OCEAN ACOUSTIC TOMOGRAPHY
Radiating Wideband Sonar Pulses with Resonant Sandwich Transducers by Designing the Driving Voltage Waveform

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A technique to radiate short length, high resolution, pulses with conventional piezoelectric transducers is described. It consists on designing the driving voltage waveform so that the radiated pulse has a zero-phase cosine-magnitude spectrum compatible with the natural frequency response of the transducer. According to Berkhout [1], zero-phase cosine-magnitude pulses have the minimum length, maximum resolution, within a prescribed frequency band. When applied to a 9 kHz sandwich transducer, this technique decreases the pulse length from 1 ms to 0.13 ms, increases the bandwidth from 1.4 kHz to 11.25 kHz, and lowers the Q factor from 6.2 to 1.23, at the cost of 33% of amplitude loss.

INTRODUCTION

An underwater transducer is driven usually by a tone-burst. However, Winter et al. [2] and Mazzola and Raff [3] showed that is possible to use Fourier techniques to find the electrical driving function so that the transducer radiates a prescribed acoustical waveform. Holly et al. [4] reported that a transducer driven with a shaped function responded in two octaves, with an amplitude loss of 15 dB.

Cobo [5] applied this technique to synthesize zero-phase cosine-magnitude, gaussian, and bionic pulses, with a conventional sandwich transducer. According to Berkhout [1], zero-phase cosine-magnitude waveforms provide minimum length, therefore maximum resolution, pulses within a prescribed frequency band.

METHODS

A transducer can be modelled as a linear system, with a transfer function $H(f)$, which can be measured. In conventional performance, the transducer is driven by an electrical input, and radiates an acoustical waveform. However, an acoustical waveform can be prescribed, $Y_e(f)$, and the corresponding electrical driving function, $X_e(f)$, can be accordingly synthesized

$$X_e(f) = Y_e(f) \cdot \frac{H^*(f)}{|H(f)|^2 + p^2}, \quad (2)$$

where $p^2$ is a regularisation constant, and * denotes conjugate complex. Therefore, the electrical function which must be synthesized is

$$x_e(t) = \mathcal{F}^{-1}\{X_e(f)\} = \mathcal{F}^{-1}\left\{Y_e(f) \cdot \frac{H^*(f)}{|H(f)|^2 + p^2}\right\} \quad (3)$$

where $\mathcal{F}^{-1}$ denotes inverse Fourier transform. Zero-phase cosine-magnitude pulses are given by

$$|Y_e(f)| = \begin{cases} \cos\left(\frac{\pi(f - f_0)}{B}\right) & f_1 \leq f \leq f_2 \\ 0 & f < f_1, f > f_2 \end{cases} \quad (4a)$$

and

$$\psi_e(f) = 0 \quad (4b)$$

where $|Y_e(f)|$ and $\psi_e(f)$ are the magnitude and phase spectra, respectively, $\gamma$ is a trade-off parameter between main and side-lobes on the shaped pulse [6], $(f_1, f_2)$ is the frequency band, $f_0 = (f_2 + f_1)/2$ is the central frequency, and $B = (f_2 - f_1)$ is the bandwidth.

Thus, the shaped waveform depends on four parameters:

- $(f_1, f_2)$, the lower and upper frequencies of the band.
- $\gamma$, the trade-off parameter.
- $p^2$, the regularisation parameter.

These parameters are chosen according with the mechanical frequency response of the transducer.
RESULTS

To illustrate the technique, let’s apply it to a transducer resonant at 9 kHz. The transducer is a Tonpiltz with 4 PZT-4A ring ceramics sandwiched between backing (steel) and matching (aluminium) layers. Figure 1 shows the acoustic waveforms radiated by this transducer when it is driven by tone-burst (4 cycles at 9 kHz) and synthesized electrical functions, as well as the corresponding envelopes. The synthesized function was designed to radiate zero-phase cosine-magnitude pulses with \( \{f_1, f_2, \gamma, \rho^2\} = \{6 \text{ kHz}, 20 \text{ kHz}, 0.25, 2.5\} \). Figure 2 shows the log-magnitude spectra of both pulses. Notice as this technique flattens (equalises) the response of the transducer within its natural frequency band. Table 1 summarises the characteristics of both pulses, in the time and frequency domains. The drastic reduction in pulse length (widening of the frequency band) involves, as a counterpart, an amplitude loss.

![Figure 1: Tone-burst and zero-phase cosine-magnitude pulses (a) and envelopes (b)](image)

**Table 1.** Summary of properties of the conventional and synthesized pulses

<table>
<thead>
<tr>
<th>Time</th>
<th>Frequency</th>
<th>Ampl. (V)</th>
<th>Length (ms)</th>
<th>B (kHz)</th>
<th>Q</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tone-burst</td>
<td></td>
<td>9.47</td>
<td>1.015</td>
<td>1.4</td>
<td>6.2</td>
</tr>
<tr>
<td>Zero-phase cosine-magnitude</td>
<td></td>
<td>6.29</td>
<td>0.135</td>
<td>11.25</td>
<td>1.23</td>
</tr>
</tbody>
</table>

![Figure 2: Log-magnitude spectra of the tone-burst and zero-phase cosine-magnitude pulses](image)

CONCLUSION

A drastic improvement of the vertical resolving power of acoustic pulses can be gained by driving the transducers with a more sophisticated electrical waveform.

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REFERENCES

Transverse current monitoring in a range dependent ocean

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Space-time scintillation analysis [1] offers a promising method of remote acoustic sensing of current in ocean. The evolution of the scintillation pattern at receiving array is related to the advection of the inhomogeneous medium through the sound beam, thus providing a basis for flow velocity measurement. In a range dependent ocean the sound signal structure changes due to variation of rays configuration. The source/environmental information smeared in mode space -- that means the information given by a single mode at receiving array is not presented by this mode. However, using the average characteristics of signal arrivals provides the opportunity to apply the signal scintillation technique for the transverse current measurement in complex environment.

\section*{ACOUSTIC SCINTILLATION METHOD}

Sound signal passing through the ocean fine structure is modulated, producing an irregular pattern of amplitude and travel time variations at the receivers. These variations evolve as the intervening medium changes. Under assumption that the fluctuations in the medium are produced mainly by the advection of a frozen fine structure field (Taylor's model of turbulence), evolution of the signal pattern can be directly linked to the motion of the medium. This allows determination of transverse component of the current by measurements of the time lag between the signal variations at two closely spaced receivers. The application of the array of the transivers allows to make the spatial filtering of acoustic scintillation and to get the spatial distribution of the current velocity [1,2].

In a range dependent ocean the sound signal structure changes due to variation of rays configuration. The source/environmental information is smeared in mode space -- that means the information given by a single mode at receiving array is not presented by this mode, but each mode made a contribution at each distance. Conventional space-time scintillation analysis is not applicable in this case. However, it is possible to find some stable characteristics of the sound signal such as collective time or cumulative sum of arrivals [3,4]. The collective time is in essence the mean arrivals; it indicates the total shift of the arrival pattern due to uniform temperature or current effect. The cumulative sum reflects the compression or dilatation of the arrivals pattern, it consists of defining the trend in the signal-arrival spectrum caused by the sound speed profile changes in the stratified ocean.

\section*{NUMERICAL MODELING}

\subsection*{Oceanographical data}

As an example of range-dependent ocean, the Fram Strait environment is considered in the present paper. Two major components of the Greenland and Norwegian Seas circulation determine the Fram Strait environment: the West Spitsbergen Current (WSC) and the East Greenland Current (EGC). The WSC consists of warm, saline water with a temperature of 2 – 5\textdegree C even in winter and a salinity of about 35.5\%, reflecting its origin far to the south, and occupies approximately the region between 3\textdegree 30\' E up to 9\textdegree 30\' E. In the region of the WSC, the shallow shelf extends from the Spitsbergen coast westward to about 8\textdegree 30\', where the depth increases sharply. The total volume transport of the WSC scaling is in a range of 3 - 10 and varies when measured during a period of several months. The model of the current velocity distribution in the strait is presented in Fig. 2 by the solid line. It is important to provide the permanent monitoring of the mass and heat flux through the Fram Strait that can be made by the application of acoustic methods. The temperature and salinity distributions in the cross section of the strait are rather complex, resulting in the complex structure of the sound speed profiles (Fig. 1). The
Spatial filtering of acoustic scintillation

The spatial filtering exploits an array of transmitters and receivers to focus on a particular wave number in the refractive index variability and a particular paths location. The sound propagation modeling was performed for the array consisted of four equally spaced transmitters and four receivers follow [2]. Corresponding to that, the network of 16 paths, which covers the area of $210 \times 15$ km$^2$ between the sources and the receivers, is considered. Inhomogeneities of the medium while they are flowing through the area covered by the network affect the travel time for the signals that are propagated along each of these paths. Travel-time fluctuations of the signals propagated along the different paths are retarded with respect to each other. The time lag is determined by the flow velocity distribution and the geometry of the system. The travel time variation is calculated for the sample of the current profile, presented in Fig. 2 by the solid line for the environment corresponding to that presented in Fig. 1. Summing up the travel time fluctuations for the signals passed along the different paths with appropriate time lags leads to the rise of the correlated fluctuations with respect to number of signal realizations $N$ (that is in 16 times), whereas the other fluctuations in process of random summation will be increased only in proportion of $N^{1/2}$ (or in 4 times in our case). Maximum travel time variation corresponds to the most probable value of the current velocity at the given location. The result of retrieval of the current profile in the Fram Strait is presented in Fig. 2 by shadow zones that indicate the probability of presence of the corresponding velocity in the location of zone.

CONCLUSION

The scintillation method of the transverse current retrieval in complex environment of the Fram Strait was modeled in the approximation of small aperture tomography scheme. The accuracy of the method is shown to be sufficient to resolve the features of the spatial distribution of the current profile.

