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Welcome to the 2001 workshop on Underwater Acoustic Signal Processing. This year we were fortunate enough to have Jim Preisig of Woods Hole and Tom Green of DARPA organizing the special sessions on *Adaptive Signal Processing in Nonstationary Environments*. These sessions have fewer papers in order to encourage more discussion and interaction. We have additionally converted the Wednesday evening session, which is traditionally a plenary lecture, into a regular lecture session with presentations related to the special topic.

The organizing committee would like to thank and acknowledge the sponsorship of John Tague at the Office of Naval Research and thanks Raytheon Systems Company for sponsoring our Wednesday evening dinner. We are also honored to present this year's UASP Award to Dr. John Costas.

We trust that you will enjoy the beautiful surroundings of the Whispering Pines Conference Center and hope you enjoy the workshop as much as the dining.

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2001 UASP Award Presented to Dr. John Costas

Some of the Contributions of John Costas to Communications and Underwater Acoustic Signal Processing

by Donald W. Tufts

As a result of historical development it became an axiom of communications theory and practice that bandwidth was a precious commodity and should not be "wasted" such as in the case of amplitude modulation systems which create a pair of identical side-band terms. In the early fifties the crusade to save bandwidth took special aim at the military and amateur radio services. Conversion from amplitude modulation to single side band (SSB) modulation was strongly encouraged. One issue of the Proceedings of the IRE in those days devoted itself almost entirely to SSB and became known as the "SSB Issue." One dissenting paper by John Costas, "Synchronous Communications," was permitted in that issue. He argued that there were actually many potential advantages to having a pair of identical side bands, especially for military applications. He further argued that the use of additional bandwidth, if properly exploited, could provide significant benefits and that such use should not be classified out of hand as a "waste" of bandwidth.

In that dissenting paper Dr. Costas also described a method for receiver oscillator phase synchronization for use with suppressed-carrier AM, double-side-band (DSB) signals. This synchronization procedure has been widely copied and used world-wide in direct-sequence spread-spectrum systems and has become universally known as the "Costas Loop." Experts in this field have often remarked that in spite of the longevity and widespread use of the "Costas Loop," its form remains essentially unchanged from the description in the original paper.

In a later paper in the Dec. 1959 issue of the IRE Proceedings, "Poisson, Shannon and the Radio Amateur," Dr. Costas expanded considerably on the potential advantages of spread-spectrum operation. This paper has become a classic in the intervening years, giving many communications engineering students their first serious exposure to a generalized view of bandwidth utilization. Some professors of electrical engineering have incorporated this paper directly into their communications theory courses. It is interesting to note in Section VII of this paper, "The Question of Channels," that some of the advantages claimed today for CDMA usage in cell-phone service are clearly detailed in this forty-two-year-old publication. (This paper has been reprinted and appears as the lead paper of a 1992 IEEE Press Book, "Multiple Access Communications-Foundations for Emerging Technologies," edited by Norman Abramson.)

In the 1960's the General Electric Company, Syracuse, NY, asked Dr. Costas to assist in the investigation of certain sonar system anomalies that had been observed. These unexplained anomalies were seriously limiting detection performance of certain sonar systems. An extensive investigation led to the conclusion that the large-TW-product waveforms being used were incompatible with the acoustic medium time spread and frequency spread parameters. John Costas identified the problem from sea data analyses, and he introduced a modified waveform composed of individual pulses whose duration and bandwidth were appropriate for the medium-spreading parameters, which had been identified. The processing of this pulse train involved coherent processing of the individual pulses separately with the multiple processor base-band outputs being combined in a hybrid coherent/non-coherent arrangement. This hybrid approach proved to be very effective in minimizing medium de-correlation losses. It also provided a very effective solution to both the original echo-ranging sonar application and the acoustic communications applications, which were further developed by Dr. Costas.

The unique nature of acoustic path interference (noise and reverberation) dictated that the transmitted burst be comprised of one pulse only per frequency channel and time position. The resulting transmission time-frequency array then took the form of a permutation matrix having special properties for optimum echo ranging and communications services. These special forms of the permutation matrix, discovered and developed originally by Dr. Costas, have become known as "Costas Arrays." Subsequently these special permutation matrices have been studied extensively by the mathematician Professor S. W. Golomb and others. John Costas' originality seems always to be at least 20 years ahead of conventional engineering thinking. We are just starting to understand the wide applicability of his concept of Residual Signal Analysis, which was presented at a UASP workshop in the 1980's.

Schedule at a glance

Wednesday October 3, 2001		Thursday October 4, 2001		Friday October 5, 2001	
		8:00-9:45	Session B Laurel	8:00-9:45	Session F Laurel
		9:45-10:15	Break Laurel	9:45-10:15	Break Laurel
		10:15-12:00	Session C Laurel	10:15-12:00	Session G Laurel
		12:00-1:00	Lunch Whisp. Pines	12:00-1:00	Lunch Whisp. Pines
		1:00-2:45	Session D Laurel		
		2:45-3:15	Break Laurel		
		3:15-5:00	Session E Laurel		
5:00-6:00	Welcome Reception				
6:00-8:00	Raytheon Dinner Whisp. Pines	6:00-8:00	Dinner Whisp. Pines		
8:00-9:30	Session A Laurel	8:00-?	SOB Session Laurel		

Sessions

Session A: Wednesday Evening, 8:00–9:30

Special Session on Adaptive Signal Processing in Nonstationary Environments I

A–1 *Complex Spatial Envelopes, Invariants, and Matched Field Processing*,
Harry Cox and Kevin Heaney, Orincon Corporation

A–2 *Improvements in FAST Subspace Tracking*,
Donald Tufts, University of Rhode Island

