A Book of Abstracts for the

2009 Underwater Acoustic Signal Processing Workshop

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UASP 2009

Welcome to the 2009 IEEE workshop on Underwater Acoustic Signal Processing. This year the special session, organized by Dr. Phil Schniter will be on Underwater Acoustics Communications Fields Experiment Results.

The organizing committee would like to thank and acknowledge the continued support of the Office of Naval Research, the IEEE Oceanic Engineering Society for their sponsorship of the Wednesday evening dinner, and thanks Jerry Bradshaw for his efforts in arranging for Raytheon Systems Company to sponsor our Thursday evening dinner. We are also honored to present this year's UASP Award to Prof. Arthur B. Baggeroer of MIT.

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The 2009 UASP Award is Presented to Prof. Arthur B. Baggeroer

for sustained leadership in the theory and experimental practice of underwater acoustic signal processing

Arthur B. Baggeroer graduated from Purdue University in 1963 with a B.S.E.E. degree and from the Massachusetts Institute of Technology in 1968 with an Sc.D. degree. He immediately joined the faculty of MIT, and was promoted to full professor in 1980. Arthur is the Ford Professor of Engineering, and holds the prestigious Secretary of the Navy/Chief of Naval Operations Chair in Oceanographic Science. Arthur has had a long affiliation with both MIT Lincoln Laboratory and the Woods Hole Oceanographic Institution, in addition to visiting appointments at the NATO Undersea Research Center (formerly SACLANT) and the Scripps Institute of Oceanography. Arthur generously gives his time and expertise in the national interest, serving on many advisory panels for the Navy and the National Research Council. The National Academy of Engineering elected Arthur a member in 1995, and he is a fellow of both the IEEE and the Acoustical Society of America. The ASA awarded Arthur the Rayleigh-Helmholtz Silver Medal in 2003, and the IEEE Oceanic Engineering Society choose him for its Distinguished Technical Achievement Award in 1991.

Arthur's accomplishments include an impressive blend of theoretical advances and important experiments. Arthur was a leader in bringing "high resolution" spectral estimation techniques to adaptive array processing. Early in his career, he formulated important performance bounds for the Maximum Entropy Method spectral estimator. Arthur was a pioneer in developing Matched Field Processing, a topic summarized in his 1993 overview paper co-authored with Mikhalevsky and Kuperman. His chapter on "Sonar Signal Processing" in the 1978 compendium *Applications of Digital Signal Processing* is a solid overview of the field that remains relevant more than three decades later. His interest in performance bounds continued in his collaboration with Kuperman and Schmidt to formulate and calculate the Cramer-Rao Lower Bound on passive sonar performance in realistic propagation environments, published in the *Journal of the Acoustical Society of America* in 1988. Most recently, his work with Wen Xu and Christ Richmond to applied Bayesian bounds to realistic ocean environments. Even following his alleged retirement from the MIT faculty, he remains active on the forefront of theoretical array processing research, collaborating with Raj Rao on new applications of Stochastic Eigenanalysis techniques from mathematics to obtain important closed form results on the performance of adaptive array processing algorithms.

Arthur also made major contributions to the experimental side of underwater acoustics and acoustical oceanography, serving as Chief Scientist on more than fifteen field experiments or cruises. He collaborated closely with Walter Munk on the Heard Island experiment, a pioneering experiment in long range acoustical propagation and thermography. This collaboration continued with subsequent acoustical thermography experiments in the Pacific and Arctic Oceans. Arthur played a leadership role in the Digital Acoustic Telemetry System (DATS) program, the first such system to employ MFSK modulation. He also led several field camps on the Arctic ice cap, resulting in major advances to our understanding of acoustical propagation in the Arctic Ocean, as well as the seismic properties of the ocean floor.

Finally, Arthur has mentored many graduate students who have themselves gone on to be leaders in the fields of array processing, underwater acoustic propagation and acoustical oceanography. When considering the combined impact of his technical contributions, his service to both the Navy and professional societies, and his advising of numerous graduate students, Professor Arthur Baggeroer is truly a fitting recipient of the 2009 IEEE Underwater Acoustic Signal Processing Workshop Award.

Schedule at a glance

Wednesday **October 14, 2009**		**Thursday** **October 15, 2009**		**Friday** **October 16, 2009**	
		8:15–9:55	Session B UW ACOMMS I Laurel	8:15–9:30	Session G UW ACOMMS III Laurel
		9:55–10:20	Break Laurel	9:30–9:55	Break Laurel
		10:20–12:00	Session C UW ACOMMS II Laurel	9:55–10:45	Session H Passive Laurel
		12:00–1:00	Lunch Whisp. Pines	10:45–12:00	Session I Array I Laurel
		1:00–2:15	Session D Active Laurel	12:00–1:00	Lunch Whisp. Pines
		2:15–3:05	Session E Passive I Laurel	1:00–2:15	Session J Array II Laurel
		3:05–3:30	Break Laurel		
		3:30–5:10	Session F Prop.-Based Laurel		
5:00–6:00	Welcome Reception Whisp. Pines				
6:00–8:00	Raytheon Dinner Whisp. Pines	6:00–8:00	OES Dinner Whisp. Pines		
8:00–9:30	Session A Plenary Laurel	8:00–?	SOB Session Laurel		

Sessions: Titles and presenters

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

Special Session Plenary Talk

A–1 *Underwater Acoustic Communications: Working at the Intersection of Physics, Signal Processing, and Communications Theory*,
James C. Preisig, Woods Hole Oceanographic Institution

Session B: Thursday Morning, 8:15am–9:55am

Special Session I: Underwater Acoustics Communications Fields Experiment Results

B–1 *Taxonomy of Underwater Communication for US Navy Needs*,
Arthur B. Baggeroer, Massachusetts Institute of Technology

B–2 *Recent results on underwater acoustic communication research*,
T. C. Yang, Naval Research Lab

B–3 *Measurement-based simulation of underwater acoustic communication channels*,
Paul Van Walree, Norwegian Defence Research Establishment FFI

B–4 *Training Sequence Synthesis, Channel Estimation and Symbol Detection for MIMO Underwater Acoustic Communications*,
Jian Li, University of Florida

Session C: Thursday Morning, 10:20am–12:00pm

Special Session II: Underwater Acoustics Communications Fields Experiment Results

C–1 *Signal Processing for Underwater Acoustic MIMO OFDM*,
Milica Stojanovic, Northeastern University

C–2 *Results of OFDM Transmissions in the Kauai Acomms MURI 2008 Experiment*,
Tolga Duman, Arizona State University

C–3 *Receiver Comparisons on an OFDM Design for Doppler Spread Channels*,
Shengli Zhou, University of Connecticut