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REFERENCES

A model based approach to acoustic subbottom profiling

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Acoustic classification of the subbottom is highly complicated due to the number of bottom properties that influence the acoustic response, especially when there are only a limited number of calibration measurements available. To review the possibilities of acoustic classification, the identifiability of a parametric model of the subbottom is investigated. To do so, the Fisher information matrix of the stochastic model for the measured acoustic response has been determined. The problem is identifiable when this matrix is non-singular. Its inverse, the Cramér Rao lower bound, gives the best possible precision of the estimated parameters. This bound thereby enables the design of appropriate signals for precise measurement of the subbottom. This has been done for a model of a surface layer with varying density as given by Lyons and Orsi [IEEE J. Oceanic Eng., 23, 411-422 (1998)]. Identification proved possible and is optimal when a signal is used with a frequency range of 2-60 kHz. [Work supported by TNO TPD and Rijkswaterstaat, The Netherlands.]

INTRODUCTION

In subbottom surveys, acoustic techniques are primarily used to determine subbottom geometry. The composition is usually determined by taking sediment cores. The costs of these surveys would be significantly reduced when remote, acoustic techniques could be used to classify the composition. Little is known about the theoretical possibilities of acoustic identification, even though several classification techniques have been developed [1]. This article, therefore, presents a general method for investigating the identifiability of parametric models of the subbottom using acoustic observations. To clarify, the method will be used to investigate the identifiability of a continuously varying density profile.

THEORY

A general result from signal processing is that a problem is unidentifiable when the associated Fisher information matrix $M$ is singular [2]. The derivation of this matrix requires a stochastic model of the observations. Under the assumption that the noise is additive, colored and has a Gaussian distribution, the following stochastic expression is obtained for the $N \times 1$ vector $\mathbf{s}$ that describes an observed trace:

$$\mathbf{s} = \mathbf{s}(\theta) + \mathbf{n}, \text{ with } \mathbf{n} = \mathcal{N}(\mathbf{0}, \mathbf{C}_n).$$

(1)

The $N \times 1$ vector $\mathbf{s}(\theta)$ is the parametric model of the observations with the parameters contained in the $K \times 1$ vector $\theta$. The $N \times 1$ vector $\mathbf{n}$ is the noise described by the $N \times N$ covariance matrix $\mathbf{C}_n$. Using Eqn.1, the following expression can be derived for the $(i,j)^{th}$ element of the Fisher information matrix:

$$M_{(i,j)} = \frac{\partial \mathbf{s}^\top(\theta)}{\partial \theta_i} \mathbf{C}_n^{-1} \frac{\partial \mathbf{s}(\theta)}{\partial \theta_j}.$$

(2)

Thus, all that is required is the derivative of the model $\mathbf{s}(\theta)$ with respect to the parameters $\theta_1, \ldots, \theta_K$ and the covariance matrix $\mathbf{C}_n$.

The Cramér Rao lower bound is the inverse of matrix $M$. This bound represents an upper limit to the precision of an unbiased estimator of the model parameters. In other words, the variance of an estimate of $\theta_k$ is always larger than or equal to the associated element on the diagonal of $M^{-1}$. This bound has its application in experimental design. For example, it can be used to find the optimal center frequency for the identification of a certain subbottom profile.

APPLICATION

A current problem in river maintenance is the estimation of the amount of silt before and after dredging. A potential source of errors in this estimation is the use of a discrete model for the silt layer, because a silt layer gradually changes from 'clean' water into silt. A more accurate model is that of a continuously varying density profile. Such a profile, and a description of the reflection of the profile, is given in by Lyons and Orsi [3]. With the sound speed assumed to be constant, the only parameters that have to be estimated are found in the description of the density profile:

$$\rho(z) = \begin{cases} \rho_w & z < z_l \\ \rho_w - \frac{\rho_w - \rho_0}{1 + \alpha (z - z_l)} & z \geq z_l \end{cases}.$$

(3)

This model consists of a jump in density at $z = z_l$ from water ($\rho_w$) to a starting density $\rho_0$, after which it increases to a final density $\rho_f$ under control of the parameter $\alpha$. To summarize, the vector of parameters $\theta$ is given by $(z_l, \rho_0, \rho_f, \alpha)\top$. The derivatives of $\mathbf{s}(\theta)$ with respect to these parameters can now be calculated and used to derive the Fisher information matrix according to Eqn.2.
RESULTS

The Fisher information matrix turns out to be nonsingular. Thus, the profile, described by Eqn.3, is identifiable. The goal now becomes finding the signal that minimizes the Cramér Rao lower bound. This search will be limited to the center frequency and bandwidth of a signal that is composed of a sine with a Gaussian envelope. To further reduce the complexity of the analysis, only the scale dependent parameter $\alpha$ will be investigated. The other parameters are set to: $z_l = 0 \, m$, $\rho_w = 1 \cdot 10^3 \, kg/m^3$ (known), $\rho_0 = 1.5 \cdot 10^3 \, kg/m^3$ and $\rho_s = 2 \cdot 10^3 \, kg/m^3$. The noise is assumed to be white. Sampling frequency is linked to the center frequency with a ratio of 8:1.

First, the optimal center frequency for the identification of $\alpha$ is determined. For this purpose, the relative precision of $\alpha$ at a $SNR$ of 0 dB has been plotted in Fig.1 as a function of $\alpha$ for a fixed bandwidth of 32. To make the graph independent of scale, the value of $\alpha$ is multiplied by the sound speed $c$ and divided by the center frequency $f_c$. Identification is optimal when $\alpha c / f_c = 6$. At this optimum, the relative precision in $\alpha$ is 236. Each increase in $SNR$ of 20 dB improves the relative precision tenfold. A usable precision requires a high $SNR$, eg. 80 dB results in a standard deviation of 2.36%. The range of realistic values for $\alpha$ is 8.9 to 224. The optimal center frequencies for the minimum an maximum of this range are, respectively, 2 kHz and 60 kHz. The optimum for the entire range of $\alpha$ is found at 10 kHz. At this frequency, the relative precision at the extremes of $\alpha$ is equal to 650. This is the worst relative precision over the entire realistic range of $\alpha$.

The top graph in Fig.2 shows the influence of bandwidth on the worst relative precision. It is a quantization of the well known effect that precision increases with bandwidth. The bottom graph in Fig.2 shows the influence of bandwidth on optimal center frequency. The optimal center frequency is approximately 10 kHz when the bandwidth is below 50%, above this value the optimal center frequency moves to 30 kHz. The optimum is a result of balancing the precision of all four parameters: a low frequency benefits the estimation of $\rho_s$, a high frequency benefits both $\rho_0$ and $z_l$. All four parameters influence each others precision. The balance is in favor of $\rho_s$. The optimal center frequency only moves up when the bandwidth is sufficiently great.

CONCLUSIONS

In this article the identifiability of parametric models for acoustic observations of the subbottom has been addressed. This has been approached with a method that is commonly used in signal processing: the Fisher information matrix. A general formulation has been presented for the calculation of this matrix.

The case of a continuously varying density profile has been investigated in this article. Identification proved possible. The center frequency that optimizes the identification as well as the influence of bandwidth have been determined. The requirements on the $SNR$, in order to obtain a sufficiently precise estimate, are quit demanding, but probably solvable. Both frequency sweeps and combinations of signals are promising options that will be investigated to see whether or not they provide a solution.

REFERENCES

Environmental-acoustic Impact on Optimum Sonar Search

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The Genetic Range-dependent Algorithm for Search Planning (GRASP) contains a genetic algorithm approach using Monte Carlo simulation and Bayesian detection statistics to optimize sonar search tracks in non-homogeneous environments. The optimization metric is maximum Cumulative Detection Probability (CDP) for a specified sonar (passive or active) against a target with specified characteristics (acoustic and tactical) during a fixed time period. GRASP solutions to simple and realistic ocean environments indicate that it produces intuitive search tracks that correlate highly with acoustic signal excess, as expected.

The essence of good search planning is to maximize the cumulative detection probability (CDP) within the search time available, but maximum possible CDP cannot be determined analytically except in special cases. In a homogeneous, isotropic environment, the standard Navy-doctrine ladder search path is near-optimal against stationary targets [1]. Further, maximizing CDP in real environments is an NP-hard problem, so the true optimum CDP achievable will never be known with certainty.

The Genetic Range-dependent Algorithm for Search Planning (GRASP) is an application of genetic algorithms which evolves search tracks to maximize some measure of effectiveness (MOE), typically CDP. Since the true optimum cannot be calculated, GRASP has been benchmarked against ladder patterns, where it often outperforms Navy doctrine by 20-50%. To assess the GRASP results, simulations in simplistic, non-homogeneous, anisotropic environments, where the “best answer” can be determined intuitively, were performed. If GRASP were to consistently attain CDPs close to those solutions, it would increase confidence in using the genetic approach in more complex and realistic environments.

Initial GRASP results for a sensor modelled with simple definite-range performance reproduced intuitive optimum solutions in a modest number of generations, proving its computational efficiency [2]. This second round of benchmarking is more realistic, assuming a range-dependent transmission loss in an environment with flattened coarse sand ridges 10-nm wide and 200m below the surface (high detection range), rising over a silty clay basin 1700m deep (low detection range). The ridges are laid out in various configurations: a single ridge, a double ridge, a square annulus, and a circular annulus. An example configuration with both intuitive and GRASP paths is shown in Figure 1, along with a CDP comparison.

FIGURE 1. The square annulus test case. The intuitive solution (dotted line) goes straight down the center-line of the ridge. The GRASP solution (dashed line) correlates well, but meanders and outperforms the intuitive solution at all search times.
When an intuitive solution is obvious, GRASP tracks do spatially correlate with intuitive solutions. They also usually out-perform intuition and standard tactics (in terms of CDP), in spite of the fact that GRASP paths are rarely the perfectly straight paths that intuition favors. This in turn improves intuition about what sorts of search paths are most efficient. In particular, sometimes the shortest path to success is not a straight line. Analysis of these and the previous benchmark trials reveals four competing principles, which are only strictly true for stationary targets and cookie-cutter detection functions.

1) **Straight lines are not uniquely optimal in a homogeneous, isotropic environment.** It can be easily shown that as long as the path’s radius of curvature is everywhere greater than the sweep radius, all non-redundant paths sweep the same area. Since CDP is generally proportional to sweep area, GRASP has no incentive to straighten the paths any further.