Session B: Thursday Morning, 8:00–9:45

Special Session on Adaptive Signal Processing in Nonstationary Environments II

B–1 *Covariance Matrix Filtering for Adaptive Beamforming with Moving Interference in Shallow Water*,
Bruce K. Newhall, Johns Hopkins Applied Physics Laboratory

B–2 *Scanned-Beam Dominant-Mode Rejection Of Non-Stationary Shipping Noise*,
Juan I. Arvelo, Jr., John Hopkins University, APL

B–3 *Cramer-Rao Bounds for Source Localization and Matched Field Tomography in a Partially Saturated Ocean*,
Arthur B. Baggeroer, Massachusetts Institute of Technology

Session C: Thursday Morning, 10:15–12:00

Special Session on Adaptive Signal Processing in Nonstationary Environments III

C–1 *Measurement-Based Adaptive Array Processing Performance Prediction*,
Norman Owsley, ONR and URI and Harry DeFerrari, University of Miami

C–2 *Source Motion Compensation using Waveguide Invariant Theory*,
Lisa Zurk, MIT Lincoln Laboratories

C–3 *Matched Field Processing – How Much Does It Really Benefit?*,
Nigel Lee and Brian Tracey, MIT Lincoln Laboratories

Session D: Thursday Afternoon, 1:00–2:45

Model Related Processing and Analysis

D–1 *Mismatch Effects on Long-range Target Detection**,
Brian Tracey and Nigel Lee, MIT Lincoln Laboratory

D–2 *Efficient inversion in underwater acoustics*,
Zoi-Heleni Michalopoulou and Xiaoqun Ma, New Jersey Institute of Technology

D–3 *Estimation and Interpretation of Broadband Mode Statistics*,
Kathleen E. Wage, George Mason University

D–4 *Optimal Array Design and Sensitivity for Mode Filtering*,
Tsung-Jieh Shiao and John R. Buck, University of Massachusetts Dartmouth

Session E: Thursday Afternoon, 3:15–5:00

Active SAS and Adaptive Methods

E–1 *Coherence of Pulsed Signals and Implications to Synthetic Aperture Sonar Processing*,
Enson Chang and Mark D. Tinkle, Dynamics Technology, Inc.

E–2 *Synthetic Aperture Sonar Image Segmentation using the Fuzzy C-Means Clustering Algorithm*,
J. P. Stitt, R. L. Tutwiler, and A. S. Lewis, Penn State University, ARL

E–3 *Suppressing Reverberation by Multipath Separation for Improved Buried Object Detection*,
Ivars Kirsteins and John Fay, Naval Undersea Warfare Center

E–4 *ULV Decomposition as Preprocessor for Nonlinear Adaptive Algorithms Used for Blind Source Separation*,
Leon H. Sibul and Peter A. Yoon, Penn State ARL

<u>Session F</u>: Friday Morning, 8:00–9:45

Active Sonar

F–1 *The Application of M-Sequences to Bi-Static Active Sonar*,
Harry A. DeFerrari, University of Miami

F–2 *Multiple Ping Processing for Active Towed Array Sonar Systems*,
D. Maiwald, S. Benen, H. Schmidt-Schierhorn, STN ATLAS Elektronik GmbH, Germany

F–3 *Broadband active sonar detections using the time-reversal operator*,
David M. Fromm and Charles F. Gaumond, Naval Research Laboratory

F–4 *A Split/Merge Image Segmentation Algorithm for Detection of Targets in Range-Doppler Maps*,
A. Scott Lewis, J.P. Stitt, and R. L. Tutwiler, Penn State University, ARL

<u>Session G</u>: Friday Morning, 10:15–12:00

Passive Localization

G–1 *Direction Finding of Narrowband Signals Based on the Maximum Likelihood-Expectation Maximization Technique*,
Carol T. Christou,The MITRE Corporation

G–2 *A New Approach to Range-Focused Beamforming*,
Anton J. Haug

G–3 *Passive Sonar Interference Suppression using Matrix Filters*,
Richard J. Vaccaro, URI, Brian Harrison, NUWC, Amit Chhetri, URI

G–4 *Information Theory for Source Localization*,
John R. Buck, University of Massachusetts Dartmouth

Complex Spatial Envelopes, Invariants, and Matched Field Processing

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It is common practice in signal processing and time series analysis to use a representation of a band-limited random process as consisting of a slowly varying complex envelope multiplying a single frequency carrier. The convenience of this representation has lead to its nearly universal use in radar and active sonar. We introduce a spatial analogy to this representation that is useful for pressure fields in underwater acoustics. It is shown how the normal mode representation on an underwater acoustic field can be interpreted in terms of a single wave-number carrier and a complex spatial envelope. The extension to the range dependent case is discussed. This representation is particularly useful and can provide new insight in a number of applications. Using this approach, we develop the waveguide invariant recently popularized in the US by Kuperman. Matched field applications are then presented including the simplified computation of replicas for broadband applications and beam-based replicas for large volumetric arrays. Computational savings are shown to involve orders of magnitude.

Improvements in FAST Subspace Tracking

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The original FAST algorithm (Tufts, Real, and Cooley, ICASSP 97, IEEE SP Trans 99) tracks the dimension, principal singular values, and principal singular vectors of a signal or interference subspace from single column matrix updates. It uses rectangular windowed data support. Recently Fast Approximate Subspace Tracking(FAST) has been improved in the following ways:

1. Latency in initialization has been reduced by a FAST start for FAST.
2. Speed can be improved through multiple column updates with some loss in accuracy.
3. Accuracy is improved by incorporation of the effect of the discarded data, as well as the added data.

The basic principles of FAST will be explained. And some applications to real data and to simulated data will be presented.