C–4 *On Optimal Resampling for OFDM Signaling in Doubly Selective Underwater Acoustic Channels*,
Srinivas Yerramalli, University of Southern California

Session D: Thursday Afternoon, 1:00pm–2:15pm

Active Sonar

D–1 *Detection-Threshold Approximation for non-Gaussian Backgrounds*,
Douglas Abraham, CausaSci LLC

D–2 *A Generalized Linear Filtering Approach for Sonar Receivers*,
Nabin Sharma, University of Massachusetts Dartmouth

D–3 *Robust Sonar Identification and Authentication in Simulated Shallow Water*,
Bijan Mobasseri, Villanova University

Session E: Thursday Afternoon, 2:15pm–3:05pm

Passive Sonar I

E–1 *Multiband DEMON Estimation with Compensation for Channel-Induced Modulation Distortions*,
Ivars Kirsteins, Naval Undersea Warfare Center

E–2 *Minimum Hellinger Distance Classification of Passive Underwater Acoustic Signals*,
Brett Bissinger, Penn State University

Session F: Thursday Afternoon, 3:30pm–5:10pm

Propagation-based Sonar Processing

F–1 *Time-Frequency Approach and Approximation to Range-Dependent Pulse Propagation*,
Patrick Loughlin, University of Pittsburgh

F–2 *The propagation of noise fields in a dispersive medium: a phase-space approach*,
Leon Cohen, City University of New York

F–3 *Sequential and iterative Bayesian methods for acoustic feature estimation*,
Zoi-Heleni Michalopoulou, New Jersey Institute of Technology

F–4 *Statistics-based classification of passive sonar signals*,
R. Lee Culver, Penn State University

Session G: Friday Morning, 8:15am–9:30am

Special Session III: Underwater Acoustics Communications Fields Experiment Results

G–1 *Application of the Turbo principle to Underwater Acoustic Communications: Receiver Architectures and Experimental Results*,
Jun Won Choi, University of Illinois at Urbana-Champaign

G–2 *Markov Chain Monte Carlo Detectors for underwater acoustic channels*,
Rong-Rong Chen, University of Utah

G–3 *High rate time reversal communications for underwater MIMO channels*,
Aijun Song, University of Delaware

Session H: Friday Morning, 9:55am–10:45am

Passive Sonar II

H–1 *Maximum entropy probability density functions and recursive Bayesian state estimation*,
Colin Jemmott, Penn State University

H–2 *Detection and classification of odontocetes - algorithms and architecture for autonomous, deployable system*,
Paul Hursky, Heat, Light, and Sound Research Inc.

Session I: Friday Morning, 10:45am–12:00pm

Array Processing I

I–1 *Performance Prediction Analysis of Depth Discrimination Algorithm*,
Shawn Kraut, MIT Lincoln Laboratory

I–2 *A Globally Convergent CML Algorithm*,
Neil Malloy, Multisensor Science LLC

I–3 *Application of Multitaper Spectral Estimation Methods For Adaptive Beamforming*,
Joseph Schwarzwalder, Argon ST, Inc.

Session J: Friday Afternoon, 1:00pm–2:15pm

Array Processing II

J–1 *Spatial Compressive Sensing for Passive Sonar Arrays*,
Igal Bilik, University of Massachusetts Dartmouth

J–2 *Exploiting Towed Array Dynamics for Passive Field Directionality Mapping*,
Jeffrey Rogers, Duke University

J–3 *Passive Ranging Capability of Multi-Module Arrays in Underwater Acoustic Environments*,
Hongya Ge, New Jersey Institute of Technology

Abstract Listings

Underwater Acoustic Communications: Working at the Intersection of Physics, Signal Processing, and Communications Theory

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The underwater environment is widely regarded as one of the most difficult communication channels. Underwater acoustic communications systems are challenged by the characteristics of acoustic propagation through the underwater environment in which they operate. There are a wide range of physical processes that impact underwater acoustic communications and the relative importance of these processes are different in different environments. The characteristics of the environment can impact everything from the determination of the optimal topology for distributing network nodes to efficient transmit signal selection and receiver algorithm design. While a large number of the pertinent questions regarding these issues are still open research topics, research over the past decade has certainly lent insights into their answers.

A particularly interesting and challenging environment is that in which surface scattered signals form a significant portion of the received signal. Rapid channel fluctuations and a extended delay spread of the channel can combine to make it difficult for coherent communications algorithms to track the channel accurately enough to enable reliable high-rate communications. The rapid channel fluctuations limit the averaging time that most simple algorithms can employ to estimate important channel parameters which often results in ill-condition estimation or adaptation problems. Effective techniques for addressing this problem include slowing down the apparent rate of channel fluctuation or reducing the number of free parameters that must be adjusted to effectively track the channel. The consideration and/or exploitation of relevant channel physics can be used to realize improvements using these techniques as demonstrated with several examples in this talk.

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Taxonomy of Underwater Communication for US Navy Needs

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For over 40 years many US Navy programs have required acoustic communications (ACOMMS) for their concept of operations and every one has led to an individual effort and/or organization. There is no catalog of systems with specified performance for applications and environments. Part of this is because of the diverse number of acoustic channels from low bandwidth, long range to high bandwidth, short range problems. Nevertheless there are several application areas where we can provide a taxonomy:

1. Distributed systems - communication among Unmanned Underwater Vehicle (UUV) platforms and central nodes,
2. Data exfiltration for remote, passive surveillance,
3. Communication from submarines at speed and depth,
4. Command, Control, Communications, Intelligence, Surveillance and Reconnaissance (C^3ISR) for UUV's and mines,
5. Reach back for Special Operations Forces and intrasquad communication.

The most challenging problem is to communicate covertly. This implies operating within a limited transmitted power. Too much will trip the acoustic intercept receiver on direct energy detect while too little will not permit decoding. All this is a strong function of channel complexity, coding, and bandwidth. Mimicing has also been considered, but a well trained acoustic intelligence rider is hard to fool.

This talk will cover the the various applications of ACOMMS and provide some perspective on their use and the problems encountered.

Recent results on underwater acoustic communication research

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The Naval Research Laboratory has conducted many (> 10) underwater acoustic communication (ACOMM) experiments at various oceans around the world over the last ten years. The objectives are to study the environmental impacts on underwater acoustic communications, develop algorithms to mitigate the environmental impact, and develop performance modeling capability. The research conducted covers mid-frequency (2–5 kHz) ACOMMs, high-frequency (15–25 kHz) ACOMMS, covert ACOMMS, and MIMO ACOMMs. This paper will review the approach and summarize recent results based on experimental analysis and theoretical studies.

