A Book of Abstracts for the

2013 Underwater Acoustic Signal Processing Workshop

October 16-18, 2013

Alton Jones Campus

University of Rhode Island

West Greenwich, RI, USA

Sponsored by the IEEE Providence Section
with promotional partners the Office of Naval Research
the IEEE Oceanic Engineering Society, the Acoustical Society of America, and Raytheon

UASP 2013

Welcome to the 2013 IEEE workshop on Underwater Acoustic Signal Processing. This year we have two special sessions. The first is a memorial session honoring Prof. Donald W. Tufts and organized by Ashwin Sarma. The second is a special session on continuous active sonar, organized by Douglas Abraham and Keith Davidson.

The organizing committee thanks and acknowledges the continued support of our promotional partners, the Office of Naval Research, the IEEE Oceanic Engineering Society, Raytheon Integrated Defense Systems, and the Acoustical Society of America. We also thank Michael Janik for his efforts in arranging for Raytheon Integrated Defense Systems to sponsor our Wednesday evening dinner. The IEEE Oceanic Engineering Society is sponsoring the Thursday evening dinner, and the Acoustical Society of America is sponsoring the Thursday lunch.

Finally, the organizing committee is proud to announce that this year's recipients of the Donald W. Tufts UASP Award are Prof. Louis L. Scharf of Colorado State University and Prof. Richard J. Vaccaro of the University of Rhode Island. This year we renamed the award in memory of Prof. Tufts, and it is fitting that the first two recipients are his long-time colleagues.

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The 2013 Donald W. Tufts UASP Award is Presented to Prof. Louis L. Scharf

In recognition for outstanding contributions to adaptive beamforming and statistical signal processing

Prof. Louis L. Scharf received the B.S., M.S. and Ph.D. degrees from the University of Washington in 1964, 1966 and 1969 respectively. After two years at Honeywell's Marine Systems Center, he joined the faculty of Colorado State University in the Departments of Electrical Engineering and Statistics in 1971. In 1982, he moved to the University of Rhode Island to serve as Chair and Professor of the Electrical Engineering department. He joined the University of Colorado Boulder in 1985 as a Professor of Electrical Engineering, then returned to Colorado State in 2001 where he holds an appointment as Professor of Electrical and Computer Engineering and Statistics. Louis has been a Fellow of the IEEE since 1986, and has received several honors from the IEEE including the IEEE Third Millenium Medal (2000), the IEEE Signal Processing Society Pioneer Award (1996), an IEEE Distinguished Lectureship (1993, 1994) and the IEEE Signal Processing Society Technical Achievement Award (1994). Louis has generously given his time to several IEEE Conferences and Committees, including chairing the Asilomar Conference (2002) and the ICASSP Technical Program (1980). He is frequently in demand as a Plenary Lecturer.

Louis' research has had a sustained and pervasive impact on the field of underwater acoustic signal processing. He has made fundamental contributions in the areas of matched and adaptive subspace detectors, invariance theories, and canonical representations. His 1991 textbook on *Statistical Signal Processing* is widely revered as a classic in the field. His more recent collaboration with Peter Schreier *Statistical Signal Processing of Complex-Valued Data: The Theory of Improper and Noncircular Signals* has provided new insights into the field of improper random variables. Louis' talks and seminars are admired for their lucid presentation accompanied by insightful geometric figures providing straightforward interpretations of complicated algorithms. Louis is an excellent mentor to younger researchers, and many of his PhD students have gone on to successful careers in both academia and industry.

This year we renamed our award in memory of the UASP workshop founder, the late Professor Donald W. Tufts. Given his long history of collaboration with Professor Tufts and association with this workshop, it is especially fitting that Professor Louis L. Scharf is one of the first recipients of the newly renamed award.

The 2013 Donald W. Tufts UASP Award is Presented to Prof. Richard J. Vaccaro

For outstanding contributions to subspace signal processing and dedicated service to the UASP Workshop

Richard J. Vaccaro received both the B.S. and M.S. degrees from Drexel University in 1979 and the Ph.D. degree from Princeton University in 1983. He immediately joined the Electrical Engineering faculty of the University of Rhode Island (URI) where he has spent his entire professional career and where he served as Department Chair from 1999 to 2004. Rick served the IEEE Signal Processing Society as Chair of the Underwater Acoustic Signal Processing Technical Committee from 1995 to 1998. In this capacity, he provided leadership in the subsequent formation of the Sensor Array and Multichannel Signal Processing Technical Committee. In 2009, Prof. Vaccaro received the IEEE Region 1 Award for Outstanding Support for the IEEE Mission. He has also served as Faculty Advisor to the URI IEEE Student Branch since 1999.

This award recognizes Rick's contributions to underwater acoustic signal processing through his research papers and books. Rick published the textbook *Digital Control: A State-Space Approach* and edited the volume *SVD and Signal Processing II*. His journal papers feature insightful applications of subspace signal processing, perturbation expansions, and matrix filters to important detection and estimation problems. These insights have lead to a deeper understanding of existing algorithms and the development of important new algorithms. Rick's research has made significant contributions to the estimation of direction of arrival, location and parameters of sources, as well as novel approaches to interference suppression and signal subspace dimension estimation. Rick also leveraged these techniques to estimate perturbations of array elements to improve the robustness of array processing algorithms. Much of this work was accomplished in close partnership with the graduate students that he has successfully mentored over the past 30 years.

The UASP Workshop Organizing Committee is exceedingly grateful to Rick for his strong and consistent support of the UASP Workshop. Rick has been a leading presence on the UASP Workshop Organizing Committee further back than any other present member of the Workshop Committee can remember. Most recently, Rick has served as the Local Arrangements Chair since at least 2001. He is the steady hand that has made the biennial returns to the Alton Jones Campus so important and enjoyable for so many of us. In recognition of this commitment to the UASP Workshop and for his contributions to subspace-based signal processing methods, we are honored to present the UASP Award to Professor Richard J. Vaccaro in the first year that it has been renamed in memory of his long-time colleague and friend, Professor Donald W. Tufts.