Covariance Matrix Filtering for Adaptive Beamforming with Moving Interference in Shallow Water

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An approach has been developed for adaptive beamforming for mobile sonars operating in an environment with moving interference from surface shipping [1]. The approach is reviewed and its application to horizontal and vertical line passive sonar arrays in shallow water is examined. It is assumed that the sound source of each ship is drawn from an ensemble of Gaussian random noise, but each ship moves at constant speed along a deterministic course. An analytic expression for the ensemble mean covariance is obtained. In practice the parameters of each interferer are not known with sufficient precision to use the modeled ensemble mean as a basis for adaptive beamforming. Hence, techniques to accurately estimate the ensemble mean based on data samples are developed. The ensemble of covariance samples consists of rapidly varying random terms associated with the emitted noise and more slowly oscillating deterministic terms associated with the source and receiver motion. The non-stationary ensemble covariance mean can be estimated by filtering out the rapidly varying noise while retaining the slow oscillatory terms. Performance of the filters can be visualized and assessed in the "epoch frequency domain", the Fourier transform of the covariance samples. Techniques that identify and include the appropriate non-zero frequency contributions are better non-stationary estimators than the sample mean. Several such techniques are offered and compared. Previous simulations [1] considered only the case of horizontal line arrays in a single-path propagation environment. Extensions to a severe multipath shallow water environment will be shown. Differences in the effects of moving ships on horizontal and vertical line arrays in realistic underwater environments are elucidated.

[1] B.K. Newhall, "Covariance Matrix Filtering for Adaptive Beamforming with Moving Interference," presented at ASAP, March, 2001.

Scanned-Beam Dominant-Mode Rejection Of Non-Stationary Shipping Noise

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Passive acoustic detection of underwater targets is commonly limited by ambient noise. The shipping component of this noise field is usually dominant over the wind-driven component at low frequencies ($<$ 1kHz). Adaptive beamformers have been implemented to take advantage of the high degree of anisotropy introduced by shipping. Non-stationary ships, however, may require additional eigenvalues for complete suppression over the selected integration time. In the presence of multiple non-stationary ships, the number of available eigenvalues may be smaller than the required number for complete suppression of the noise field to its isotropic background level. Scanned-beam dominant-mode rejection is applied to the most dominant non-stationary ships before the adaptive beamformer in an effort to reduce the number of required eigenvalues to one per ship. The approach will be described and results will be presented with modeled and real data. This effort is supported by DARPA's Robust Passive Sonar (RPS) program.

Cramer-Rao Bounds for Source Localization and Matched Field Tomography in a Partially Saturated Ocean

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Performance analysis for matched field methods to date have used fully coherent models for the acoustic propagation. This implies that the modes or rays remain "phase locked." Since the phase interaction provides much of the information for matched field processing, any model, e.g. coupled modes or partially correlated rays, cannot use a single degree of freedom replica. Such examples are often described as "partially saturated" propagation Moreover, the closed form formulae for performance analysis using Cramer-Rao bounds are not applicable. Since partially saturated acoustic propagation are often more realistic models, it is important to determine the performance of matched field processors. Here, we derive closed form formulae for computing the Cramer-Rao performance bounds for partially saturated acoustic propagation.

Measurement-Based Adaptive Array Processing Performance Prediction

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A simple method that uses single hydrophone recorded data to predict the performance of a full-scale fixed array is described. The need to assess quickly and economically the likely site-specific performance of adaptive beamforming methods for a candidate fixed array site in a high shipping traffic area is the primary motivation for the work described.

A one week, 24 hour per day set of recorded hydrophone data is spectrum analyzed. These frequency-time "grams" are searched for events that correspond to the acoustic energy due to the passage of ships and periods of ship activity lulls. The so-called quiet intervals wherein no discernible high level discrete ships are present are used to estimate the broadband spectrum noise floor corresponding to the assumed continuous spatial wave-number spectrum and spatially uncorrelated ambient noise, $N_0(f)$, at frequency f. The high level ship passage events, HLEs, are tabulated with respect to the following: day of the week; time of day; frequency band, 100 to 200 Hz for example; and absolute peak spectrum level of the ship interference, $I(f) + N_0(f)$, in the reference band. The interference to noise ratio

$$r(f) = N\frac{I(f)}{N_0(f)}$$

for an array with N sensors referenced to the output of a conventional time delay-and sum beamformer steered at the interference is the estimated parameter required for performance prediction. Clearly, because the sensor is omnidirectional, only the dominant instantaneous HLE is identified. This masking of lower level events produces a downward bias in the HLE count. However, it is noted that it is frequently a single discrete interfering source that dominates the array performance at any one instant. Moreover, if the biased, single hydrophone HLE analysis argues in favor of the value of interference nulling, then the analysis described herein gives a solid lower bound on the performance that can be expected. With the data thus sorted, the following site acoustic profile is formed: HLEs versus day of the week averaged over 24-hours, HLEs versus time-of-day averaged over days of the week and the expected number of HLEs per hour that would represent either a mainlobe or a sidelobe limiting condition. The analysis is presented for a candidate site on the East Coast of Florida and simulations based on the estimated HLE density are given. Future work will augment this analysis with logged radar surface ship traffic patterns and spatial sorting of the shipping patterns to provide additional realism to site specific simulation based performance predictions.

Source Motion Compensation using Waveguide Invariant Theory

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Passive detection and localization of quiet, submerged sources moving in shallow water in the presence of shipping noise is a critical problem for the ASW community. Past work has demonstrated that adaptive processing can be applied to detect and localize stationary sources while mitigating loud interferers. However, source motion can cause significant signal gain degradation, loss of localization accuracy, and incomplete interference rejection. The severity of the motion loss is channel dependent, and a function of the amount of motion during the observation interval. Motion compensation techniques have been demonstrated in past work with data from the Santa Barbara Channel Experiment. These algorithms utilized a normal mode propagation model and a velocity hypothesis to successfully compensate for source motion. However, the technique requires good environmental knowledge and can be computationally expensive if a large hypothesis space must be searched. In this paper, the waveguide invariance principle is used to combine the time-frequency snapshots acquired from a broadband source to compensate for motion. The technique does not require knowledge of the underwater environment for the compensation and the search across the velocity space can be structured so it is computationally efficient. Results from this technique will be presented and discussed.