2) **Sharp turns, however, do introduce losses.** Sharp turns always introduce an area of overlapping sweep area. See Figure 2. This overlap maximizes (to 100%) for a U-turn, but soft changes in direction are low-cost.

3) **Anisotropy implies a preferred direction of travel.** In a homogeneous but anisotropic region (i.e., sweep width in a given direction is constant from point to point, but sweep widths in different directions vary at any given point), the preferred travel direction is perpendicular to the direction of maximum sweep width. See Figure 3. This obviously maximizes sweep area, and therefore CDP. This means a straight line is optimal in any such region if search time will be exhausted before a boundary is encountered.

4) **Inhomogeneities and boundaries often favor wiggling.** A single, straight search path is no longer possible if the preferred-direction dimension of the search region is small compared to the search time available, and may no longer be optimal if the search area is inhomogeneous. The optimal path will negotiate the best possible trade-off between the first three principles, which will usually involve some wiggling. Such will typically be the case on a ridge: detection range will be greatest when the searcher is on the ridge and looking along it, and so the preferred direction of travel will be across the ridge (and therefore quite short). Wiggling is the inevitable result, but wiggling with the most gentle serpentine motion possible. It is precisely such paths that GRASP tends toward as the number of generations increases, even in the simple environments studied during these benchmarking trials. In real environments, such cases are quite common, and this appears to be a major reason GRASP usually outperforms intuition.

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Degradation of reconstructed images in acoustical tomography

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Methods of acoustical diffraction tomography are examined for the reconstruction of objects with complex inhomogeneities. It is demonstrated that the inherent linearization errors from the Born approximation is unexpectedly less dominant than the limitations in real experiments including finite range of measuring angle and finite frequency band of transmitted pulse.

INTRODUCTION

In most of acoustical diffraction tomography methods [1-4] two kinds of error are usually established. First kind is caused by the linearization of inverse problem, which is a fundamental approximation in reconstruction algorithm. The second one is due to finite region of integration in the spatial frequency domain caused by non-ideal measurement conditions such as limited measuring angle (aperture) and finite frequency band of transmitted pulse. Linearization errors have been investigated by many authors and several nonlinear methods have been proposed. These methods are used to improve the quality of image reconstruction. However, the relative significance of the two errors should be compared a priori.

The non-ideal measurements can cause a significant degradation of reconstructed images in addition to the linearization errors. Moreover, the quality of reconstruction for the objects with complex inhomogeneities cannot be fully described with such parameters as resolution and point spread function [4].

In this paper we will focus on the first of these two effects and investigate it by means of computer simulation.

RESULTS OF COMPUTER SIMULATION AND CONCLUSIONS

Let us consider a small size transceiver T (Fig. 1) rotating around the origin of a polar coordinate system (r, φ). At each point (r, φ) (r = const) during scanning the transceiver emits a narrow pulse and then receives a scattered acoustical field. Inside the scanning area there is a finite size inhomogeneity Ω. It is assumed that the surrounding infinite medium outside Ω is homogeneous.

Function of inhomogeneities O(r) that will be reconstructed for the given measurement geometry is [2]

$$O(r) = \gamma_k(r) - \gamma_0(r),$$

where

$$\gamma_k(r) = \frac{\rho_0 c_0^2}{\rho(r)c(r)^2} - 1, \quad \gamma_0(r) = \frac{\rho(r) - \rho_0}{\rho(r)} - 1,$$

ρ(r), c(r) are density and sound velocity distributions inside Ω, outside Ω ρ(r) = ρ0 and c(r) = c0. Two models [2] of inhomogeneities have been investigated: a single scatterer and a local inhomogeneity embedded in a large one. The first model is an infinite liquid cylinder of radius r1 with density ρ1 and sound velocity c1. The second model is a system of two coaxial liquid cylinders of radii r1 and r2 < r1 with density ρ2 and sound velocity c2 if r ≤ r2, and ρ1, c1 if r2 < r < r1.
As it was shown [4] measurement data form a ring $k_r \in [2k_{\min}, 2k_{\max}]$ in the spatial frequency domain $(k_r, k_q)$ corresponds to the frequency band of the transceiver $T$ with borders $k_{\min}$ and $k_{\max}$. To compare errors caused by the Born approximation with those corresponding to the lack of measurement data inside a ring $k_r < 2k_{\min}$ a reconstruction of $O(r)$ was performed by two ways: using the Born approximation for $k_r \in [0, 2k_{\max}]$ and using exact solution for $k_r \in [2k_{\min}, 2k_{\max}]$. Some of these results are presented in Fig. 2-3 (reconstruction of $O(r)$ ) and Fig. 4-5 (mean-square errors in reconstruction of $O(r)$ ).

One can see that the lack of measurement data causes much stronger errors than the Born approximation. It should also be noted that only edges of the inhomogeneities are reconstructed for exact solution. Finally, within a region where the Born approximation is still valid it generates small errors compare to those for the lack of measurement data. Therefore, in this region the Born approximation can be considered to be exact and nonlinear algorithms of acoustical diffraction tomography are not as much as useful. With further increases of $\gamma_k$ and $\rho_{\gamma}$ the Born approximation becomes invalid. However nonlinear algorithms do not sufficiently improve quality of reconstruction even in this region. Using these methods one can only reconstruct the edges of inhomogeneities and it is difficult to obtain any information about inhomogeneities inside these edges.

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Understanding the Structures of Ocean Sound; Lessons From Experiments And Modeled Analyses Since 1951

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Following discovery of the deep sound channel, a program of intensive experiments at sea was begun in 1951 by the Bell Laboratories to explore and define the behavior of low frequency signal propagation and its corresponding sea noise spectrum [1]. The sea measurements were paralleled by development of computer codes to analyze and deconstruct the data and eventually to predict propagation loss and arrival structure in realistic, range varying ways. Modeling verified by measurement gradually evolved into a set of rules for maximizing coherent, high level signal reception over long ranges. These guidelines are outlined in this paper for both sound channel and positive gradient ocean acoustic environments.

BACKGROUND

By the end of the 1940’s naval experience in World War II and early progress in oceanography and undersea physics had brought about the recognition of low frequency acoustics as a means of passive surveillance. Experimental field stations for deepwater research to explore this potential were in place at Bermuda and in the Bahamas in 1951. Both locations evolved into a complex of bottomed and off-bottom suspensions of hydrophones and arrays cable-connected to shore, enabling continuous undersea observation. An intensive program of experiments was begun, supplemented by ship-based surveys over wide-ranging regions from 10 to 70 degrees N.Lat., testing differences in acoustic environments at those locations. The results presented here are concerned with propagation over long range in deep water, and the identification of its salient features from those measurements and from modeled reconstructions.

MEASUREMENTS AND MODELS

Omnidirectional hydrophones were used for all measurements, both shore facility and ship-deployed. Their various effective bandwidths were from 3-5Hz to 400Hz and from 2Hz to 2kHz. Most test hydrophones were bottomed at 220fms on slopes down to basin depths at 3100fms; there were suspensions off-bottom and also at the sound channel axis. Impulse sources (SUS charges) were detonated at 18m and 240m, and continuous wave (CW) transducers towed at depths from 18m to 120m. The CW test frequencies varied from 10-12Hz to 385Hz. The impulse sources were used to measure time-separated arrival order sequence and to identify approach angles at the receivers by model matching. The CW data was narrowband processed to measure and map distinctive features of propagation over range.

Ray diagnostic codes derived from computing ray trajectories in range, depth and time are powerful analysis tools for dissection and interpreting signal arrivals[2]. They have been used for that purpose in all data described here to identify the causes of the special effects which are of value in selecting observation sites of superior quality.

RESULTS AND SUMMARY

COMMENT

Forty years of testing at sea on over 200 field experiments, and ongoing assessment and model matching ashore, resulted in a set of guidelines for maximizing coherent, high-level signal reception over long range. They hold for all deepwater environments for sources at depths down to 240m and water depths of 2000fms or more and are listed in Table 1. Corresponding experiments which verified their validity are given in Table 2.

The unsurprising results outlined in Table 1 support all of the concepts of underwater sound propagation which have been developing since the 1950’s. Their value is that they have been repeatedly demonstrated by extensive and careful measurements at sea. Because these concepts can be so simply described graphically, the figures used for presentation are available on request.

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Table 1. Factors Conditioning High-Level Signal Reception Over Long Range

**Sound Channel**

(a) Optimal depth for a receiver is at conjugate (reciprocal) depth, where sound speed again equals sound speed at the source. All signal paths and their associated caustics radiated by the source are periodically recovered and the regularity of their arrival sequence assures near-continuous mapping of high-level waterborne signal reception over range. Maximizing reception of waterborne paths automatically yields the best coherence a particular undersea environment will provide.

(b) The 'source channel' defined by source and conjugate depths lies within the deeper sound channel and is thereby less vulnerable to disruption by intervening bathymetric features.

(c) As the vertical dimension of the sound channel shrinks or expands with changes in cooler or warmer oceanographic regimes, so does the source channel. A downchanneling (or upchanneling) of high energy signal paths results which can be exploited in choosing the depth and position of a remote receiver.

(d) A receiver on a downslope (or a downslope beneath a remote source) will capture (or launch) waterborne signal paths after one or two reflections. Slope enhancement is a major factor in recovery of high-energy signals from very long range.

(e) Surface image interference among refracting RSR rays recurs throughout long range signal reception and can periodically create a deep notch in the signal pattern.

(f) Reciprocity holds, as do the classic bottom interface parameters of critical angle and angle of intromission. Loss-per-reflection on most deep sea floors is 0.1 dB.

(g) Caustics can form within unconsolidated sediments and signals emerge to resume travel in the water. Efficient ray path travel through sediment layers has been observed often enough to be of significance.

(h) The effect of surface scattering on long range propagation in the open sea is minimal. Direction of swell orientation may have a moderate effect on signal quality in the form of modulating its fluctuation rate.

(i) At the low frequencies useful for long range sound propagation volume loss is negligible. Frequency spreading is also not significant; measurements at around 300Hz showed spreading to be somewhat less than 1%.

**Positive Gradient 'half channel' (with no ice cover)**

(j) Receiver and source at same depth is optimal. Strong caustics are formed but do not compensate for loss of waterborne paths as the receiver goes deeper.