[Work supported by the US Office of Naval Research.]

Measurement-based simulation of underwater acoustic communication channels

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A channel simulator is being developed for underwater acoustic communications. The simulator consists of a tapped delay line and features several modes of operation, where the generation of realistic tap gains is the primary challenge of each mode. The first mode is based on in situ measurements of the time-varying impulse response, and reproduces the measured channel exactly. The second mode is stochastic and reproduces statistical properties of measured channels. A third and future mode is based on acoustic modeling. The channel replay mode is highly reliable so long as the measurements are reliable, which requires a channel probe scheme that i) delivers a high signal-to-noise ratio; ii) samples the impulse response fast enough to track its changes; iii) captures the entire delay spread in a single impulse response measurement. The stochastic mode puts the same demands on the channel measurements, but additionally requires compensation for Doppler shifts due to relative TX/RX motion. Without such compensation, the measured Doppler variance is entirely, and incorrectly, attributed to multipath fading. The simulator assumes wide-sense stationary scattering, but allows for correlation between taps, which is a feature of many in situ measurements. The presentation shows results from channel measurements and simulation in the Baltic Sea and on the Norwegian continental shelf. The required steps between measurement and simulation are detailed and illustrated. Different channels are examined, and the fidelity and limitations of the channel simulator in its present form are discussed.

Training Sequence Synthesis, Channel Estimation and Symbol Detection for MIMO Underwater Acoustic Communications

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High data rate underwater acoustic communications (UAC) can be realized by using multi-input multi-output (MIMO) systems. To overcome the challenges of UAC channels being time-varying and frequency-selective with long memory, effective and efficient training sequences are desired. We present herein a cyclic approach for synthesizing MIMO training sequence sets with good auto- and cross-correlation properties. Moreover, we use an iterative adaptive approach (IAA) coupled with the Bayesian information criterion (BIC) for sparse channel estimation and tracking. Furthermore, for payload symbol detection, a relaxation based minimum mean-squared error detection scheme is presented and its extension to incorporate decoding is considered. SPACE'08 experimental results are provided to validate the proposed MIMO scheme.

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Signal Processing for Underwater Acoustic MIMO OFDM

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Long acoustic multipath limits the applicability of MIMO OFDM channel estimation methods that require matrix inversion of size Mt x L, where Mt is the number of transmit elements and L is the multipath spread measured in 1/B, the bandwidth inverse. To overcome this problem, sparse nature of the channel is exploited in an algorithm based on a compact signal representation which leads to two forms of adaptive implementation: one that requires matrix inversion but of a reduced size, and another that completely eliminates it. Channel estimation is coupled with phase tracking and prediction to enable decision-directed operation in the presence of non-uniform Doppler distortion, which in turn provides improved performance and reduced overhead. Performance is demonstrated using real data gathered during the RACE 08 and SPACE 08 experiments.

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Results of OFDM Transmissions in the Kauai Acomms MURI 2008 Experiment

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We focus on results obtained by orthogonal frequency division multiplexing (OFDM) transmissions during the Kauai Acomms MURI 2008 (KAM 08) experiment, which was performed in shallow water off the western coast of Kauai, Hawaii, in June 2008. We consider underwater acoustic OFDM signals operating in the band spanning 12 kHz to 20 kHz, for a fixed-source fixed-destination 4-km link, as well as for a towed-source fixed-destination 1.5-km link. We discuss the effectiveness of the existing techniques for time-frequency synchronization and Doppler-shift compensation, and we apply a recently-developed intercarrier interference (ICI) mitigation technique to remedy the time variations in the channel. We present results for both single-input single-output (SISO) and multi-input multi-output (MIMO) OFDM transmissions, showing that nearly error-free communications could be provided in the considered experimental environment, in some cases by combining received signals from multiple hydrophones.

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Receiver Comparisons on an OFDM Design for Doppler Spread Channels

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Underwater acoustic channels induce large Doppler drifts that render intercarrier interference (ICI) for OFDM transmissions. Assuming that after proper Doppler compensation the residual ICI is limited to only direct neighbors, we propose an OFDM signal design that decouples channel estimation and data demodulation. We investigate five receivers that are categorized into three groups: (i) two receivers that ignore the residual ICI, (ii) two receivers that are based on a basis expansion model (BEM) and pursue channel estimation independently along each basis, and (iii) one receiver that is based on a path-based model. The receiver performance is compared based on data from the GLINT experiment conducted south of the island Elba in the Mediterranean, July 2008, and the SPACE experiment conducted off the coast of Martha's Vineyard, Massachusetts, October 2008. The receiver based on the path-based model and a basis pursuit (BP) algorithm achieves the best performance, followed by the ICI-ignorant and BEM versions of BP. The least-squares channel estimation performs the worst, especially in combination with the BEM. The BEM based receivers are often inferior to the ICI-ignorant counterparts, except for conditions with very large Doppler spread. This implies that there exists a trade-off between ICI compensation and the estimation accuracy of the much increased number of BEM parameters. On the contrary, the path-based channel model facilitates ICI compensation without increasing the number of model parameters, by exploiting the sparse representation in the joint delay-Doppler domain.

[This work is supported by ONR grants N00014-07-1-0805, N00014-09-1-0704 (PECASE), and the NSF grant CNS-0721834.]

On Optimal Resampling for OFDM Signaling in Doubly Selective Underwater Acoustic Channels

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Efficient communication between mobile underwater nodes via Orthogonal Frequency Division Multiplexing (OFDM) is considered. Underwater acoustic channels, characterized by long delay spreads, sparsity and time variation, are better modeled by considering a triplet of parameters, i.e. gain, delay and Doppler shift, for each tap, which is a generalization of the commonly used single common Doppler shift model. The path dependent Doppler shifts and wide band signaling introduce significant time variation in the channel, destroying carrier orthogonality and introducing inter-carrier interference in an OFDM system.

This work examines the choice of resampling factor for the case of distinct Doppler shifts on each path. It is shown that resampling is equivalent to filtering the incoming signal with a set of filters matched to the resampling parameter. A bound on the performance of data detection schemes is derived using the Hammersley-Chapman-Robbins (HCR) bound, which when optimized, yields the best resampling parameter that minimizes the error in data detection. An approximate analysis of the expression for the HCR bound suggests that when one of the channel taps dominate in magnitude, the optimal resampling parameter is near the Doppler shift on this tap. When several taps of similar magnitude occur, this factor is very close to the channel gain weighted combination of Doppler shifts on each tap. Furthermore, it is shown that the sufficient statistics for data detection after resampling as can also be derived by approximating the Doppler shifts on each channel tap with a common Doppler shift equal to the resampling factor. This interpretation of resampling is used to derive practical schemes for jointly estimating the resampling factor and channel parameters with only modest information loss.