Matched Field Processing – How Much Does It Really Benefit?

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Matched field processing (MFP) has been widely proposed as an array processing technique for passive sonar detection in littoral environments. MFP is considered suitable for littoral environments because it models and exploits the coherent multipath energy of signals propagating in the shallow-water waveguide. By contrast, the more traditional plane-wave beamforming (PWBF) techniques assume only direct-path acoustic energy and ignore all multipath effects. PWBF, however, is much simpler and faster to implement than MFP.

This paper addresses the important question of how much is gained by using MFP in a shallow-water environment instead of PWBF. Under the assumption that MFP models the "truth," various simulations are presented to determine how much mismatch is incurred by using PWBF instead of MFP. The simulation environment assumes a 100-meter water channel, includes both horizontal and vertical arrays, and varies several parameters including array aperture, array spacing, frequency, and look direction. The simulation results indicate that the greatest mismatch occurs (i.e., MFP is most beneficial) for higher frequencies, longer arrays, and arrays with large vertical aperture.

Mismatch Effects on Long-range Target Detection*

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Acoustic propagation in shallow water is characterized by the presence of coherent multipath arrivals. Multipath propagation ensures that mismatch exists between received data and the single-path steering vectors (plane-wave or range-focused) typically used in sonar beamforming. This mismatch causes target self-nulling losses for adaptive processing that will reduce detection performance. Matched field processing (MFP) is a means of addressing this problem through the use of full-field steering vectors. In addition to giving localization in range and depth, MFP in principle allows multipath arrivals to be combined coherently, yielding higher output energy on target and extended detection ranges. However, MFP performance is sensitive to errors in the environmental model used to generate steering vectors. This leads to the basic question: does the mismatch introduced into MFP from environmental uncertainty outweigh the advantages of using a more realistic signal model?

In this talk we present results showing that MFP mismatch due to environmental uncertainty grows with increasing source range. This trend is predicted theoretically, and is seen in numerical simulations as well as data from the 1998 Santa Barbara Channel Experiment (SBCX). In contrast, simulations show that the mismatch introduced by using a single-path model remains constant with range. MFP steering vectors are generally less mismatched at short ranges, but may have higher mismatch than single-path steering vectors at long ranges.

Mismatch reduction for plane-wave or range-focused beamforming can be accomplished through subarray processing. Mismatch reduction for matched field processing can potentially be accomplished by in situ calibration of the environmental model using acoustic data. Concepts for mismatch reduction in both types of processing will be examined.

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Efficient inversion in underwater acoustics

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To address source localization in the ocean, matched field processing methods can be implemented leading to source location estimates through a matching process between measured and theoretically predicted replica fields. Although matched field processing can have excellent results, it is usually computationally intensive requiring multiple full field calculations. To avoid the excessive computational load, in this work we employ arrival time information of select ray paths and linearization of the inverse problem, obtaining estimates of source and receiver locations, sound speed, and bottom depth in a computationally efficient manner. The approach is tested on synthetic arrivals and also real data collected during the Haro Strait experiment. Localization, sound speed, and bottom depth estimation are performed successfully and efficiently, indicating the potential of the method for fast inversion when path identification and accurate arrival time estimation is feasible. For the Haro Strait data, results are compared to matched field processing estimates, that deviate from the expected values of the source and environmental parameters.

Estimation and Interpretation of Broadband Mode Statistics

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The low-order acoustic modes constitute some of the most energetic arrivals at long ranges. Analysis of both simulated and experimental data indicates that, for ranges on the order of megameters, these arrivals have a complicated structure due to internal-wave-induced coupling. Despite the complexity, broadband receptions from the Acoustic Thermometry of Ocean Climate (ATOC) experiment demonstrate that the mode signals do retain travel-time information at ranges of 3515 km, e.g., there are detectable trends in the arrival time of the first 10 modes over a period of five months [1]. Recent work suggests that the relative arrival times of modes 1-10 may offer a measure of internal wave strength.

Understanding the mechanisms and effects of mode coupling is a prerequisite for using the mode signals in applications such as tomography or matched field processing. The North Pacific Acoustic Laboratory (NPAL) experiment has provided an opportunity for further research on the statistics of broadband mode signals. From July 1998 to June 1999, the NPAL billboard array located west of Sur Ridge, California recorded transmissions from the ATOC source north of Kauai. The billboard array consisted of four 20-element vertical line arrays (VLA's) and one 40-element vertical line array, which were located transverse to the 3900 km path to the source. Horizontal aperture of the billboard was 3600 meters and vertical apertures of the 20-element and 40-element arrays were 700 meters and 1400 meters, respectively. The 40-phone VLA was equipped with temperature sensors to facilitate evaluation of the environmental fluctuations near the receivers over the year-long deployment.

An initial analysis of the NPAL receptions shows frequency-selective fading in the arrivals that is typical of long-range propagation through random internal wave fields. Coherence times of the arrival peaks are on the order of 2-3 minutes, which is somewhat less than the 5.5 minutes seen in the ATOC analysis. Early arrivals appear to be less coherent than the late arrivals. One possible explanation for the reduced coherence is that interference from higher order modes is contaminating the estimates of modes 1 through 10. Failure of four hydrophones on the 40-element VLA reduces the ability to reject interference from modes above 10 with conventional mode filtering techniques. Scattering due to upslope propagation near the NPAL receivers also affects the mode statistics.