(k) As bottom depth shallows, matching source and receiver depths becomes less critical. At 800fms the ocean waveguide is almost insonified.

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Table 2. Confirming Experiments

<table>
<thead>
<tr>
<th>What:</th>
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<td>(a) and (b)</td>
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<td>(b) and (c)</td>
<td>1955-1960</td>
<td>Antilles Arc – Mid Atlantic Ridge</td>
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<td>(c)</td>
<td>1983</td>
<td>Norwegian Sea</td>
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<tr>
<td>(d)</td>
<td>1951-1980s</td>
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<td></td>
<td>1952-1984</td>
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<td>(e)</td>
<td>1952-1984</td>
<td>North Atlantic and Pacific surveys</td>
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<td>Bermuda – Puerto Rico</td>
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<td>(g)</td>
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<td></td>
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<td>(h) and (i)</td>
<td>1958-1970s</td>
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<td>(j) and (k)</td>
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<td></td>
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REFERENCES


Underwater Acoustical Measurement of Wind and Rainfall in the Bahamas during Hurricane Irene

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Abstract

Underwater acoustic measurements of wind and rainfall were recorded during passage of Hurricane Irene through the Bahamas using an array of bottom-mounted hydrophones at a depth of 1.5 km. Sound spectrum levels were measured in the range extending from a few tenths of a kilohertz to approximately 50 kHz. Rainfall spectra associated with rain bands in the hurricane were observed to display the same acoustic/drop-size cycle response as reported in the literature \cite{2} for subtropical mesoscale convective systems in coastal ocean regions. Observations from the hydrophone array provided high resolution time series data of both wind and rain fields from which information on the life cycles of wind-rain systems and on the extent of their formation was obtained.

INTRODUCTION

Background

Of the many natural sources of ambient noise in the open ocean, wind and rain dominate the underwater sound field in the region 1 and 50 kHz. Wind sound is generated by agitation of the surface water from wave action and from the oscillation of trapped air bubbles within the surface layer during wave breaking. Rain sound on the other hand is produced by the impact of rain drops on the water surface and by the oscillation of trapped air bubbles sometime formed beneath the drops during impact \cite{1}

Recently, an extensive study of the acoustic method of rainfall detection, monitoring, and classification was conducted at three acoustically distinct sites on the U.S. Atlantic coast. This study, supported by non-acoustic meteorological and radar measurements established the existence of distinctly different spectral signatures for convective and stratiform rainfall \cite{2}. To determine if these proven coastal-water monitoring techniques apply equally well at sea, the authors undertook an investigation at a deep ocean site in the subtropics. The area, located in the Tongue of the Ocean, is part of the U.S. Navy Undersea Test and Evaluation Center (AUTEC) on Andros Island, Bahamas.

Acoustic data were recorded from a 12-hydrophone array secured to the ocean floor at a depth of 1.5 km. The array monitored an area approximately 425 km$^2$ at the ocean surface. Acoustic data were transmitted by fiber optic cable to shore where each of the twelve hydrophones (A through L) were sampled once every 12 seconds and recorded as FFT spectra.

A central question in going from shallow to deep water was whether the spectral shapes for convective and stratiform rainfall were preserved when monitored at depths of a kilometer or more.

INVESTIGATION

Case Study - Hurricane Irene

Of the many rain events recorded at the site one series associated with hurricane Irene was of special interest. As this tropical cyclone passed through the Caribbean Sea, associated rain bands produced three intense periods of precipitation over the array during a 24 hour period on 14-15 October 1999 (Figure 1).

Analysis of the acoustic spectra revealed the following: 1) acoustic signatures of rain recorded at 1.5 km depth contain the same basic features as those recorded in shallow water as cited in \cite{2}. See Figure 2 for sample ambient acoustic spectra from rain event I.
2) Acoustic wind spectra following a rain event can have significantly lower sound levels than wind spectra present prior to the onset of rain as in event I (Figures 1 and 2). 3) The speed of translation of a rain system over the array can be measured based on the difference in arrival times of the leading edge of the system as it moves between two widely separated hydrophones.

Figure 3. 4 kHz time series for hydrophones A and F for rain event 1 showing the difference in arrival times (21 min) of the leading edge of the rain system. The separation between A and F (along the wind direction) suggests a speed of translation of 13 m/sec.

4) The cross-correlations between the time series of hydrophones A and F for 4 and 15 kHz for rain event I are 0.93 and 0.89 respectively, indicating that the rain system not only preserved its main features but translated approximately uniformly over the array. See Figure 4.

Figure 4. Cross-correlations of the 4 and 15 kHz time series for hydrophones A and F for rain event I.

The acoustic representation of the life cycle of rainfall, as in event I from hydrophone A, i.e., pre-rain wind, convective rain, stratiform rain, and equilibrium post-rain wind form a closed curve as shown in the scatter plots of Figure 5. This description is consistent with present meteorological understanding [3]. While events II and III displayed typical acoustic spectra of convective rainfall, they contained no stratiform rain signatures. This suggests that these events were produced by immature systems in the early stages of development where only growing convective cells are present within the precipitating area [3]. It is believed that the systems producing events II and III moved beyond the array before the convective cells weakened sufficiently to form a region of trailing stratiform precipitation typical of a mature system.

Figure 5. Life cycles of rain events I and II.

During periods of no rain, on 14-15 October, averaged acoustic wind speed estimates for the array (using average deep-ocean working curves from [1]) were compared with 10-min vector averaged winds from an AUTEC offshore weather station, approximately 30 km from the array. The results are shown in Figure 6.

Figure 6. Anemometer and acoustically–derived wind speed time series corresponding to Figure 1.

Conclusions

Underwater acoustic methods can provide high resolution, large scale, long-term time series of both wind and rain fields associated with mesoscale convective systems. Information on the stage of maturity of individual wind-rain systems is obtainable based on acoustical observations of convective rain, the extent of formation of stratiform rain and of trailing wind fields.

REFERENCES

An Experiment in Underwater Acoustic Transmission at Suruga-Bay Using 4-DPSK and 16-QAM


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A basic experiment for allowing acoustic communication in the sea was conducted in an actual sea area using acoustic signals modulated by 4-DPSK and 16-QAM. The experiment, using vertical directional communication, was carried out at Suruga Bay (Japan). The transmission distance was approximately 900m. Encouraging results were obtained during experiment, using different bandwidths of 4kHz and 8kHz. In 4-DPSK, the result was obtained with 4kHz bandwidth. In 16-QAM method, the demodulation used both 4kHz and 8kHz bandwidths.

INTRODUCTION

Recently, the demand for underwater acoustic communication has risen. It is used for data collection from observation equipments, which are deployed in the sea, or for communication between underwater unmanned un-tethered vehicles and surface ships. This paper describes the results of a sea trial using the acoustic data transmission from 900 m depth.

DEMODULATOR

4-differential phase shift keying (4-DPSK) and 16-quadrature amplitude modulation (16QAM) were used for modulation methods. The carrier frequency ($f_c$) was 20 kHz, a transmission bandwidth (BW) was 4 kHz, and sampling frequency is 200 kHz. Accordingly the transmission rate is 8 kbps for the 4-DPSK and 16 kbps for the 16-QAM. For the 16-QAM modulation, 32 kbps data was obtained also (BW=8kHz). The block diagram of the demodulator is shown in Fig. 1. This demodulator is constructed with software. A Least Mean Square (LMS) algorithm was used in the adaptive equalizer. A carrier tracking unit was applied following the adaptive equalizer. In this experiment, demodulation is carried out by post processing after the experiment.

FIGURE 1. Demodulator block diagram.

FIGURE 2. Sea trial arrangement.
EXPERIMENT

The sea trial was carried out at a water depth of approximately 1,000 m in Suruga Bay in December 2000. Fig. 2 shows the arrangement of the sea trial. The transmitter was suspended from R/V "KAIYO". The hydrophone of the ship’s Acoustic Navigation System (ANS) was used for receiving signal. Fig. 3 and 4 are examples of the results of the demodulation, process for 4-DPSK and 16-QAM respectively. Figures b) and c) of Fig.3 and Fig.4 relate to the observation of points of Fig.1 indicated b) and c) respectively. These figures show that the carrier tracking unit works well to compensate for the phase error introduced by the heaving of the ship. It was blowing a gale. The wind speed was 19 m/s and wave height was 1.8m. The amounts of data for each transmission were 8,000 symbols for 4-DPSK and 4,000 symbols for 16-QAM. Symbol error rates (SER) to signal-to-interference- noise ratio (SINR) are shown in Fig. 5. In this figure, cases of error free are plotted as $10^{-4}$. In this sea trial, SINR were extremely high for 4-DPSK, and there were no cases of symbol error. For the 32kbps 16-QAM modulation, to SNR performance is down about 3-4 dB to the theoretical performance. It is thought that this is the performance deterioration by the receiver.

SUMMARY

The possibility of 32kbps transmission was confirmed during real sea trials. This result will be applied to design of the communication equipment of JAMSTEC’s autonomous underwater vehicle, “URASHIMA”.

FIGURE 3. 4-DPSK example signal. (BW=4kHz, SINR=22.4dB, output SNR=20.2dB, SER=0/8000)

FIGURE 4. 16-QAM example signal. (BW=8kHz, SINR=22.1dB, output SNR=18.1dB, SER=1/4000)

FIGURE 5. Symbol error rate vs. SINR.
Acoustic Tomography Experiment in the Lake of Geneva: Intermediate Results

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Based on ocean acoustic tomography principles, an acoustic experiment was performed in the Lake of Geneva so as to define future developments for limnological phenomena observations. The fluctuations in acoustic propagation between bottom fixed broadband sound source and receiver at ninety meter depth under the sound channel axis at thirteen kilometer range are examined. One of the purposes of theses transmissions is to study multipath propagation during the winter stratification in the major part of the lake. Measurements of seventeen days at pulse rate of twelve per hour, controlled by GPS clocks, was carried out. The analysis of travel time fluctuations, derived from classical cross correlation method, shows the high variability of the propagation path induced by internal waves and other currents within the field of study.