Detection-Threshold Approximation for non-Gaussian Backgrounds

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The detection threshold (DT) term in the sonar equation describes the signal-to-noise ratio (SNR) required to achieve a specified probability of detection (P_d) for a given probability of false alarm (P_{fa}). Direct evaluation of DT requires obtaining the detector threshold (h) as a function of P_{fa} and then using h while inverting the often complicated relationship between SNR and P_d. Easily evaluated approximations to DT exist in the form of Albersheim's seminal work [1] and the refinements of Hmam [2]. However, these approximations require the background additive noise or reverberation to be Gaussian (e.g., a Rayleigh-distributed envelope). While these approximations are extremely accurate for Gaussian backgrounds, they are erroneously low when the background has a heavy-tailed probability density function. In this presentation it is shown that by obtaining h appropriately from the non-Gaussian background while approximating P_d for a target in the non-Gaussian background by that for a Gaussian background, the easily evaluated approximations to DT derived by Hmam [2] extend to non-Gaussian backgrounds with minimal loss in accuracy. Both fluctuating and non-fluctuating targets are considered in Weibull- and K-distributed backgrounds. While the P_d approximation for fluctuating targets is very accurate (errors approximately one tenth of a dB) for a wide range of P_d and P_{fa} values, it is coarser for non-fluctuating targets, necessitating a correction factor to the DT approximations which reigns in the errors to mostly less than half a dB.

These results should be useful in predicting the performance of sonar or radar systems when the background is non-Gaussian. The idea of approximating the PDF of a target in a non-Gaussian background by that for a Gaussian background may be particularly useful in areas such as target tracking, classification or image segmentation.

[1] W.J. Albersheim, "A closed-form approximation to Robertson's detection characteristics," *Proceedings of the IEEE*, vol. 69, no. 7, pp. 839–839, 1981.

[2] H. Hmam, "Approximating the SNR value in detection problems," *IEEE Transactions Aerospace and Electronic Systems*, vol. 39, no. 4, pp. 1446–1452, 2003.

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A Generalized Linear Filtering Approach for Sonar Receivers

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Active sonar systems operating in realistic environments face the challenge of simultaneously detecting and resolving closely spaced targets. Receiver signal-to-noise-ratio (SNR) gain and main-lobe width of the receiver response to the transmitted signal limit the detection and resolution capability of an active sonar receiver, respectively. A matched filter is commonly used for target detection to exploit its optimal detection property. Theoretically, an inverse filter achieves optimal resolution performance for multiple targets. In practice, the inverse filter is difficult to implement and requires unrealistically high SNR to perform well. For an active sonar receiver operating in a cluttered environment with closely spaced targets, a matched filter might not resolve some targets and an inverse filter might not detect some targets. For such environments, it is desirable to have a receiver that can achieve a tradeoff between SNR gain (detection) and main-lobe width (resolution).

Senmoto and Childers [1] proposed an approach to vary the resolution capability of a detector by multiplying the inverse filter frequency response by a Gaussian taper to bandlimit the inverse filter. In the time domain, this produces narrow Gaussian pulses at the true target delays. The tradeoff between the detection and resolution is achieved by changing the variance of the Gaussian taper.

We propose a generalized linear filter (GLF) that combines the matched and inverse filter properties in a controlled manner to achieve a smooth tradeoff between detection and resolution. The frequency responses of the matched and inverse filter are combined via a tuning parameter to realize the frequency response of the GLF. The inverse filter is the resolution extreme and the matched filter is the detection extreme of the GLF. Selecting a value of the tuning parameter between these two extremes provides a way to trade detection for resolution or vice versa.

Simulations comparing the matched and inverse filter outputs with the GLF output are presented to demonstrate how the GLF can detect and resolve some targets that are not detected and resolved by the matched and inverse filters. In addition, the detection and resolution performances of the GLF and the filter proposed by Senmoto and Childers are compared. The study shows that the GLF is better receiver than a active sonar receiver based on Senmoto and Childers filter.

[1] S. Senmoto and D. G. Childers, "Signal resolution via digital inverse filtering," *IEEE Trans. Aerosp. Electron. Syst.*, vol. 8, pp. 633–640, Sept. 1972.

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Robust Sonar Identification and Authentication in Simulated Shallow Water

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In a previous work the idea of embedding a secure digital watermark in the time-frequency representation of sonar was proposed [1]. The watermark, if successfully extracted, can be used for source authentication based on originating vessels, mission and a variety of other metadata. In this work we have shown that our spread spectrum watermarking can be designed to be robust in undersea environments under realistic conditions as modeled by the Sonar Simulation Toolset (SST). For the sonar pulse we have used an exponential FM pulse and embedded it with a pseudorandom sequence. Watermark embedding follows the spread spectrum model. The pulse is windowed in time and each window is transformed by DCT(discrete cosine transform). Selected DCT coefficients are additively modified by a spread watermark sequence. Spreading code is secret and shared only with authorized receivers. The pulse is then inverse transformed to generate the watermarked sonar. Coefficient selection can also be controlled by a key that is shared between the transmitter and authorized receivers. A correlation receiver despreads the watermark using the spreading key. Unauthorized watermark detection will fail without access to this key even if the presence of watermark in the sonar is suspected. Detection is "blind" as the original unmarked sonar is not needed. Detection robustness can be improved by integrating correlation plots across all windows, and more effectively, over multiple sonar pings. The key issue is watermark detectability in light of spreading and volume attenuation loss as well as backscattering, multipaths and reverberation.

Using SST, a shallow water channel is simulated by placing the transmitter and receiver 10 meters below the surface, separated by 2000 meters. The ocean was set to 200 meters deep with a surface wind speed of 10 meters/sec. The volume attenuation coefficient was set to -0.1 dB per kilometer, the speed of sound was kept constant at 1500 meters/sec and the number of Eigenrays was set to 5. It is determined that the transmission loss for this path is 67 dB. An exponential FM pulse of 1 second duration is swept between 100 Hz and 1000 Hz and sampled at 5000 samples per second. The pulse is divided into 100 nonoverlapping windows, DCT transformed and watermarked. The choice of which frequencies to watermark depends on the frequency response of the channel. The frequency response of reverberant and multipaths channels are highly selective, with frequent peaks and notches. From the frequency response curve a set of 10 "good" frequencies per window were chosen. The factor critical to watermark detection is the signal to watermark ratio (SWR) at embedding stage. It is desired to have the highest SWR (weakest watermark) for a given detection probability. Two propagation modes in shallow water are considered. The first mode is a direct path with only spreading and absorption. The second mode adds two surface and two bottom bounces. Plots of the miss probability and false positives vs. number of pings are estimated. With no multipaths an SWR of 30dB and 15 pings are needed to achieve 90% correct classification rate. Interestingly, similar performance levels are achieved in a multipaths environment if embedding strategy is matched to channels's frequency response.