[1] K. E. Wage, A. B. Baggeroer, and J. C. Preisig, "Modal Analysis of Broadband Acoustic Receptions at Megameter Ranges," in IEEE Sensor Array and Multichannel Signal Processing Workshop Proceedings, (Cambridge, MA), pp. 102–106, March 2000.

Optimal Array Design and Sensitivity for Mode Filtering

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The pressure field in a shallow water acoustic channel is often well characterized by a finite, discrete set of propagating normal modes. Mode filtering is defined as the estimation of the amplitudes of these normal modes from the observed pressure samples obtained at a vertical hydrophone array. Many researchers have proposed the use of mode amplitudes, or coefficients, for underwater source localization or remote sensing of oceanographic features. The success of any of these techniques relies on accurate estimates of the mode amplitudes.

The Cramer-Rao Lower Bound (CRLB) gives the smallest possible variance of any unbiased estimator. Buck, Preisig, and Wage [JASA 103(4) (1998)] derived this bound for the variance of unbiased mode filters and observed that the array geometry is the only factor controlling performance under scientists' control. The difficulty lies in designing an appropriate array for variable ocean environments. This research includes two parts.

Robust array design for a range of environments is a difficult problem. Therefore, our initial efforts examine the problem of designing the optimal array for a known environment. Specifically the optimal array is optimized to minimize the total error power of the mode estimates. Two search methods are applied to find the optimal array: exhaustive search method and steepest descent method. In several simple prototype problems, it is possible to design nonuniform arrays whose total error power is significantly lower than an uniformly spaced array. Moreover, the steepest descent method is more computationally efficient than the exhaustive search method of optimizing arrays.

Second, the sensitivity of the optimal array's performance to environmental mismatch is studied since ocean is well-known to be capricious. Two common causes of mismatch for mode filtering in the shallow water environment are tidal water depth changes and sensor location errors. The perturbation of the optimal array's performance can be analytically bounded and thus the deterioration of mode filter performance due to mismatch can be predicted.

Coherence of Pulsed Signals and Implications to Synthetic Aperture Sonar Processing

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The horizontal coherence length of acoustic signals in the ocean is on the order of several ten to several hundred wavelengths, even under favorable propagation conditions. Recent synthetic aperture sonar (SAS) processing of low-frequency active acoustic data, however, suggests that coherent integration apertures on the order of several thousand wavelengths can be achieved by using data-driven adaptive compensation techniques. This paper presents an analytical approach to understanding the interplay between adaptive compensation and medium fluctuations in SAS processing.

Our analysis concentrates on internal wave-induced signal fluctuations in a shallow-water region of the Mediterranean, corresponding to where low-frequency acoustic data was collected and successfully processed with SAS techniques. We follow Zhu and Guan's basic formalism for analyzing horizontal coherence, but extend it to two-way propagation of a pulsed signal. A Garrett-Munk-like model is used to calculate the internal wave vertical displacement correlation. The result is then used in an adiabatic normal mode calculation in conjunction with first order perturbation theory to derive the full spatio-temporal correlation of the signal field. The correlation is in turn used to simulate an ensemble of signal realizations. Each realization is then processed with synthetic aperture algorithms, including the adaptive compensation techniques.

We show that the internal wave-induced image smearing in azimuth can be overcome with adaptive compensation. The two specific techniques adopted here, phase gradient autofocus and map-drift algorithms, are both commonly used in synthetic aperture radar processing. The range of internal wave spectral level tested is typical of shallow water areas. We speculate that advanced compensation techniques can further extend the operational envelope of SAS into more adverse environments.

These results indicate that our ability to form a large synthetic aperture is not limited by the apparent natural coherence length of the ocean. The adaptive compensation techniques transfer the role of the coherence length from being the upper limit of the integration aperture to one that merely determines the size of the regions over which local phase adjustments need be made.

Synthetic Aperture Sonar Image Segmentation using the Fuzzy C-Means Clustering Algorithm

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Synthetic aperture side-scan sonar (SAS) provides an imaging modality for detecting objects on the sea floor. SAS provides an excellent tool for shallow water characterization where immobile, submerged threats would not be detected by Range-Doppler techniques. SAS images provide both an image of an object and its shadow both of which can be used in the classification and localization of potential threats. This document discusses the development of an image segmentation algorithm that was capable of segmenting (detecting) the image of an object and its acoustic shadow in the presence of reverberation noise. As a component of an autonomous deployable active sonar system, no human input was required. An unsupervised form of cluster analysis, the Fuzzy C-Means Algorithm (FCM) was used to implement the segmentation procedure. FCM is a generalization of the classical ISODATA or Hard C-Means (HCM) clustering algorithm and FCM outperformed the HCM in the segmentation of SAS images. The FCM image-segmentation algorithm was capable of dealing with variability in gray level range from one image to the next as well as the low signal-to-noise ratio (SNR) that were inherent to SAS images. The algorithm functioned in an unsupervised mode, requiring no user input and no training sets.

Suppressing Reverberation by Multipath Separation for Improved Buried Object Detection

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The detection of objects buried in the seabed by sonar is a topic receiving much interest today. Examples of applications include mine hunting, cable and pipeline surveying, and archaeological research. Detecting buried objects is difficult because sound penetrates into the seabed poorly at shallow angles, except at low frequencies. Over the past several years, low frequency, but highly directive, parametric sonars have been studied for buried object detection. However, detection is hindered by high levels of reverberation leakage resulting from the poor receive array directivity at low frequencies. This problem is further compounded by the fact that in addition to direct path reverberation components, there are also substantial reverberation components impinging on the array via surface bounce paths. These surface bounce components arise from seabed upward scattering. Since the direct and upward scattering reverberation components are often of comparable strength, this results in mutual interference that degrades the detection, localization, and classification (DCL) performance of the sonar system.