LIMNOLOGY OF THE LAKE OF GENEVA

Physical limnology in the Lake of Geneva tries to identify the dominant processes of transport and mixing which influence the water quality of the lake. In order to achieve this, a nearly continuous observation of the movements of the water masses and the development of the thermal stratification is required. Studies of these phenomena by acoustic means will consist in deploying electroacoustic systems of increasing complexity around the central part of the lake with the aim of characterizing the seasonal variations of the thermal stratification and detecting and quantifying the dynamics of different components of the current field present in the volume of water.

During the summer, stratification is strong with a shallow thermocline whereas it is weak with a deep thermocline during winter. Internal seiches with shore hugging Kelvin waves and offshore Poincaré waves dominate the basinwide water movement (Fig. 1). Short progressive internal waves in lakes has been documented for the first time in Lake Geneva. In the near shore regions, these waves are omnipresent throughout the stratified period [1]. In the offshore region, these waves often break, opening up temporary mixing windows between the epilimnion and the hypolimnion [2].

ACOUSTIC EXPERIMENTATION

We developed a low cost autonomous measurement system based on the monitoring of two computers synchronised by GPS clocks governing the emission and the reception of the signal inside the lake. Sound emitter and receiver, respectively in the nearshore of Lausanne and Amphion, are connected to permanent shore stations by reinforced cable of up to a 1000 m length. The acoustic transmission between a broadband source projector and a single omnidirectional hydrophone, both fixed at 2 m above the bottom of the lake at a 90 m depth under the sound channel axis and at a distance of 13 km, was first examined. For this experiment, we used a sweep 1-10 kHz of duration 500ms. Measurements over seventeen days at a pulse rate of twelve per hour were carried out between the 16th January 2001 and the 2nd February 2001.

The analysis of the travel time perturbations, derived from a classical cross correlation method shows the high heterogeneity of the thermally stratified propagation medium (Fig. 2). As a whole, figure 2 shows distinctive traces representing stable, resolvable arrivals. From this figure, a different evolution of the different resolvable ray paths in time and in amplitude can be observed, describing the temporal evolution of different water masses. To validate the preliminary measured results, we used a ray tracing software package. The simulated results are computed from bathythermic profiles measured during the acoustic experiments (Fig. 3).
The observed coherence in time between the measured results and simulated validates the method. We found that only refracted - refracted (heavy lines) and refracted - bottom reflected rays (light lines) are detected. The classification of such rays is not possible since different paths present same propagation time. This is a limitation of a single omnidirectional receiver.

During this period, an Acoustic Doppler Velocity Profiler (ADVP) was placed in the vicinity of the receiving station at about the same water depth. A first analysis of the combined data sets obtained during these measurements indicates already the following: the passage of a Kelvin wave during the last two days of the recording period is documented in the current data and the temperature data. This leads to a strong tilt of the thermocline on the south shore where the receiver is placed. During this same period, the travel times of acoustic rays are greatly reduced. Throughout the recording period, correlated oscillations with a period of 16 to 17 h are observed in the acoustic as well as in the current data sets. This period is close to the inertial period for Lake Geneva. Previous work has indicated that this is the period of Poincaré waves for the observed stratification.

CONCLUSIONS

It has been shown that repeated transmissions of the signal over an extended period of time can yield long-term high resolution time series of the evolution of the multipath propagation medium.

In addition, we have found in both the acoustic and the hydrodynamic data, indications for basinwide phenomena (Kelvin waves and Poincaré waves) as well as medium scale process (progressive internal waves; see [3]) which strongly correlate between the two types of observations.

REFERENCES

Influence of underwater climate on acoustic wave propagation in the Southern Baltic

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The variety of potential applications of devices for underwater observation and communication involves a visible increase in interest in their range. Reaching a defined point with a ray of acoustical wave depends significantly on spatial distribution of the speed of the acoustical wave, which is determined by hydrological conditions. The hydrological conditions in the Baltic Sea change seasonally and are influenced by the phenomena occurring at the sea – atmosphere boundary layer, as well as by inflows of highly saline water from the Northern Sea. The paper contains results of investigation of impact of seasonal changes in conditions of elastic wave propagation. Linear and nonlinear propagation is considered. Typical seasonal distributions of sound speed as well as nonlinearity parameter B/A were determined for the Southern Baltic. A research was also conducted to find the answer to the question to what extent the synoptic patterns of the sound speed and nonlinearity parameter B/A distribution differ from the averaged respective data. To get the answer several synoptic distributions established in different seasons are compared with the averaged ones.

\textbf{SEASONAL CHANGES IN ACOUSTICAL CONDITIONS}

The Baltic is a nontypical kind of the sea. It is a shallow sea of the lowest salinity in the world. Moreover, the conditions of acoustic wave propagation are much more complex than in other shallow waters. In typical shallow water appear seasonal changes in acoustical conditions in the upper layer of the depth of about 60-70 m caused by the changes in annual meteorological conditions. In deep water layer most often acoustical conditions are stable throughout the whole year. However acoustical conditions in the deep-water layer in the Southern Baltic change during the year. They depend on inflows of highly saline water from the Northern Sea through the Danish Straits, which evoke dense bottom current increasing the salinity at the bottom. The difference in vertical sound speed distribution in the Southern Baltic and the other shallow water could be noticed when we compare the data presented in Fig. 1.

\textbf{FINITE AMPLITUDE WAVE PROPAGATION}

Nonlinear properties of the sea are characterised by the nonlinearity parameter B/A, however the nonlinear distortion is influenced also by absorption. The quantities characterising the phenomenon of nonlinear propagation are contained in the following expression

$$\varepsilon \text{Re}_a = \left(\frac{B}{2A} + 1\right) \frac{\rho_0 c_0^2 v_0}{b\omega}$$

(1)

where $\varepsilon$ is the nonlinear coefficient, $\text{Re}_a$ – Reynolds number.
number, $B/A$ – nonlinearity parameter, $\rho_o$ - density, $c_o$ – sound speed, $\nu_o$ - velocity, $b$ – absorption parameter, $\omega$ - angular frequency. Temperature of water is the factor having the strongest impact on the nonlinearity parameter $B/A$. Therefore the spatial distribution of the parameter $B/A$ in the Southern Baltic is similar to the temperature distribution and the sound speed distribution. The absorption coefficient depends more than the nonlinearity parameter on salinity of the sea water, and changes with the frequency. That is why, the spatial distribution of the value $\varepsilon \text{Re}_a$ differs significantly from the spatial distribution of the sound speed and the parameter $B/A$, as it is shown in Fig. 2.

**ACKNOWLEDGEMENTS**

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**FIGURE 2.** Vertical distribution of the sound speed, nonlinearity parameter $B/A$ and $\varepsilon \text{Re}_a$ in the Southern Baltic
Real-time Imaging of Synthetic Aperture Sonar

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Synthetic aperture sonar (SAS) imaging is a worldwide high-tech. Because the high bearing resolution of SAS is based on a very large virtual aperture through array motion, the computation load for real-time processing is very heavy, especially the memory capacity requirement is a bottleneck for the processing. A new null-deposit twin-pipeline parallel processing algorithm for aperture synthesis is proposed in this paper. A wide-band low-frequency SAS trial along with the result of the real-time target contour imaging is also presented. The results show that the new parallel processing algorithm and target contour imaging are practicable for real-time processing of SAS.

INTRODUCTION

It is well known that a synthetic aperture sonar (SAS) can provide high-resolution images with a bearing discrimination independent of both range and frequency. It is much promising in ocean exploration and mine hunting applications [1][2]. But the practical application of SAS has hardly been seen in literatures.

There are two problems for the application of SAS. First, it is difficult to process in real-time. Because the high bearing resolution of SAS is based on a very large virtual aperture through a small physical array motion, the element number of the synthetic aperture can be more than thousands. Especially, all element inputs of the synthetic aperture must be stored in each DSP memory in order to calculate the aperture output. So the memory capacity requirement becomes a bottleneck in SAS real-time processing.

Further more, the synthetic aperture system design must take into account the requirement to preserve signal coherence. It is also confronted with the anti-reverberation detection problem [3]. Several techniques, such as a very stable tow-fish, a wide-band low-frequency Vernier-array, the simplification and parallelization of the algorithms, are adopted in our SAS trial prototype.

In this paper, a new null-deposit twin-pipeline parallel processing algorithm of aperture synthesis is proposed, and a wide-band low-frequency SAS shipboard trial along with the result of real-time imaging of target contour is presented.

THE REAL-TIME PROCESSING ALGORITHM

The most convenient synthetic aperture algorithm is the coherent beamforming with exact sphere wavefront delay compensation, shown as:

\[ b(k, j) = \sum_{i=k-N/2}^{N/2} x_i(\tau(i, j)) \]  

(1)

\[ \tau(i, j) = \frac{1}{c} \sqrt{\left(\frac{i \cdot d}{2c}\right)^2 + (j \Delta)^2} \]

Where \( b(k, j) \) is the synthetic aperture beam-output with \( k \) the displacement of array and \( j \) the focusing slant range, \( x_i(t) \) is the space sampling input of SAS, \( N \) the synthetic aperture element number, \( d \) the space between elements of the physical array, \( \Delta \) the time interval of sampling, \( c \) the sound speed.

Obviously, the calculation of the equation needs all of \( N \) element inputs to be stored in a processor memory. For the general parallel processing with available multi-processors, the \( b(k, j) \) is calculated by range or time division. But even so, its memory capacity requirement of each processor would be more than ten mage-words (in the case of 5m’s range division), which is usually much difficult for a general DSP hardware.

Further more, since the beam-output is a sum of \( N \) element inputs, then any element input is a contributor to \( N \) beam-outputs. It can be derived from equation (1)

\[ bb(i,j)=bb(i+1,j)+x_k \left( Tr(i,j) \right) \]

\[ i=N, N-1, ..., 0 \]

(2)

where \( bb(i,j) \) are the middle product of beam-outputs, \( x_k \) the current element input. It is seen that let \( bb(N+1,j)=0 \), then \( bb(0,j)=bb(k,j) \).

Accordingly, a new null-deposit twin-pipeline parallel processing algorithm for the aperture synthesis is formed, as shown in Fig.1. It incepts only the current element input on the input pipeline, calculates the middle product of \( N \) beam-outputs by range division,
and only one final beam-output is obtained on the output pipeline in a step. It is a null-deposit algorithm from the point of view of the input-output data flow.