[1] B. G. Mobasseri, R. S. Lynch, G. C. Carter, "Information Embedding in Sonar for Authentication and Identification," IEEE Aerospace Conference, 1-8 March 2008 Page(s):1 - 5

[This work is supported by a grant from the Office of Naval Research and the Naval Undersea Warfare Center, DIVNPT on Grant Number N66604-09-M-0050. Contributions of Susram Kanamarlapudi, Andrew Richardson and Kevin Hinds to this research are greatly appreciated.]

Multiband DEMON Estimation with Compensation for Channel-Induced Modulation Distortions

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The radiated noise from ship propellers has quasi-periodic amplitude modulation-like characteristics that are a function of the shaft rotational rate and the number of blades. The phenomena thought to be responsible for the noise is cavitation of the propeller blade surfaces and edges modulated by wake in-flow and pressure inhomogeneities. Estimation of the shaft and blade rates and modulation waveform are important problems in passive sonar since the ship type can be inferred from these measurements [1].

Propeller sounds are commonly modeled as a stationary broadband random process (carrier) that is amplitude modulated by a periodic positive waveform. The optimal estimator in the maximum likelihood (ML) sense for modulation frequency was derived by Lourens [1] for the case when both the carrier and additive noise are a sequence of independent and identically distributed (IID) Gaussian random variates. He showed that when the signal is weak, the ML estimator for the shaft rate and blade rate modulation frequencies reduces to a Fourier analysis of the magnitude-squared measured data sequence. However, this ML estimator does not take into account the colored characteristics of both the carrier and background noise. Another problem is that the acoustic channel, e.g. multipath propagation, can induce a dispersive-like distortion upon the modulation waveform which leads to acoustic frequency-dependent modulator fundamental and harmonic line phase shifts. Consequently, the traditional ML estimator of Lourens can perform poorly because of modulator phase misalignment across acoustic frequency.

Motivated by these problems, we propose a new procedure for estimating modulation frequencies based on subband processing that is akin to multitaper spectral estimation. This procedure is shown to be an approximate realization of the ML estimator when the carrier and background noise are colored. Furthermore, this new approach allows the development of parametric and blind methods for adaptively compensating for channel effects. In this paper, we review the traditional ML modulation estimator and then describe our new procedure and compensation for channel effects. This is followed by a demonstration on actual merchant ship radiated noise.

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Minimum Hellinger Distance Classification of Passive Underwater Acoustic Signals

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Passive source classification in the underwater environment is a challenging problem in part because propagation through the space- and time-varying medium introduces variability and uncertainty in the signal. Acoustic propagation codes can predict received fields accurately but they are sensitive to input environmental parameters which cannot be known exactly. This uncertainty in environmental knowledge used in signal predictions results in imperfect statistical class models. Classifiers that rely on simulations of the environment must therefore be robust to imperfect environmental models. Maximum likelihood methods provide ideal performance when the class models are correct but their performance quickly deteriorates when class models are imperfect. Minimum distance methods generally can offer robustness to mismatches at the expense of performance, with that tradeoff governed by the distance metric used. Hellinger distance, when used as a distance metric, offers robustness to outliers while retaining the performance of a maximum likelihood method, properties that make it well-suited for classification of passive underwater acoustic signals. In the present work the robustness of the Minimum Hellinger Distance Classifier (MHDC) is quantified and its performance is compared to a Log-Likelihood Ratio Classifier(LLRC) with three different data sets: synthetic Gaussian data, synthetic acoustic data from propagation simulations and real acoustic data. In both cases of acoustic data, class models are derived from acoustic propagation simulations. In each case ROC curves show that the MHDC exhibits performance equivalent or superior to that of the LLRC, responding in a robust manner to imperfect class models.

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Time-Frequency Approach and Approximation to Range-Dependent Pulse Propagation

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In shallow water channels, different frequencies of a sound wave propagate at different velocities (dispersion), and are attenuated at different rates (damping). Because of these frequency-dependent propagation effects, it is natural to consider a phase space view of dispersive propagation, in either time-frequency or position-wavenumber phase space. This approach has been investigated by many researchers and, recently, a simple but accurate phase space approximation for dispersive pulse propagation was developed by Cohen and Loughlin. This approximation has been shown to be more accurate than the stationary phase approximation, and has further lead to new "propagation-invariant" features for automatic classification of the sonar backscatter from objects. The focus to date has been on range-independent propagation, which can be modeled by a parallel plate waveguide with stratified layers, such as the Pekeris waveguide. However, in coastal waters, the ocean often has a sloping bottom, and thus a better model for this range-dependent environment may be obtained by considering propagation in a wedge, which has received considerable attention. In this talk, we will extend the range-independent phase space approximation to the range-dependent case of propagation in a wedge.

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The propagation of noise fields in a dispersive medium: a phase-space approach

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Our aim is to show how a noise field propagates in a medium that has dispersion and frequency-dependent attenuation. There are many mechanism that produce noise in the ocean and while some are stationary many are not. This is particularly the case of noise fields commonly called clutter and reverberation. We address and solve the following problem: Suppose a noise field with known statistics is generated in a region or point in space, then, what are the statistics at a later time and at an arbitrary point in space where the propagation is in a dispersive medium. We emphasize that the method presented deals with noise fields and stochastic processes that are not necessarily stationary either locally or globally. We show that considerable insight may be achieved if the problem is analyzed in phase-space where the phase-space distribution may be time-frequency or position-wavenumber. We obtain an exact differential equation for propagating noise fields and show that it leads to effective approximate methods for practical calculations. The approach is to deal with deterministic wave propagation in phase space where we derive exact expressions for the spreading of a propagating pulse in a dispersive medium. The conditions for contraction of a pulse before it eventually spreads to infinity are derived. One then ensemble averages to do the noise case. We first deal with the case where the environment is deterministic. By a deterministic environment we mean that propagation parameters or medium have no random aspects and that the random aspects come in only in the initial generation of the noise field. A number of examples will be given, and application to a train of pulses relevant to reverberation and clutter is also discussed.