In principle, mutual interference between multipaths could be alleviated or reduced by using a vertical receiving aperture to obtain some degree of vertical directivity and hence surface path rejection. Separating direct and surface-bounce paths by conventional vertical beamforming is not effective for small apertures at low frequencies because of large beamwidths and sidelobe leakage. Application of adaptive methods is complicated by reverberation spatial non-stationarity, i.e., the multipath angle of arrival changes rapidly as a function of range because the ranges are short (few 10s of meters) and water shallow (few 10s of meters deep). More problems are the lack of signal-free training data, array calibration errors, and the vertical angular spreading caused by surface and bottom roughness and nearfield propagation effects such as wavefront curvature, resulting in non-planar wavefronts.

Using at-sea gathered SACLANTCEN vertical array acoustic data, we investigate the separation of the direct and surface bounce path reverberation and target echo components and the inherent difficulties using a variety of methods. These include conventional beamforming, generalized sidelobe cancellers, the Principal Component Inverse (PCI) method [1], and using Independent Component Analysis-like [2] ideas for separation.

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[2] A. Hyvarinen and E. Oja, “Independent Component Analysis: A Tutorial,” Neural Networks, 1999.

ULV Decomposition as Preprocessor for Nonlinear Adaptive Algorithms Used for Blind Source Separation

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It is standard practice to in application of nonlinear adaptive algorithms that are used for blind source separation (BSS) and independent component analysis (ICA) to preprocess the input data by a transformation that decorrelates and normalizes the input data. [1,2] This is also called data sphering. Sphering transformations significantly increase convergence of subsequent nonlinear algorithms. In addition the preprocessing transformation compresses the data by eliminating from data components that correspond to small singular values that usually correspond to noise or weak sources. This reduces the dimensionality of the subsequent nonlinear processing. Ideas of preprocessing can be explained by singular value decomposition (SVD). [3]

Let $\mathbf{X}$ be an mxn input data matrix where m is number of sensors or data channels and n is number of data points. SVD of $\mathbf{X}$ is:

$$\mathbf{X} = \left[\mathbf{U}_r,\ \overline{\mathbf{U}}_r\right] \Sigma \left[\mathbf{V}_r,\ \overline{\mathbf{V}}_r\right] \tag{1}$$

where r is approximate rank of $\mathbf{X}$. The preprocessing transformation $\mathbf{T}$ is:

$$\mathbf{T} = \Sigma^{\#} \left[\mathbf{U}_r,\ \overline{\mathbf{U}}_r\right]^H \tag{2}$$

where $\Sigma^{\#}$ denotes an mxn pseudo inverse matrix:

$$\Sigma^{\#} = \mathrm{Diag}\left[\sigma_1^{-1},\ \cdots\ \sigma_r^{-1}, 0\ \cdots\ 0\right] \tag{3}$$

Preprocessed data is:

$$\mathbf{TX} = \Sigma^{\#} \left[\mathbf{U}_r, 0\right]^H \left[\mathbf{U}_r, \overline{\mathbf{U}}_r\right] \Sigma \left[\mathbf{V}_r,\ \overline{\mathbf{V}}_r\right]^H = \left[\mathbf{V}_r, 0\right]^H \tag{4}$$

As Equation (1) shows SVD decorrelates and normalizes the data and reduces dimensionality of data by using only left singular vectors that correspond to r largest singular values, that is singular vectors that belong to signal subspace. In BSS and ICA these signal subspace singular vectors are further processed by nonlinear adaptive algorithms. SVD preprocessor accomplishes the objectives of decorrelation, normalization and dimensionality reduction. This preprocessing is necessary for theoretical reasons (nonlinear algorithms usually assume that input data are normalized and uncorrelated) and significantly speeds up convergence of nonlinear adaptive algorithms that are used for blind separation of statistically independent sources. SVD also provides most complete analysis of the input data matrix $\mathbf{X}$.

In spite of many advantages of SVD, it has several computational shortcomings: it is a computationally intensive, block processing algorithm that does not have updating and down dating algorithms that are needed for tracking nonstationary input data. To overcome these shortcomings we propose to apply ULV decomposition (ULVD) to BSS and ICA. ULVD has been used previously in signal processing, however it is worthwhile to exploit recent developments of ULVD methods for BSS, ICA and other adaptive signal processing applications. ULVD is well suited for processing nonstationary data because it has efficient updating and down dating algorithms that can track time-varying processes.

We present a new, efficient ULVD algorithm that is decorrelates and normalizes input data. It also reveals the rank of input data matrix. This algorithm has many applications to linear and nonlinear adaptive signal processing.

References

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The Application of M-Sequences to Bi-Static Active Sonar

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M-sequences have long served as enabling technology for numerous basic research experiments to observe the fluctuations of acoustic propagation and for many acoustical oceanographic studies including ocean acoustic tomography. They provide coherent processing gain without loss of temporal resolution. Simply stated m-sequences combine the energy of CW with the resolution of a pulse. In addition to processing gain, the M-sequences can eliminate ping stealing and allow for environmentally friendly transmissions (lower peak pressure levels). Yet, early attempts (circa 1960) to apply m-sequences to mono-static active sonar were unsuccessful. In hindsight, the inability to process Doppler and possibly an inappropriate application of a matched filter correlation are the reasons. An analysis of an application to bi-static active is presented here. The approach includes continuous transmission of long m-sequences, synchronous sampling to form a CON (Complete Ortho-Normal) data sets, direct blast removal by HCCO (Hyperspace Cancellation by Coordinate Zeroing), and a full range waveform Doppler search. Ultra-fast Hadamard Transforms speed up the direct "waveform - pulse" m-sequence pulse compression and the inverse "pulse - waveform" transform and thereby allow timely execution of the intensive computational burden. A similar approach can be used to process and separate simultaneous m-sequence transmissions from more than one source. If sets of nearly orthogonal sequences are use, the residual arrivals can be eliminated by HCCO processing resulting in almost perfect orthogonality. We consider activating a distributed passive array consisting of n nodes with a set of m active sources each transmitting a "quasi - orthogonal" m-sequence. The result is nxm independent bi-static sonars operating in the same space. The fusion of such data sets hold promise for wide area coverage with finite and possibly small number of transmits and receive nodes.