FIGURE 1. Null-deposit twin-pipeline parallel processing algorithm of SAS

The delay-sum beamforming usually needs a high sampling frequency of inputs for exactly delay compensation, but the beam output can be calculated down-sampled. It means that the memory capacity requirement of the middle product of $N$ beam-outputs could be less than that of $N$ array-element inputs. So the memory capacity requirement can be further reduced in proportional to the rate of down-sampling by adopting the twin-pipeline parallel processing algorithm. As the result, the real-time SAS processing machine is total consisted of three standard SHARC21060 boards (eighteen CPUs), in which each CPU has 512K words memory capacity.

TRIAL AND RESULTS

In the SAS trial, a custom tow-fish was towed dynamically onboard ship, four Vernier-arrays were adopted to solve the contradiction between the tow-speed and the detection range, its working frequency is 7.5–15kHz. The real-time imaging of SAS trial results are shown as Fig.2. There are two cross targets with 1m and 2m size respectively placed in the lake. Because the bearing resolution of the SAS prototype is 0.12m, the targets contour imaging can be seen clearly in the figure.

FIGURE 2. The result of SAS real-time imaging

CONCLUSION

The effectiveness of the new null-deposit twin-pipeline parallel processing algorithm of aperture synthesis has been proved successful from the practical SAS trial results, and it takes an important role in the design of SAS signal processing machine with general DSP chips available. Taking the advantage of the wide-band low-frequency Vernier-array and a very stable tow-fish design, the targets contour imaging is obtained in the SAS trial. All of that are useful for the further research of the SAS application.

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Underwater Imaging Sonar using an Acoustic Lens for AUV “URASHIMA”

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Underwater imaging sonar has been developed for the obstacle avoidance of the Autonomous Underwater Vehicle (AUV) “URASHIMA”. The acoustic image, the distance and the azimuth of the target are obtained by using an acoustic lens. Transmitting three frequencies of 400k, 500k and 600 kHz, the direct image can be created on the receiving array, which consists of 128 x 128 elements. By using acoustic lens system, it was possible to omit the enormous calculation for two-dimensional beam forming, and reducing the size of the equipment was achieved. In this report, the comparison of theoretical calculation using PE method, which is used in the field of sound propagation in the ocean, with the experimental results in a water tank for acoustic lens are shown.

INTRODUCTION

In March 2000, Japan Marine Science and Technology Center (JAMSTEC) has developed AUV “URASHIMA” [1]. This AUV is an experimental one for investigating the performance of observation equipments, motion of the vehicle and autonomous cruising. The AUV “URASHIMA” has six kinds of acoustic systems and now under experiment in the sea [2]. These are an underwater imaging sonar for obstacle avoidance, an acoustic telemetry for communication to mother ship, a homing sonar for tracking the own position, a side scan sonar for surveying seafloor, a Doppler sonar for measuring own speed and an altimeter for measuring own height above sea-floor. This report describes the outline of the imaging sonar, results of water-tank experiments and simulation by PE method.

Acoustic lens for imaging sonar

Imaging sonar installed on AUV is the forward looking sonar using acoustic lens method, and the sound image and the distance and the azimuth of the target are output for obstacle avoidance. Transmitting frequencies are 400k, 500k and 600 kHz, and the direct image through the lens can be created on the receiving array, which consists of 128 x 128 elements as shown in Fig.1. By using acoustic lens system, it was possible to reduce the enormous calculation for two-dimensional beam forming, and the reducing the size of...
the equipment was achieved. The present lens used in this sonar is a spherical single lens with the diameter of approximately 400 mm and made of Acrylic resin. Figure 2 shows an experimental result in water tank for sound pressure level on axis of the lens at the frequency of 500 kHz. It is shown in this figure, that the focal length of this lens is approximately 240 mm. Figure 3 shows an observed result for sound pressure on vertical plane at the focal point at the same frequency. Equivalent beam width of the lens is approximately 1 degree.

Simulation of Acoustic Lens using Parabolic Equation Method

With the rapid progress of the ocean acoustic tomography, the Parabolic Equation methods that accurately simulate the underwater sound wave propagation are developed and can be analyzed within a reasonable cpu-time and memory. In order to develop a new field of application of this PE method, the underwater acoustic lens used to comprise ultrasonic imaging system is studied from the standpoint on wave theory with PE method. The finite-difference PE method combined the Douglas operator scheme to the Pade approximation form developed for sound propagation in range-dependent ocean model is applied to numerical analysis of an acoustic lens. Generally, acoustic lens for the ultrasonic image equipment is designed based on the ray-optics. However, it is difficult to know the effect of loss and diffraction by the lens. While, the PE method may be easily obtained sound field distributions, focal length, effects of finite size, and resolution of the acoustic lens, etc. We simulated the 3-dimensional sound field using the present lens geometry. Figure 4 shows a simulated result of sound field through the lens. Gaussian sound source is placed at x = 0, and the lens is placed at x = 2 m in the figure. We intend to compare simulated results with observed data and verify the accuracy of this PE method. We will design the lens with better performance using the PE method which includes aspherical and multiple lens system for future problem.

Conclusion

We developed an underwater imaging sonar for obstacle avoidance for the AUV “URASHIMA” using acoustic lens method. The result of the experiment in water tank shows a good resolution for a target. We are now under testing this imaging sonar installed on the AUV in the real sea. The simulattin by PE method is a powerful tool for the design of acoustic lenses. We will improve acoustic lens for this sonar using PE method.

References

Reconstruction of Layered Elastic Bottom Parameters by Measurements of Sound Reflection Coefficient

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The relation between the sound reflection losses measured at fixed grazing angles and frequencies and the characteristics of the sediment layer and underlying half-space is considered. Based on this relation, a method of the reconstruction of the sea bottom parameters is developed for a ocean bottom consisting of a single sediment layer overlaying a semi-infinite elastic half-space. Using this bottom model, the reconstruction of the characteristics of a layered elastic bottom is performed from the experimental and numerically simulated data with induced synthetic error.

The determination of the geoacoustic parameters of the sea bottom from acoustical data is an important problem with numerous applications in geophysics, oceanology, geology, and seismology [1,2].

METHOD OF INVERSION

Our approach is based on the fact that the shape of the projection of the selected level of the reflection losses onto the plane of medium characteristics depends on frequency at a fixed grazing angle and depends on grazing angle at a fixed frequency.

For arbitrary bottom model biunique correspondence between layered bottom parameters and value of reflection losses exist. On the other hand specified value of reflection losses may be obtained for different sets of layered bottom parameters. In the bottom parameters space, many points correspond to every specified value of the reflection losses. Coordinates of each point determined the set of layered bottom parameters, which provide obtaining specified value of the reflection losses. Existence of measurements errors lead to transformation of points to regions, because all nearest points should be accepted if difference between calculated and measured values of reflection losses less then measurement error. For single measurement, all points in the bottom parameters space meet specified value of the reflection losses are equivalent. If number of independent measurements is enough, in the bottom parameters space will exist only one common region which meet all measurements. It’s coordinates will determine the estimated values of layered bottom parameters.

Let’s illustrate this approach using numerical examples. The subscript $\infty$ indicates the parameters characterizing the elastic half-space.

Using measured value of the reflection losses, we can select the region of bottom parameters at which the given value of the reflection losses is possible. The error in the measurements of the reflection loss $\varepsilon$ will determine the range of the possible variations of the quantity $RL(\theta)$ relative to the experimentally measured values $RL(\theta)$, and, hence, in the plane $(d,C_{\infty})$ it will determine the width of the region where the calculated losses coincide with the experimental data within the required accuracy. We will call such regions the "parameter regions".

The shape of the parameter regions can considerably vary with changes in the selected value of RL, as well as with variations in the frequency and in the grazing angle. The fact that the shapes of the parameter regions corresponding to a given range of the variations of the reflection losses depend both on the frequency and the grazing angle may be used for the determination of the parameters of a layered bottom.

Let us consider a numerical realization of our method for synthetic model data, where two bottom parameters $d$ and $C_{\infty}$ are unknown. In the graph representing the results of calculations for fixed values of the frequency and the grazing angle, we select the regions corresponding to the variation of the reflection loss from $(RL_{m-\varepsilon})$ to $(RL_{m+\varepsilon})$. We perform a similar operation for all frequencies and angles, at which the experimental values of the reflection loss were obtained. The resulting parameter regions determined in the $(d,C_{\infty})$ plane will coincide only partially, due to the difference in their shapes.

Thus, we determine the interval of the values of the sea bottom parameters that provide the experimental values of the reflection loss at all frequencies and angles used in the measurements.

The procedure of $d$ and $C_{\infty}$ determination is illustrated in Fig. 1. In the graphs the values of the function are represented by shading: the dark areas in the graphs correspond to the maximal values of the function.

In the Fig.1a parameter regions for three frequencies under study ($f_1=16$ Hz, $f_2=500$ Hz, $f_3=1000$ Hz) are superimposed. The region of overlap is marked by dark gray and corresponds to the estimated values of $d$ and $C_{\infty}$. The obtained $d$ and $C_{\infty}$ values agree well with the exact values of $d$ and $C_{\infty}$. Result of estimation for three different grazing angles and for three angles and three frequencies are shown in the Fig. 1b,c. The estimation error is determined by the width of the region of overlap.

The higher is the measurement accuracy and the greater number of the measurements, the higher accuracy can be achieved in the estimation of medium characteristics. For procedure formalization, we introduce the function $\phi$ determined in the domain of the parameters to be
estimated as

$$\phi = \sum_{i} \sum_{k} \Theta(p_1, p_2, \ldots, p_n)$$

where

$$\Theta(p_1, p_2, \ldots, p_n) = \begin{cases} 1, & |RL_i - RL_m| \leq \varepsilon \\ 0, & |RL_i - RL_m| > \varepsilon \end{cases}$$

$p_1, p_2, \ldots, p_n$ are layered bottom parameters, $k$ is number of frequencies, at which experimental measurements were done, $i$ is number of angles used for the experimental measurements, $\varepsilon$ is measurement error, $RL_m$ and $RL_c$ are the measured and calculated values of the reflection losses. We determine the parameters of medium by the maximum of the function

RESULTS OF RECONSTRUCTION

Angular dependencies of reflection loss published by N.R.Chapman [1] were used for estimation of $C_l = \rho = \rho_{in}$. On the Fig.2. During calculation of $\phi(C_l, \rho)$, $C_l$ was varied for excluding strong $C_l$ influence on reflection loss. Results of $\phi(C_l, \rho)$ calculation are presented on the Fig. 2.

