughness scattering. The Small Slope Approximation is a unifying method able to reconcile the Small Perturbation Method and the Kirchhoff Approximation without needing to split the spectrum of the rough surface into large and small scales. Another important component is to consider the type of rough surface used to describe the ocean floor. The seafloor is usually chosen as isotropic and this choice gives the opportunity to simplify the implementation of the scattering model. Nevertheless the seafloor has been chosen to be anisotropic in order to cope with surfaces having special features. One of the simplest case of anisotropic rough surface has got a Gaussian distribution and different correlation lengths in the x and y directions. Results show that SSA can predict roughness scattering for this type of surface. Another case, close to a real situation, is a surface having the shape of sandy ripples, thus a surface with small hills and slopes in a particular direction. Scattering predictions are presented for different anisotropic surfaces without forgetting the major problem of the method. The slope of the rough surface should not incline steeply to avoid shadow effects. Thus, depending on the sound source and the receiver configurations, SSA is analysed with the maximum dimensions of the surface where SSA still works.

Keywords: surface scattering, roughness, bistatic, anisotropic surfaces,
1. INTRODUCTION

Despite the amount of work done to understand the scattering phenomena in the marine environment, there are still many delicate points to solve. Sound propagation is greatly influenced by the interactions with the seafloor. This influence can be a limiting factor for some sonar systems or for transmitting acoustic signals. It can also be a tool for examining the properties of the seafloor. There have been many efforts to understand the scattering process, first with the monostatic scattering [1] and then with the bi-static scattering [1,2]. The bi-static first-order small slope approximation [3,4] is studied, without being simplified as it could be the case for isotropic rough surfaces [1,5]. This scattering model is applied to anisotropic rough surfaces considering acoustic wavelengths of order of few millimeters (about 5 mm) to 15 cm. One should note that the scattering strength in one direction or another is completely different depending on the orientation of the roughness. One direction of the surface may be seen, by the acoustic wave, less rough than another, and so on. This situation requires that different types of rough surfaces to be taken into account, thus different kinds of height correlation function. Furthermore, the scattering process applied to anisotropic surfaces is limited by the smallness of the height slopes. Thus depending on the surfaces and on the directions of the scattering modelling, the predictions of the scattering strength are more or less limited by certain grazing angles. Besides the introduction, the theoretical background necessary to understand the scattering phenomenon by anisotropic rough surface is explained in Section 2. Simulations are built for analyzing and for clarifying the scattering problem of rough surfaces with particular features in Section 3.

2. THEORY

The geometry of the scattering modeling under study is depicted in Fig.1, where the incident sound wave vector and the scattered one are denoted as $k_i$ and $k_s$, respectively.

\[
\{k_i = \{K_i, -k_{zi}\}, \quad k_s = \{K_s, -k_{zs}\}\}
\]
where $\mathbf{K}_i$, $\mathbf{K}_s$ are the transverse components of the incident and scattered waves in the $(x,y)$-directions, so $\mathbf{K}_i = \{k_{xi}, k_{yi}\}$ and $\mathbf{K}_s = \{k_{xs}, k_{ys}\}$. The vertical components in the $z$-direction, $k_{zi}$ and $k_{zs}$, meet the relation $k_z = \sqrt{k_i^2 - (k_{xi}^2 + k_{yi}^2)}$, where $k$ is the wavenumber. The scattering theory depends on both the grazing angles ($\theta_i, \theta_s$) and the azimuth ones ($\phi_i, \phi_s$). The scattering process is also dependent upon the parameters of the media where sound scattering takes place. As here, medium 1 is water, its sound speed is considered as constant ($c_1 = 1500\text{m/s}$) and its mass density is equal to $\rho_1 = 1000\text{kg/m}^3$. Medium 2 is the sediment and is described by its sound speed $c_2$, its mass density $\rho_2$ and its loss parameter $\delta_2$.

### 2.1. Rough-interface modelling

Let us assume that the interface between the two fluid media is rough. This surface is considered as plane on average and is defined by a deviation, $h$, to the interface relative to its means plane $z=0$ depending on the position $\mathbf{r}=(x,y)$ on the $(x,y)$-plane. A way to represent the roughness of a surface is the relief structure function, $D(\mathbf{r})$, defined as $D(\mathbf{r}) = \langle [h(\mathbf{r}+\mathbf{r}_0)-h(\mathbf{r}_0)]^2 \rangle$ where $h(\mathbf{r})$ is the elevation at the position $\mathbf{r}=(x,y)$, and $h(\mathbf{r}_0)$ is the elevation at the origin [1]. Moreover $D(\mathbf{r})$ is also related to the covariance, (2).

$$D(\mathbf{r}) = 2(C(\mathbf{r}) - C(0,0))$$  \hspace{1cm} (2)$$

where $C(0,0)$ is the covariance at the origin and $C(\mathbf{r})$ is the one at the position $\mathbf{r}$.

---

Fig.2: Left, an anisotropic surface with rms height of 5cm, wavelength of 20cm and deviation angle from the y-axis:0°; Right, its corresponding height correlation function (top: in 3D; middle: x-direction and $y=0$; bottom: y-direction and $x=0$).

Fig.2 shows an anisotropic rough surface, similar to sandy ripples. This surface shows particular features with a specific orientation. Its dimensions are close to the acoustic wavelengths of interest. Its height correlation function depends on the dimensions of the ripple shape and on its orientation.
2.2. **Acoustic scattering modelling**

The scattering cross-section is evaluated by using the first-order Small Slope Approximation and is expressed as:

\[
    m = \frac{k^4 |A|^2}{4\pi(k_{zi}-k_{ni})^2} \int_{-\infty}^{\infty} e^{-\frac{1}{2}((k_{zi}-k_{ni})D(x))^2} - e^{-\frac{1}{2}((k_{zi}-k_{ni})D(\infty))^2} \right] e^{-iK \cdot r} d^2r
\]

(3)

where the different terms are already described in the previous sections, except \( A \) which is derived from the perturbations theory and depends on the parameters of the environment [1].