Multiple Ping Processing for Active Towed Array Sonar Systems

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Active towed array sonar systems consist of a transmitter and a receiving towed array which both are operated at the same adjustable depth. The main characteristics of these systems are the large detection ranges and the applicability to shallow as well to deep water situations.

Experiences using active towed array sonar systems in low and medium frequency ranges, at high and low towing speeds, showed that the detection of moving targets in shallow water with areas of high clutter proved a challenging task.

Reverberation is a severe problem in high target like clutter areas for both CW and FM pulse processing. To resolve this situation, we propose to take advantage of the ping history for CW and FM pulses. The proposed algorithms process data from different pings incoherently because of the insufficient coherence from ping to ping which is due to the complex sound propagation conditions.

In radar applications, space-time adaptive processing (STAP) techniques are used for detection of slowly moving objects by airborne radar systems. In contrast to sonar applications, coherent pulse trains are usually used in radar STAP systems. Then, a Doppler shift can be measured by coherent analysis of several pings. The differences between radar and sonar application prevent a direct transfer of radar techniques to sonar. In this paper we will adapt radar STAP processing techniques to improve the moving target detection for CW pulses. For FM pulse processing this is not possible. Nevertheless, for FM pulses we can use the positional change of moving targets during several pings to discriminate between moving targets and clutter.

In this contribution the data model is described. Then, we present the CW and FM processing algorithms. Finally, we apply the algorithms to simulated and measured sonar data and discuss the result. Data presented were recorded by the low and medium frequency active towed array sonar systems of STN ATLAS Elektronik GmbH.

Broadband active sonar detections using the time-reversal operator

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The eigenvalue decomposition of the time reversal operator (DORT) is a single-frequency technique that isolates scatterers using a system of multiple sources and receivers [C. Prada, S. Manneville, D. Spoliansky and M. Fink, J. Acoust. Soc. Am. 99, 2067-2076 (1996)]. Through simulation and experiment, this technique has been shown to work well for narrowband, fixed source- receiver geometries, but it has not been applied to bistatic sonar performance in reverberation-limited environments.

To assess the robustness of the DORT technique for use with broadband active sonars, a coherent modeling capability is developed to simulate performance in a realistic oceanic environment. Using MATLAB, models were developed to generate coherent broad-band signals using RAM PE. Currently, the models are able to generate coherent, broad-band acoustic fields, target echoes and reverberation time series from a single radial for monostatic or vertically bistatic source/receiver geometries.

Two simulations are shown to elucidate the resolution and performance of a DORT system in a reverberant, shallow-water environment. The system consists of vertical line arrays of sources and receivers in monostatic and vertically bistatic configurations.

First, two single-frequency resolution problems are considered:the ability to discriminate two scatterers in the water column and the ability to detect a single scatterer in the presence of bottom clutter. The performance of the DORT system is shown relative to array configuration, location of scatterer in the water column, and the level and type of bottom clutter.

Second, the application of this technique to a broad-band active sonar system is demonstrated. A broad-band signal containing reverberation, noise, and a target return is simulated. Exploiting the broad bandwidth to isolate point targets or clutter in range, the returns of interest are then time windowed, Fourier transformed, and DORT processing is applied. Eigenvectors of the time reversal operator are then back-propagated through the Green function for the medium to create images of the located scatterers. The method is shown to isolate the target in a single eigenvalue/eigenvector in the presence of distributed reverberation. Due to multipath fading throughout the water column, the target was isolated into different eigenvalues at different frequencies.

A Split/Merge Image Segmentation Algorithm for Detection of Targets in Range-Doppler Maps

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This paper presents an image segmentation region-growing algorithm based on the split and merge approach to detect targets in range-Doppler maps at low signal to noise ratios. Edge intensity information is used in the predicates for splitting and merging regions and the rationale behind the predicate selection is explored. The details of each predicate are explained as well as the importance of each predicate to the overall processing scheme. An important contribution of the work is the automatic selection of the predicate parameters that enables on-line processing. The essential point of automatic selection is the fact that no user intervention is required. In addition, this technique presents a new approach to detection/classification of structured signals since no background estimation is required. Examples from preliminary investigations are shown that indicate the viability of the proposed approach and recommendations are given for further exploration.