![Figure 2](image2.png)

FIGURE 2. Results of calculations of $\phi(C_l, \rho)$ and $\phi(C_l, d)$ for angular dependence of reflection losses.

The function $\phi(C_l, \rho)$ has maximum in the point with coordinates $C_l = 2550 \text{ m/s}$, $\rho = 2.65 \text{ g/cm}^3$.

During calculation of function $\phi(C_l, d)$ density $\rho$ was varied (note that $\rho$ was determined during the first step of estimation). Result of $\phi(C_l, d)$ calculation is presented on the Fig 2.

We also use the method described above for synthetic data at three different grazing angles and 30 frequencies in the range from 16Hz to 1000 Hz for every angle. We calculate the function $\phi$ for $C_l$ varying within the limits determined by the statistical data on its variations in the real conditions. The result obtained from the calculation of the function $\phi(\rho, C_l)$ is shown in Fig.3 in the $\rho - C_l$ plane. From Fig. 3, one can see that the function $\phi(\rho, C_l)$ reaches its maximum at the plane $(\rho, C_l)$ in the point with the coordinates $(C_l = 2660 \text{ m/s}, \rho = 2.8 \text{ g/cm}^3)$. Since the value of $C_l$ was varied in the calculation of the function $\phi(\rho, C_l)$ and the variations of $C_l$ have the most profound effect on the reflection loss, one should expect that the variations of all other parameters of the model will cause no additional changes in the estimated values of $C_l$ or $\rho$. The accuracy of the estimation of these values is determined by the width of the maximum of the function under study along the $C_l$ or, correspondingly, the $\rho$ axis.

To determine the values of $d$ and $C_l$, we calculate the function $\phi(d, C_l)$ with a simultaneous variation of the half-space density $\rho$. The results obtained by calculating the function $\phi(d, C_l)$ are shown in Fig. 3. From Fig. 3 one can see that the estimated values are $C_l = 440 \text{ m/s}$ and $d = 18 \text{ m}$.

![Figure 3](image3.png)

FIGURE 3. Results of calculations of $\phi(C_l, \rho)$ and $\phi(C_l, d)$ for frequency dependence of sound reflectivity.

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Dual-frequency Echosounder Seabed Sediments Classification System

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This paper describes an upper seafloor sediments classification system based on a dual-frequency echosounder. This work is a development of our previous researches on seabed classification such as remote measurement of attenuation in sediments and normalized cumulative energy function for classification of seabed sediments. This method includes extraction of features from normal incident echo from seabed. Horizontal variation of echo is measured for evaluation of roughness of seabed. PCA(Principle Component Analysis) of features is made using Adaptive Principle-component Extraction Neural Network. Neural network (Pi-Sigma network) approach is used for classification of seabed sediments. For sea region, where no a priori knowledge of seabed is available, clustering analysis by fuzzy c-means is made and then results are compared with direct sampling. System was constructed and experiments were conducted in lakes and seas. Results were compared with direct sampling and sidescan sonar sonogram.

INTRODUCTION

Sea bed classification has many applications, including trenching, pipeline and cable laying, construction of off shore foundations, marine anchorage sites, studies of slope stability and some military applications such as mine warfare and mine counter measures operations. Monitoring of seafloor sediment has been mainly carried out by grab sampling, which is very inefficient and expensive. Acoustic techniques provides capability for real time sea-bed classification without stopping the ship, thus reduce the survey labor intensity. We have conducted many research works of remote sensing method[1,2]. This paper is implementation of previous researches on upper seabed sediment classification and based on dual-frequency echosounder.

FEATURES EXTRACTED FROM NORMAL INCIDENT ECHO

From reflection coefficient of normally incident impulse at seabed surface we can get impedance of sediments and then we can make classification of sediments. But reflection coefficient is difficult to measure and must be corrected by source level, sea depth and deviation of sound beam from vertical direction caused by pitch and row of ships in motion.

In previous work [1] we define normalized cumulative energy function as

\[ E(t) = \int S_2^2(s) d\xi S_2^2(s) d\xi / \int S_2^2(\xi) d\xi \]

as a measure of part of energy of reflected impulse at any time compared with total energy at reflected impulse. The greater the attenuation in seabed sediments, the faster the cumulative energy function tends to 1. For convenience of calculation we use some quantatized values, such as the lengths of signal, in which energy reaches 80% and 95% of total energy and percentage of total energy at lengths of signal of 0.7ms, 1.5ms and 3ms.

As to count roughness of seabed, We can (a) measure the variation of depth of water while ship is moving on, (b) make comparison of envelope of successive echoes, which gives a measure of nonhomogenieties in sediments or roughness of seabed surface, (c) measure correlation between emitted and reflected signals, which gives measure of distribution of energy between reflected and scattered sound and also roughness of seabed surface.

PRINCIPLE COMPONENT ANALYSIS, CLASSIFICATION AND DISTERING

If there are too many features, classification process is very complicated. So it is need to extract principle components. For convenience of operation we use adaptive principle component extraction neural network (APEX) for extraction.

To make geometrical classification of samples in space of features we must determine boundary of classification. For convenience we use Pi-Sigma Network (PSN) for classification.

We suggest two mode of operation, trained and untrained. For sea area, where types of sediments at
some part are known we use these types of sediments to train our system and construct database for classification and then use this database to make classification of sediments at other parts of sea area. For sea area, where no sediments data are available, we make survey of whole sea area, calculate feature vectors of samples. After measurement we make clustering analysis according similarities of samples. After clustering we determine characteristics of samples by comparing with samples taken directly from sea bottom. For clustering we use fuzzy C-Means (FCM) algorithm.

RESULT of EXPERIMENT

Sea trail has been conducted at lake and sea area near Qingdao and Sanya. Types of sediments are checked by divers and compared with chart data and sonogram of side scan sonar.

We used two-dimentional analysis on: reflection coefficient—normalized energy cumulative coefficient, percentage of energy in 0.7ms—percentage of energy in 1.5ms, length of 80% energy—length of 95% energy, correlation of mirror reflection—difference of envelops of echoes. We can see from Fig 1 that silt, clay, sand silt, silty sand and fine sand can be divided according to position of their features on two-dimensional plane. Features of rock some how overlap with features of other kind of sediment. That because features of rock are rather dispersed. So it is important to distinguish rock with other sediment.

It is difficult to determine boundary surface if there are many features. So we used principle component analysis of above-mentioned features. After training of PCA neural network, we get weight for features. We discover for classification of most type of sediment, not including rock and sandy silt, the contribution of first two components reaches 85-90%. So we mainly used only first two components. For rocks and sandy silt contribution of first two components is small. There are large errors in PCA. So in addition to first two components we choose errors of PCA as third classification feature.

REFERENCES


FIGURE 1. Percentage of energy in 0.7ms - percentage of energy in 1.5ms
Measurement of Characteristics for M-Sequence Signal of a 200 Hz Source for Ocean Acoustic Tomography

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The acoustic characteristics of the 200 Hz source for ocean acoustic tomography were evaluated using M-sequence signal, and compared with the results of the conventional method by using sinusoidal waves. In this report, the results of the measurement of M-sequence signal with various wave numbers per digit are shown, and then the experimental results using sinusoidal waves at the depth of 800, 1000 and 1200 m are shown. The relationship between optimum wave number per digit of M-sequences and the frequency response using sinusoidal waves is discussed.

\section*{INTRODUCTION}

We have been developed a 200 Hz sound source using a giant magnetostrictive material for ocean acoustic tomography \cite{1}, and developed a real time tomography system using this sound source for measuring the water temperature distribution in the 1000 km range \cite{2}. Our tomography system consists of a transceiver in which sound is transmitted and received, a surface buoy for real-time data transmission via INMARSAT-C and a mooring system for them. In our experiments of the ocean acoustic tomography, M-sequence signal is used in order to detect a signal in the noisy environment in the ocean and improve the resolution on the time axis after correlation of signals. The frequency of the sound source for our tomography transceiver is 200 Hz to ensure the propagation distance of 1000 km. In this report, the acoustic characteristics of the source were evaluated using M-sequence signal, and compared with the results of the conventional measurement by using sinusoidal waves.

\section*{EXPERIMENTS}

In June, 1999, the measurement was carried out at the area of approx. 1500m water depth in the Suruga Bay. As shown in Fig.1, A transceiver in which contains a 200 Hz source was suspended from a test vessel by a wire. The acoustic characteristics of the 200 Hz source was measured at the depth of 800, 1000 and 1200 m. In the transceiver, the following equipments for measurement were installed: Sine wave generator and data acquisition system in a pressure proof container and a hydrophone. Swept frequency range was constrained for the capacity of data acquisition equipment, and it was swept from 140Hz to 280Hz for every 5 Hz. The distance between a sound source and a hydrophone is 3m. Another purpose of these measurements was to fix the matching constant between sound source and power amplifier.

On the last day of the experiment, M-sequence signal of the tenth degree used in our usual tomography observation was generated in the transceiver and transmitted from the sound source at the same conditions as above measurements. The received waveform by a hydrophone was recorded in a DAT (Digital Audio Tape) recorder stored in another pressure proof container as shown in Fig.1. The wave numbers per digit of M-sequence were changed from 2 to 5 waves to evaluate the effect of the amplitude and the pulse length after correlation processing.

\begin{figure}[h]
\centering
\includegraphics[width=\textwidth]{figure1.png}
\caption{Experimental apparatus for the measurement of characteristics of a 200 Hz source.}
\end{figure}
RESULTS

Results observed by sinusoidal waves are shown in Fig.2. These are frequency characteristics of sound pressure level of the sound source of the No.7 transceiver measured at the depth of 1000m. The resonance frequency increases, as the depth deepens. It is a peculiar phenomenon of the sound source that the pressure in the air space in the sound source increases by the function of the pressure compensator, and then resonance frequency increases. The largest sound level was obtained at the depth of 1000m because the matching was adopted at this depth.