The integrand of (3) is integrated over a surface, thus in the \( x \) and \( y \)-directions. The convergence of this integral depends on the set of angles describing the incident and scattered waves, on the frequency of the emitted wave, on the rough surface (isotropic or anisotropic, the dimensions of the roughness, and so on) and varies a lot according to these parameters.

3. **SIMULATIONS**

Scattering strength is predicted with SSA for different kinds of anisotropic rough surfaces.

3.1. **Anisotropy with a Gaussian distribution**

A first example of scattering strength is a simulation considering a Gaussian height distribution to describe the roughness of the surface. One should note that such a height distribution with similar correlation lengths in the \( x \)- and \( y \)-directions corresponds to an isotropic surface, and to an anisotropic surface if the correlation lengths in the \( x \)- and \( y \)-directions are dissimilar. In one case, \( L_x = L_y = 20 \text{cm} \), where \( L_x \) and \( L_y \) are respectively the correlation lengths in the \( x \)- and \( y \)-directions. The rms height is about 5cm. In another case, \( L_x = 20 \text{cm} \) and \( L_y = 1 \text{m} \). The scattering strength is shown in Fig.3 as a function of the scattered angle \( \theta_s \), with \( \theta_i = 60^\circ \), a frequency of 30kHz, in the \((x,z)\)-plane and in the \((y,z)\)-plane, thus for \( \phi_i = \phi_s = 0^\circ \) and for \( \phi_i = \phi_s = 90^\circ \) respectively. The seafloor is made of medium sand \((c_2=1767 \text{m/s}, \ \rho_2 = 1845 \text{kg/m}^3, \ \delta_2 = 0.01624)\). For the isotropic case, the scattering strength is very similar whether the simulations are made in one plane or the other. In the case of the anisotropic surface, the scattering strength is higher in the specular direction for simulations in the \((y,z)\)-plane. The scattering strength is spreads in many directions when simulated in the \((x,z)\)-plane. In the \((y,z)\)-plane, the rough surface is flatter than the roughness in the other plane, that is why the scattering strength is more concentrated in the specular direction. The surface cannot be assumed isotropic if there are particular features. The reason is that for such an anisotropic rough surface, predictions are completely different from one configuration to another.
3.2. Anisotropy from sandy ripples

Another height correlation function is considered to describe the roughness and corresponds to a ripple shape as in Fig 2. The rms height is about 5cm, the wavelength of the ripples (distance between the crest-lines) is 20cm, the deviation angle is 0°. This means that the longitudinal part of the ripples is in the y-direction, whereas the sine shape is in the x-direction. The rough surface is assumed to vanish after 100m. The scattering strength is shown in Fig.4 as a function of $\theta_s$, with $\theta_s = 60^\circ$, a frequency of 30kHz, i.e. a wavelength of 5cm, and a seafloor made of medium sand (same sediment parameters as the ones in Section 3.1.). To compare, the anisotropic height Gaussian distribution is used with $L_x= 20cm$ and $L_y=100m$. On the left of Fig.4, the predictions are made in the (x,z)-plane, whereas on the right of Fig.4, the predictions are made in the (y,z)-plane.

Fig.4: Scattering strength as a function of $\theta_s$, considering the ripples correlation function and the anisotropic Gaussian distribution; Left, in the (x,z)-plane; Right, in the (y,z)-plane.

Fig 4. shows the scattering strength predicted by considering a ripple correlation function (line with circles) and by considering an anisotropic Gaussian distribution (line with squares). On the right of Fig.4, the energy is around the specular direction, whereas on the left of Fig.4,
the energy is spread in many directions. In the transversal direction of the ripples, i.e. in the \((x,z)\)-plane, SSA seems to show interferences when predicting the scattering strength. It may be related to the roughness shape. Between \(\theta_s = 20^\circ\) and \(\theta_s = 130^\circ\) on the left of Fig.4, the prediction with the ripple correlation function gives results which are in the average of the prediction with the anisotropic Gaussian distribution, despite the interferences. Below \(20^\circ\) and above around \(130^\circ\), errors may occur because of the limitation of SSA and of the rough surface considered in this simulation. The model is limited by the smallness of the roughness slope. From [6], the slope should be smaller than the sine of the grazing angle to avoid errors from shadow effect. Moreover, the limitation is extreme for the prediction in the \((x,z)\)-plane and becomes less important in all other directions since the slope is smaller in the other directions of this configuration. A small change of the dimensions of the ripples can considerably modify the limitation. This last point means that the predictions can still be correct at very small grazing angles depending on the configuration and on the environment under study.

4. CONCLUSIONS

The first small slope approximation can be used to predict scattering strength from an anisotropic rough surface. First, SSA shows expected results with the anisotropic Gaussian distribution. Then, interferences from the scattering from ripples appear with the general anisotropic scattering strength distribution. Nevertheless, the slope of the rough surface has to be taken into account and should be as smallest as possible to be able to predict scattering at the smallest grazing angles. Despite this limitation, SSA can cope with different dimensions of roughness and can be used with many anisotropic structure functions. A perspective would be to compare these predictions to real data to establish a relevant correlation between ripples and roughness scattering.

REFERENCES

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