Direction Finding of Narrowband Signals Based on the Maximum Likelihood-Expectation Maximization Technique

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A longstanding problem in submarine sonar development has been the accurate tracking of a target as it approaches and crosses interferers, i.e., when there are more than one target in a single sonar beam. The classic Jarvis direction finder and tracker assumes only one signal in the beam of interest, which in cases of strong interference can completely skew the tracker estimate. The purpose of this work is to implement a direction finding technique which will account for multiple targets and correctly resolve their corresponding signals. To this end, we have applied the Maximum Likelihood Expectation Maximization (ML-EM) technique to the problem of multiple narrowband signal direction finding with linear sonar arrays. The technique provides estimates which are unbiased and works at lower Signal-to-Noise Ratios (SNR) with fewer data samples than other methods. Assuming a maximum of two superimposed signals the calculations were performed in beamspace for two distinct signal models: random Gaussian signals and deterministic signals of known waveform but unknown amplitude, both in Gaussian noise. At the core of the method is the extraction of information on the complete data from incomplete observations and the separation of the signal parameter estimation into two parallel operations. We tested the limits of the ML-EM method by using only one data sample in order to accommodate dynamic targets. Performance curves were derived and compared favorably against the calculated ML Cramèr-Rao lower bounds. The use of one sample gives reasonably good values of the bearings, especially at high SNR, but impairs the estimation of the signal amplitude, which is also assumed to be an unknown parameter. The two algorithms perform equally well in the uncomplicated case of a single incoming signal, but the deterministic surpasses the random signal algorithm in the case of two unknown narrowband signals by several dB. The background noise was always assumed to be white. We also attempted to develop an empirical criterion for the important but difficult problem of the determination of the number of signals, which is assumed known in our implementation of the ML-EM algorithm.

A New Approach to Range-Focused Beamforming

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The estimation of the two-dimensional position (range and bearing) of a passive narrowband source in the near-field of an array has been the subject of much research since the seminal paper by Hahn in 1975. Hahn presented a range-focused beamformer (RFB) solution for position estimation of a single source (Bartlett beamformer). In 1977, Carter showed that this beamformer was the maximum-likelihood 2-D position estimator. Huang and Barkat, using exact near-field time delays, extended the maximum-likelihood position estimator to multiple sources in 1991 and applied it using the alternating projection technique developed by Ziskind and Wax. This technique allowed Huang and Barkat to search for both the number of sources and their positions, guaranteeing a globally optimum solution.

The purpose of this presentation is to first demonstrate the problems associated with 2-D position estimation of multiple sources using a conventional RFB with a long towed array. We will show how the range-defocused sidelobes from a strong source can not only mask weaker sources but can also create false detections due to their high sidelobe levels (almost as high as -5 dB). Then we will illustrate the alternating projection technique as applied to range focused localization of four sources of decreasing SNR, using a modified version of the technique presented by Huang and Barkat. The approach is quite simple. A conventional RFB is used to form beams over a coarse grid of range-bearing points and the grid point with the largest beamformer output is used to define the center of a fine grid of range-bearing points. The fine grid point with the maximum beamformer output is assumed to be the position estimate for the first source. To localize the second strongest source, the steering vector associated with the estimated position of the strongest source is projected out of the element-level covariance matrix and the course to fine grid RFB maximization is repeated using the reduced covariance matrix. The procedure is repeated until the maximized beamformer output is below a threshold, where it is then assumed that no more sources are present. The technique will be demonstrated using simulated data in white noise.

Passive Sonar Interference Suppression using Matrix Filters

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We consider the problem of processing passive sonar array data containing contributions from a loud surface source and a quiet submerged source in a shallow-water environment. The goal is to remove or attenuate the loud-source contribution while preserving the quiet-source contribution for further processing. A linear filtering operation on the sensor outputs can be expressed as the multiplication of a matrix (called a matrix filter) times the vector of sensor outputs. Matrix filters are designed by defining an appropriate pass band and stop band in range-depth space and solving a convex optimization problem. Data components in the pass band of a matrix filter are preserved with a minimal, specified amount of distortion, while components in the stop band are attenuated as much as possible. In this talk we describe the formulation and design of matrix filters, and give simulation examples to show how these filters are used to remove the interference from a loud surface source.

Information Theory for Source Localization

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Source localization has traditionally been considered as an estimation problem in which the goal is to use the observed pressure field and an acoustic propagation model to estimate an unknown source location with the smallest possible variance [Baggeroer, *et al.*, JASA, 1988]. In many environments, reducing the variance of the estimate is less important than sidelobe suppression. This motivates an alternative perspective in which the search region is divided into a grid whose cell size is dictated by logistical constraints, and the localization algorithm attempts to assign the source to the correct cell with the minimum probability of error. In this approach, the source localization problem can be considered as a communication problem. Specifically, the source is (perhaps unwillingly) transmitting a message whose content is the grid cell it is located in, and the receiver wishes to use the pressure data observed at the array to decode this message with minimal probability of error.

In order to achieve an arbitrarily small probability of error in the grid cell assignment of the source, the Channel Coding Theorem indicates that the mutual information between the pressure observations and the source location must be equal to or greater than the entropy of the source location. The Gaussian channel model provides an upper bound on this mutual information as a function of SNR, which we evaluate in a typical shallow water scenario for both the spatially isotropic and Kuperman-Ingenito noise models. These results provide necessary, but not sufficient conditions, on the average per hydrophone SNR required to achieve a desired range resolution with arbitrarily small probability of choosing a sidelobe of the ambiguity surface. Alternatively, the results may also be formulated to provide a lower bound on the range resolution that may be achieved with arbitrarily small probability of error in choosing the correct source location.

Wednesday October 3, 2001		Thursday October 4, 2001		Friday October 5, 2001	
		8:00-9:45	Session B Laurel	8:00-9:45	Session F Laurel
		9:45-10:15	Break Laurel	9:45-10:15	Break Laurel
		10:15-12:00	Session C Laurel	10:15-12:00	Session G Laurel
		12:00-1:00	Lunch Whisp. Pines	12:00-1:00	Lunch Whisp. Pines
		1:00-2:45	Session D Laurel		
		2:45-3:15	Break Laurel		
		3:15-5:00	Session E Laurel		
5:00-6:00	Welcome Reception				
6:00-8:00	Raytheon Dinner Whisp. Pines	6:00-8:00	Dinner Whisp. Pines		
8:00-9:30	Session A Laurel	8:00-?	SOB Session Laurel		