Figure 3 shows the correlated waveform when the wave number per digit of M-sequence was made to change at 1000 m depth. Though the amplitude is small for 2 waves, the pulse duration is short, and the resolution is good for discrimination of sound ray paths for tomography observation. Though the amplitude increases with the increase of the wave number, the pulse duration is lengthened, and the resolution becomes bad. As shown in Fig.4, in the case of 5 waves per digit, pulse shape variation was observed at three depths. The largest sound pressure was obtained on 1000m, next 800m and smallest at 1200m. It is correspondent to the order in the measurement using sine wave. It is proven to be almost equaled to the result for continuous wave because the transient response of sound source has ended for the case of 5 waves for M-sequence.

CONCLUSION

In the tomography experiments, better signal to noise ratio for long-range sound propagation, and better resolution to discriminate a specific sound ray path are important. In this experiments, the acoustic characteristics of 200 Hz sound source at three depths in real sea. When the wave numbers per digit of M-sequence signal became larger, the results were closer to the results of continuous wave but the result of smaller wave numbers is different from above results. If the resolution is more important for the measurement of tomography under enough signal to noise ratio, the result that 2 waves were excellent was obtained from these results.

REFERENCES


FIGURE 2. Frequency characteristics of sound pressure level for the No.7 sound source measured at the depth of 800m (solid line), 1000m (dotted line) and 1200m (dashed line).

FIGURE 3. Correlated results for M-sequence signal for 2 - 5 career waves per digit measured at the depth of 1000 m.

FIGURE 4. Correlated results for M-sequence signal of 5 career waves measured at the depth of 800m (solid line), 1000m (dotted line) and 1200m (dashed line).
The Effect of Littoral on the Day/Night Fish Abundance Estimates by Hydroacoustics

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The Biosonics 101 echosounder, 420 kHz, was used to study fish abundance and distribution in two dam reservoirs: one without litoral and second consisting of three basins differing substantially by the % of litoral coverage. Measurements were performed during day and night using echo counting method (40logR). The ratio of day to night estimates was changing between \textgreater{}1 and \textless{}0.1. The results presented show that hydrogeomorphology and man’s activity (operating the power station) may have important consequences for the accuracy of hydroacoustical monitoring of fish stocks. On the other hand hydroacoustics, if appropriately used, can indirectly supply the information on the effectiveness of litoral as refugium for fish.

INTRODUCTION

Hydroacoustics is commonly used to estimate fish stocks both at sea and in fresh waters [1]. Although night surveys are recommended it often happens that due to logistic problems measurements are performed also during a day. Underestimation of stocks using day measurements can be extremely high in shallow waters, where fish can hide within a vegetation.

The aim of this paper was to investigate what is the influence of littoral coverage on day and night estimates of fish stocks.

MATERIALS AND METHODS

The Biosonics 101 echosounder, 420 kHz, was used to study fish abundance and distribution in two dam reservoirs. In the Solina dam reservoir, in which due to the power station activities the water level fluctuation attains 10 m, littoral is practically absent. The Dobczyce reservoir, consists of three basins which have different morphological structure and differ substantially by the % of littoral coverage. Measurements were performed in May, June and September 2000 along the zig-zag transects both during the day and night. Since in both reservoirs even during the day fish were dispersed (about 90% of single fish) the echo counting method (40logR) was applied.

RESULTS

In the Solina reservoir fish density estimates during the day were always higher than at night (Fig. 1). This can be explained only by diurnal vertical migrations of fish (Fig. 2). Solina is a mesotrophic reservoir with rather low phytoplankton and fish concentrations [2]. During the day fish were distributed in the whole water column while at night majority of fish migrated to the reachest in food surface waters and could not be detected by the looking down echosounder. The difference between day and night estimates was the largest in May, when food was the most scarce. This suggest that migrations were rather caused by food distribution than avoiding predation, which agrees with anglers, reporting very few predatory fish in Solina [3].

![Figure 1](image-url)
The Dobczyce reservoir supplies drinking water for Cracov. Although it has high phosphorus loadings which lead to eutrophic conditions [4], high quality of water is possible thanks to biomanipulation aiming at controlling fish population structure in such a way that a very high proportion of predatory fish is maintained. The pike-perch (Stizostedion lucioperca) amounts to nearly 20% of fish biomass, perch (Perca fluviatilis) to 13% and pike (Esox lucius) to 5% [5]. The reservoir consists of three basins: the deepest, with sharp rocky shores closed by dam Dobczyce Basin (BD), the largest along a river valey Myślenice Basin (BM), and shallow Wolnica Bay (ZW). The three basins differ substantially by the % of litoral coverage. Comparison of day and night estimates of fish concentrations in these three basins (Fig. 3) also shows large differences between them. The higher the % of litoral coverage, the higher fish concentrations and larger difference between the day and night estimates. With no doubt these differences in day/night fish abundance estimates are caused by horizontal migrations from the litoral during a day to open water during the night. It is quite obvious that litoral offers much better habitat for fish than open pelagial not only providing refugium from predators, but also nursery grounds and leads to higher fish concentrations in basins with high litoral coverage.

The preliminary results presented here show that hydrogeomorphology and man’s activity (operating the power station) may have important consequences for the accuracy of hydroacoustical monitoring of fish stocks. On the other hand hydroacoustics, if appropriately used, can indirectly supply the information on the effectiveness of litoral as refugium for fish.

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Gulf of Gioia Ocean-Acoustic Soliton Predictions and Studies, Phase 1.

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Along the path from the Strait of Messina into the Gulf of Gioia the interaction of the acoustic field with the soliton trains is investigated. Joint oceanographic and acoustical predictions are undertaken. Transmission loss patterns, calculated with the PE model, are presented for diverse conditions along the path of propagation. This paper constitutes the first phase of the study.

SALINITY AND TEMPERATURE FRONTS

The Strait of Messina contains a sill that rises to within 80 meters of the ocean surface. As the semidiurnal tide moves over the sill in a particular direction, it generates an internal bore by depressing the isopycnals. As the tidal motion reverses the leading edge of the initial depression forms an internal bore and propagates away while the back forms another internal bore that undertakes a hydraulic jump over the sill in the direction of the reversed tidal flow. At 1.07 semidiurnal tidal cycles the salinity distributions has evolved to the configuration shown in Figure 1. There is a salinity gradient or front located close to the mid depth over the sill. There are two water masses, upper and lower, on each side of this front. Previously Hopkins, Salusti, and Settimi [2] have investigated the time evolution of the interface between the two water masses with a two layer model. The internal bore that previously jumped over the sill, from left to right, has now moved close to the 45 km location and is about to disintegrate into solitons. Another internal bore that previously jumped over the sill, from right to left, has generated the train at 5 km. The results were obtained with the Smolarkiewicz [1] nonhydrostatic model for conditions that exist in the Strait of Messina (in 2.5 dimensional geometry 2.5 means that derivatives perpendicular to the 2D plane are neglected).

SHELF BREAK EFFECTS

The basic phenomena occurring as one proceeds from the Messina Sill onto the Gulf of Gioia shelf break, can be studied with a simplified set of analogue problems. From the real topography, we constructed one configuration that consists of the Messina sill only and, another one that contains the sill and a subsequent shelf break towards the Gulf of Gioia. The configuration consisting of the sill only is identical to the geometry and parameters of the simulation shown in Figure 1. At 1.66 semidiurnal tidal cycles the bore that was located at around 45 km, Figure 1, has propagated further and has disintegrated into solitons. The resultant soliton train is located at about 57 km and is shown in Figure 2 in terms of sound speed that was calculated from the predicted temperature and salinity fields. Note the sound speed gradient or front located close to the surface on top of the sill. This corresponds to salinity and temperature fronts and indicates the interface location between the water masses on the sill. The soliton train consists of 5 visible depressions with the first 2 being the most pronounced in terms of amplitude and wavelength. In these depressions the sound speed and temperature are increased while the salinity is decreased. The amplitudes of the first 2 depressions are around 100 m and 50 m. The corresponding wavelengths across the depressions are about 2.5 km and 2 km.

The second analogue problem incorporates a shelf break into the first one. This is based on the real topography, constructed using the Smolarkiewicz [1] nonhydrostatic model for conditions that exist in the Strait of Messina (in 2.5 dimensional geometry 2.5 means that derivatives perpendicular to the 2D plane are neglected).
pographic rise that occurs when proceeding from the Messina sill to the Gulf of Gioia. The rise, however, is cut off at a depth of 270 m for computational restrictions that occur in a 2 dimensional plane. Later a 3 dimensional simulation will be undertaken. The simulation results for the case with a shelf break are shown in Figure 3, at the previous time of 1.66 semidiurnal tidal cycles. Note that there is a 10 km shift in the scale due to the use of 70 km and 80 km horizontal domains for the two cases. The most striking difference over the shelf break is the shape of the first soliton, its width is larger and the phase speed is smaller. The shelf break has a retardation and widening effect on the first soliton. Its leading edge is being drawn over the shelf. The 2 and 3 solitons are squeezed together and have larger amplitudes over the shelf. The 2 has a phase retardation while the 3 has not. The 4 and 5 are only slightly visible.

**ACOUSTICAL FIELD ON SHELF BREAK**

To illustrate the acoustic effect of the soliton field shown in Figure 3 an underwater acoustic model based on the parabolic equation [3] was used. To highlight effect of the smaller solitons in the packet an acoustic source at 825 Hz was placed at a depth of 140 m at the range A (Figure 3). The acoustic energy was directed toward the sill (upslope) thus ignoring the large feature. The results of this acoustic simulation are shown in Figure 4. The environment for a second acoustic simulation was developed by shifting the sound speed field 2 km toward the shelf which simulates the natural propagation of the solitons. The result of this second simulation is shown in Figure 5. It is apparent that the presence of the smaller solitons has a rather pronounced effect at the sill and beyond causing large differences in the acoustic field.

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