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Sequential and iterative Bayesian methods for acoustic feature estimation

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In this work, we investigate how Bayesian methods can be employed for the extraction of waveguide dispersion characteristics. We show that we can express the multimodal acoustic field through a parametric model, subsequently extracting the model parameters (frequencies and amplitudes) with a Gibbs Sampler. The process estimates iteratively the joint posterior probability distribution for all unknown parameters of interest, drawing samples from derived conditional probability distributions. At the same time, the probability distribution for the number of modes present at a specific time is also estimated. Marginal distributions and moments for all parameters can then be calculated in a straightforward manner. Extending this approach, modal dispersion trajectories can be tracked using the same parametrization and sequential Bayesian methods (particle filters), estimating frequencies and amplitudes by relating measurements at consecutive times. Results consist of posterior probability distributions of modal frequencies and amplitudes, which can be used for geoacoustic inversion including uncertainty analysis. Similar approaches are considered for arrival time tracking from short range receptions, taking advantage of a dynamic evolution of arrivals across spatially separated receiving phones. Results are employed in inversion for geometry parameters.

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Statistics-based classification of passive sonar signals

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A number of at-sea measurements have shown that fluctuations in the amplitude of passive sonar signals are affected by source depth, range and the propagation path from the source to the receiver. However, to date, passive sonar signal statistics have not been incorporated into operational sonar classification algorithms because the field results have not been accompanied by development of a statistics-based classifier architecture whose performance can be assessed using accepted signal processing metrics, e.g. a receiver operating characteristic (ROC) curve. The performance of a statistics-based classifier depends critically upon statistical knowledge of received signal, which in turn depend upon statistical knowledge of relevant environment parameters (sound speed in the water column and sediment, for example), and thus the classifier architecture must make use of the environmental information which in reality is only approximately known and must be described in statistical terms. Given a statistical description of the environment, an ocean acoustic propagation model and Monte Carlo simulation is needed to predict received signal statistics from environmental parameter statistics. We present some results showing how well two acoustic models predict the statistics of received signal frequency and amplitude from environment parameter statistics. Once received signal statistics are available for different classes, a number of classification architectures are possible. We compare Bayesian, maximum likelihood, and probability distribution function divergence methods.

[Work sponsored by ONR Undersea Signal Processing]

Application of the Turbo principle to Underwater Acoustic Communications: Receiver Architectures and Experimental Results

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We investigate a practical receiver implementing iterative equalization and decoding, or so-called turbo-equalization, for underwater acoustic communications (UAC). Over the last decade, tremendous gains, leading to near-capacity achieving performance, have been shown for a variety of communication systems through application of the “turbo principle,” i.e. the exchange extrinsic information between constituent decoders for tasks such as channel decoding, equalization, and multi-input multi-output (MIMO) detection. For the underwater channel, we used space-time bit interleaved-coded modulation, to explore the complexity and performance trade-offs of a variety of turbo-equalization-based receiver architectures. Two configurations for hydrophone arrays were considered; Nx1 single-input multi-output (SIMO) and MxN MIMO single carrier transmissions. In order to maintain low complexity for use in channels with long delay-spreads, we use an adaptive LMS turbo equalizer, which approximates the operation of the linear minimum mean square error turbo equalizer, yet does not require an explicit channel estimate. The equalizer processes both the received signals as well as extrinsic information from the MAP-decoders for the channel code, which each process soft-outputs of the equalizer. In addition, the sparse structure resulting from the underwater channel can be exploited to reduce the complexity of the equalizer and mitigate error propagation. This receiver architecture was used to process experiments from the SPACE 08 experiment. Receiver performance for different modulation orders, channel codes, and hydrophone configurations was examined at a variety of distances, up to a km, from the transmitters. Experimental results show great promise for this approach, as information rates in excess of 20 kbps data rate could readily be achieved without error.

Markov Chain Monte Carlo Detectors for underwater acoustic channels

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In this work, we propose novel statistical detectors/equalizers to combat intersymbol interference for frequency selective channels. This class of soft-in soft-out (SISO) detectors are based on Markov chain Monte Carlo (MCMC) techniques. As opposed to the widely used minimum mean square error (MMSE) based equalizers, the MCMC detector aims to approximate the optimal maximum a posteriori (MAP) detector with a low-complexity algorithm. While the MAP equalizer has a complexity that grows exponentially with the constellation size and the memory of the channel, the complexity of the MCMC equalizers grows linearly. This makes the MCMC detector particularly attractive for underwater acoustic channels with long channel memory. We examine the effectiveness of the MCMC detector processing both simulated and experimental data. For stationary channels with perfect channel state information, we show that the MCMC detector performs closely to the MAP detector under various levels of amplitude distortion. We also demonstrate through data collected during a field experiment that the MCMC detector achieves excellent performance when combined with adaptive least mean square (LMS) channel estimators. This establish MCMC detectors as promising detectors for underwater acoustic communications.

High rate time reversal communications for underwater MIMO channels

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Underwater acoustic communications is critical in a number of scientific missions in the ocean, such as ocean exploration and observation, navigation and telemetry for autonomous underwater vehicles, etc. However, present acoustic communications is insufficient in achieving the needed data rates. The main challenges include severe multi-path spread, time varying properties and limited underwater channel bandwidth. Multiple-input/multiple-output (MIMO) techniques are bandwidth efficient in rich scattering environments and have been considered to drastically improve data rates in the underwater channel. For example, various receivers for MIMO systems were developed based on the multichannel decision feedback equalizer (DFE) structure. In the highly dispersive underwater environment, the channel usually has a length of tens of symbols or more. The recovery of multiple information sequences from excessively long MIMO channels is often implemented at significant complexity if the multichannel DFE structure is used. In this paper, a low complexity receiver has been proposed based on the time reversal principle for the underwater acoustic MIMO channel. The core demodulator uses time reversal combining followed by a single channel DFE to demodulate data symbol sequences transmitted by multiple transducers. In order to combat fast channel fluctuations at high frequency, phase tracking and frequent channel estimation at individual channels are employed. Improvements through the use of interference cancellation techniques and effects of different channel estimators are also discussed. Based on the experiment conducted in a shallow water region west off the Kauai Island, Hawaii in 2005, the proposed receiver can demodulate multiple symbol sequences from up to four transducers at the carrier frequency of 37.5 kHz. For example, simultaneous transmission of four binary phase shift keying (BPSK) symbol sequences at an aggregate rate of 16 kbits/s can be demodulated. Four 4-phase shift keying (QPSK) sequences at an aggregate e rate of 32 kbits/s can be demodulated. These data rates correspond to bandwidth efficiencies of 2.29 bits/s/Hz and 4.57 bits/s/Hz in the dynamic underwater environment, where the source and the receiver were drifting at a 2 km range.

Maximum entropy probability density functions and recursive Bayesian state estimation

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There is often significant uncertainty associated with ocean environmental parameters that are important to acoustic propagation. Several methods have been developed that estimate the uncertainty in transmission loss based on uncertainty in these environmental parameters. Incorporating transmission loss uncertainty into sonar signal processing techniques using Bayesian methods is an area of ongoing research, but result in increased performance and robustness to mismatch.

This research focuses on using estimates of transmission loss uncertainty to improve shallow water passive sonar localization performance and robustness. Specifically, we are considering a problem were a source is broadcasting a tonal signal while moving through the water column. The approach is grid-based recursive Bayesian estimation of a state vector describing target motion and source level, resulting in a joint posterior PDF describing the source state vector.

A critical part of designing any Bayesian technique is defining prior probability density functions (PDFs). In this work the prior PDFs represent the uncertainty in transmission loss values, and are the result of acoustic modeling with environmental uncertainty. It is important to find PDFs that accurately represent the model results and also result in an implementable signal processing structure. For example, the assumptions of Gaussianity, Independence and stationarity greatly simplify the derivation and implementation of Bayesian signal processing techniques. This talk assumes none of those.

Monte Carlo acoustic forward models result in an ensemble of possible transmission loss values. The Maximum entropy method can be used to derive a linearly correlated non-stationary non-Gaussian joint PDF describing the ensemble of possible transmission loss values. The PDFs resulting from the maximum entropy method are members of the exponential family, which result in estimator-correlator recursive structures.

This talk presents novel architecture for recursive Bayesian estimation of source location. It begins with an ensemble of transmission loss values resulting from Monte Carlo simulation, uses the Maximum entropy method to derive an exponential family joint density, and derives the resulting grid based Bayesian localizer. The use of linear state update equations and exponential family priors guarantee that the posterior PDF will be of the exponential family.

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Detection and classification of odontocetes - algorithms and architecture for autonomous, deployable system

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Odontocetes or toothed whales include dolphins, beaked whales, killer whales, and sperm whales. These animals produce a variety of wideband and high source level vocalizations whose detection and tracking often provide the only opportunities to observe these animals, especially when submerged. These vocalizations can be grouped into clicks, whistles, and click trains (burst pulses). We have been working with both single sensors and arrays covering up to 100 kHz of bandwidth, although the clicks often extend beyond this. Typically, we have enough SNR that the arrays are used less for array gain, and more for sorting the waveforms by bearing. Sorting is important, since dolphins are often observed in groups of more than 100 animals. Whistles consist of one to five tones in a harmonic series whose fundamental frequency changes continuously over as much as several seconds. Thus, whistles provide fairly long unambiguous observations of single animals, since overlapping whistles are clearly distinguishable due to their different frequency trajectories. Whistles are typically observed in the 5-50 kHz band. Although whistles span a wide band, the animal typically travels an appreciable distance during their duration, so it is a challenge to realize a pulse compression that yields a multipath structure that can be used for ranging. Clicks, on the other hand, are very short in duration, and it is challenging to distinguish multiple clicks reflected from the animal's own anatomy, from multipath arrivals reflected from the ocean boundaries, or clicks from different animals altogether. This is unfortunate, because their extremely short durations could yield very precise ranging accuracy, including tracking multiple animals in a group, if clicks could be suitably sorted by individual animal. Interestingly, click trains provide the best of both worlds. They provide a fairly long observation of a single animal, because both the amplitude and the inter-click interval changes in a continuous fashion during a click train. At the same time, the individual clicks themselves are very short in duration, leaving plenty of gap time for whistles, other clicks, or multiple click trains to be observed.

We have been developing an autonomous processing architecture for detection and classification of marine mammals, to be deployed on fixed or mobile platforms, using hardware based on next generation mobile phone technology. We will present results of applying our processing to recordings from the Southern California Offshore Range (SCORE) site, focusing on detecting click trains and the classification features culled from these signatures.

Performance Prediction Analysis of Depth Discrimination Algorithm

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In this work we analyze the performance of a recently proposed algorithm for passive discrimination between surface and submerged sources in shallow water [1,2]. This algorithm is based on the relative excitation of high-order vs. low-order modes propagating in the acoustic waveguide. It compares the total incoherent sum of the energy of a data vector projected into two possible modal-based subspaces: one for surface sources, and the other for submerged sources.

A clear statement of this problem is one of binary hypothesis testing [1]. In this work we theoretically predict the detection performance of this algorithm. We interpret the results in terms of the fundamental physics of modal filtering: the sound-speed profile, frequency, and the length and depth of the sensing array. The algorithm's detection performance is compared with the bound given by the likelihood ratio test (LRT). For both the algorithm under consideration and the LRT, the detection statistics can be evaluated as a general quadratic form. The detection and false alarm probabilities for this quadratic form are not described by standard distribution functions; we evaluate them using the statistical method of saddlepoint analysis [3].

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A Globally Convergent CML Algorithm

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Narrowband adaptive array processing algorithms (e.g. MVDR, DMR, MUSIC) are applied in broadband applications by FFTing the element data, adaptively processing bins separately, then combining spatial spectra and/or time series across bins. Covariance estimation requires that the FFT length be several times longer than the acoustic travel time across the array. The number of FFTs required per update (the sample support) is roughly 10xM where M is the number of channels for MVDR or MUSIC, or number of dominant modes for DMR. Large wideband arrays require long update times or, for DMR, a number of dominant modes much smaller than the number of channels. Non-stationary environments require the latter approach, which reduces detection and tracking performance in cluttered environments.

Parametric adaptive processing methods, e.g. the Conditional Maximum Likelihood (CML) algorithm [1] require much less sample support than the above mentioned non-parametric methods, since they fit a single source model across the entire bandwidth. The CML algorithm estimates a fixed number of source directions $\Theta = (\theta_1, ..., \theta_p)^T$ in order to maximize $J(\Theta)$, the sum over frequency f and snapshot number m of the norm squared of the projections of M FFT data vectors onto the source subspace at each frequency:

$$J(\Theta) = \sum_f \sum_m \mathbf{x}^H_{m,f} \mathbf{P}_{\mathbf{V}_{\Theta,f}} \mathbf{x}_{m,f} \tag{1}$$

where $\mathbf{P}_{\mathbf{V}_{\Theta,f}}$ is the matrix which projects onto the column span of $\mathbf{V}_{\Theta,f}$, the steering matrix for signal directions Θ at frequency f. Local minima of this function are found using Newton's method, with a single gradient and Hessian computed over all frequency at each update. Because a single parametric model is used across all FFT bins, the required sample support is reduced by approximately the number of FFT bins used. For some torpedo defense sensors this has resulted in two orders of magnitude reduction in required sample support.

Despite its significant advantages, use of the Newton-CML algorithm has been limited due to a lack of methods to ensure global convergence. In this paper we will present a new CML method that uses knowledge of the geometric structure of the array manifold [2] to find the globally optimal CML source locations. Examples of algorithm performance and processing speed on large torpedo defense arrays will be presented.

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Application of Multitaper Spectral Estimation Methods For Adaptive Beamforming

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The covariance matrix for an array associated with a stationary space-time process is completely determined by the individual element locations, the directional response and noise of those elements, and the spatial spectrum of the process. For an ideally calibrated array, when the process is stationary in both time and space, a structure is imparted to the covariance matrix and there is a well-defined Fourier transform relationship between the elements of the covariance matrix and the spatial spectrum. An estimate of the covariance matrix can be determined from an estimate of the spatial spectrum. When positional or array manifold response errors exist the spatially stationary assumption is no longer valid and additional processing is required [1]. When the magnitude of these errors is not severe the situation can be considered partially structured.

This paper investigates application of Thompson's [2] method of multi-taper spectral estimation (MTSE) for both structured and partially structured problems. MTSE works well for spectra containing both a smooth, continuous portion (the background noise) and discrete line components (incident plane waves) and is designed to work with as little as a single snapshot. In the partially structured case, multiple snapshots are required. This paper considers how to best utilize multiple snapshots for the detection of discrete components, a process termed harmonic analysis by Thomson, as well as how to perform adaptive weighting of the spectral components with multiple snapshots. We then compare the performance of adaptive beamformers derived from such estimates of the covariance matrix with those based on the more traditional sample covariance matrix.

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Spatial Compressive Sensing for Passive Sonar Arrays

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This work utilizes recent advances in the compressive sensing (CS) theory for a problem of acoustic field directionality (AFD) estimation, using short, dynamic and not-necessarily uniform passive sonar array. When the number of measurements is significantly smaller than the number of spatial measurements, the addressed problem is underdetermined. Considering that the AFD is sparse or compressible in some basis (wavenumber-frequency domain), we propose using the CS framework to reconstruct the high-resolution AFD from a small number of spatial measurements treating the array manifold as a deterministic sensing matrix. A spatial CS (SCS) based approach for AFD estimation is proposed. The main idea of the proposed approach is to exploit the array orientation diversity (which is achievable by dynamic arrays) in the CS framework to address some challenging problems in the array signal processing, such as left-right ambiguity and poor performance at the endfire region. The proposed method is conceptually different from any classical and subspace-based methods. It provides high azimuth resolution using a short linear array without restricting requirements on the AFD spatial and temporal stationarity and correlation properties of sources and ambient noise. The performance of the SCS-based method is compared to super-resolution methods and to those that exploiting the array orientation diversity for AFD estimation. This work demonstrates via simulation that the SCS-based method outperforms other tested methods in a single-snapshot scenario with multiple sources.

Exploiting Towed Array Dynamics for Passive Field Directionality Mapping

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Towed-array processing algorithms are typically designed assuming a linear array and often treat array shape changes which occur during tow-ship maneuvers as problematic. It is well-known that applying beamforming weights designed for a straight array during a turn when in reality the towed-array is deformed can result in beam broadening [1]. Some of the resulting losses can be mitigated if the array element positions are known and used appropriately to calculate beamformer weights for a non-straight array. In [2] adaptive beamforming is performed on a dynamic array whose element positions have been estimated using a water-pulley model. Left and right ambiguities are resolved by associating peaks that maintain near constant bearing when power versus bearing is stabilized to true North. However, previous methods for dealing with changing array shape do not explicitly mitigate backlobes due to strong interferers which can mask low SNR targets of interest nor have potential to reduce loss of bearing resolution near end-fire that comes with array orientation diversity. For the case of multiple linear array orientations, 360° field directionality estimates, in which ambiguous backlobes are suppressed and endfire resolution is improved, can be obtained by combining multiple beamforming outputs from different orientations of a linear array [3, 4].

In this paper, we extend the idea of combining multiple static array orientations for the purpose of exploiting towed-array shape changes during maneuvers. The proposed recursive field-directionality mapping (RFDM) method updates field directionality maps over time using a non-negative least squares (NNLS) algorithm. A water-pulley model is assumed for modeling array dynamics during ship maneuvers and element positions are estimated from sensor heading outputs along the array. From this, a set of linear equations relating field directionality to beamformer outputs (taken while maneuvering) is solved recursively using a NNLS algorithm in order to form unambiguous 360° bearing time record (BTR). Simulation results demonstrate improved BTR performance compared to conventional BTRs as well as the ability to track low SNR sources in the presence of strong interference backlobes.

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Passive Ranging Capability of Multi-Module Arrays in Underwater Acoustic Environments

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This work presents our investigation results on utilizing an array of volumetric arrays for passive source localization in underwater acoustic environments. The array consists of multiple modules of widely separated volumetric arrays in an attempt to provide flexible capability in estimating source location information under various acoustic conditions. We study the effect of signal wavefront coherence on the performance of source localization. We also propose and develop a data-driven two-stage processing approach, a far-field distributed direction of arrival (DOA) estimation followed by a centralized near-field wavefront curvature exploration, to effectively estimate the source location. Computational as well as data collection experiments are carried out to test for the effects of different levels of signal coherence on the whole arrays capability for effective source localization. Finally, we present our comparative study results on source localization performance delivered by various solutions, including non-coherent DOA based triangulation approach, partially coherent two-stage approach, and fully-coherent centralized approach, under different levels of wavefront coherence.

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Wednesday **October 14, 2009**		**Thursday** **October 15, 2009**		**Friday** **October 16, 2009**	
		8:15–9:55	Session B UW ACOMMS I Laurel	8:15–9:30	Session G UW ACOMMS III Laurel
		9:55–10:20	Break Laurel	9:30–9:55	Break Laurel
		10:20–12:00	Session C UW ACOMMS II Laurel	9:55–10:45	Session H Passive Laurel
		12:00–1:00	Lunch Whisp. Pines	10:45–12:00	Session I Array I Laurel
		1:00–2:15	Session D Active Laurel	12:00–1:00	Lunch Whisp. Pines
		2:15–3:05	Session E Passive I Laurel	1:00–2:15	Session J Array II Laurel
		3:05–3:30	Break Laurel		
		3:30–5:10	Session F Prop.-Based Laurel		
5:00–6:00	Welcome Reception Whisp. Pines				
6:00–8:00	Raytheon Dinner Whisp. Pines	6:00–8:00	OES Dinner Whisp. Pines		
8:00–9:30	Session A Plenary Laurel	8:00–?	SOB Session Laurel		