Schedule at a glance

Wednesday **October 16, 2013**		**Thursday** **October 17, 2013**		**Friday** **October 18, 2013**	
		8:30–9:45	Session B CAS I Laurel	8:15–9:30	Session F CAS III Laurel
				9:30–10:20	Session G AUV Signal Proc. Laurel
		9:45–10:15	Break Laurel	10:20–10:45	Break Laurel
		10:15–11:55	Session C Sensing & Noise Laurel	10:45–12:00	Session H UW Acoustic Comm. Laurel
		12:00–1:00	ASA Lunch Whisp. Pines	12:00–1:00	Lunch Whisp. Pines
		1:00–3:05	Session D Array Proc. Laurel	1:00–3:30	Session I Detection, Classification, and Tracking Laurel
		3:05–3:30	Break Laurel		
		3:30–5:10	Session E CAS II 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		
8:00–9:45	Session A Tufts Memorial Laurel	8:00–?	SOB Session Laurel		

Sessions: Titles and presenters

Session A: Wednesday Evening, 8:00pm–9:45pm

Donald W. Tufts Memorial Session

A–1 *On the (Somewhat) Obscure Origins of the UASP Workshop*
Richard J. Vaccaro and John P. Ianniello

A–2 *Compressive Sensing and Threshold Effects in Signal Processing*
Louis Scharf, Colorado State University

A–3 *Rank Reduction: A Journey from Principal Components to Conjugate Directions for Sparsity Exploitation*
Ivars Kirsteins, Naval Undersea Warfare Center

A–4 *The Reasonable Effectiveness of Nonparametric Methods in Signal Processing*
Ashwin Sarma, Dept. ECBE URI and Naval Undersea Warfare Center

Session B: Thursday Morning, 8:30am–9:45am

Special Session I: Continuous Active Sonar

B–1 *Continuous Active and Pulsed Active Sonar Measurements in Littoral Waters*
Paul Hines, Defence R&D Canada

B–2 *Operational Deployable Continuous Active Sonar Processing Chain*
Roger Benton, Thales Underwater Systems Ltd

B–3 *On the Environmental Impact of Continuous Active Sonar*
Robbert van Vossen, TNO

Session C: Thursday Morning, 10:15am-11:55am

Environmental Sensing and Noise

C–1 *Sound Speed Estimation and Source Localization in the Ocean via Linearization and Particle Filtering*
Zoi-Heleni Michalopoulou, Dept. of Mathematical Sciences, New Jersey Institute of Technology

C–2 *Application of Time Frequency Analysis Techniques for Long Range Sediment Tomography*
Gopu Potty, University of Rhode Island

C–3 *The Evolution of Nonstationary Noise and an Application to Reverberation*
Leon Cohen, City University of NY

C–4 *Effects of Environmental Variation and Uncertainty on a Physics-Based Depth Discriminator*
Brett Bissinger, Applied Research Laboratory, The Pennsylvania State University

Session D: Thursday Afternoon, 1:00pm–3:05pm

Array Processing

D–1 *Fractionally Spaced Passive Synthetic Aperture Sonar for Operation Above the Design Frequency*
Kristine Bell, Metron, Inc.

D–2 *Rank-Adaptive Dominant Mode Rejection Beamforming*
John Buck, University of Massachusetts Dartmouth

D–3 *Convergence Rate of the Dominant Mode Rejection Beamformer*
Kathleen Wage, George Mason University

D–4 *A Random Matrix Theory Model for the Mean Notch Depth of a Diagonally Loaded MVDR Beamformer*
Saurav Tuladhar, University of Massachusetts Dartmouth

D–5 *Implementing a Minimum Variance Distortionless Response Beamformer - DSP Chips versus Graphical Processing Units (GPUs)*
Paul Hursky, HLS Research Inc

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

Special Session II: Continuous Active Sonar

E–1 *On Designing the Transmission and Reception of Multistatic Continuous Active Sonar Systems*
Jian Li, University of Florida

E–2 *Composite Continuous Active Sonar*
John Murray, Signal Systems Corporation

E–3 *The Generalized SFM Waveform for Continuous Active Sonar*
David Hague, University of Massachusetts Dartmouth

E–4 *Wideband Slow-time Costas Continuous Active Sonar in a Shallow-Water Channel*
Jeffrey Krolik, Duke University

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

Special Session III: Continuous Active Sonar

F–1 *Multistatic Tracking for Continuous Active Sonar Using Doppler-Bearing Measurements*
Doug Grimmett, SPAWAR Systems Center Pacific

F–2 *Range and Velocity Estimation Methods for Continuous Active Sonar with a Fast Moving Target*
Chunsheng Liu, Defence R&D Canada Atlantic

F–3 *A Multiuser Framework for Sonar Watermarking*
Bijan Mobasseri, Villanova University

Session G: Friday Morning, 9:30am-10:20am

AUV-Based Signal Processing

G–1 *Environmentally Sensitive AUV Behaviors for Collaborative Multistatic Surveillance Networks*
Ryan Goldhahn, NATO Centre for Maritime Research and Experimentation (CMRE)

G–2 *Measuring the Directionality of the Ambient Noise Field from an Autonomous Underwater Vehicle*
Stephanie Fried, Massachusetts Inst. of Technology

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

Underwater Acoustic Communications

H–1 *Optimal Joint Channel Estimation and Data Decoding, and Problems with Sequential Analysis in Underwater Communications*
Andrey Morozov, Teledyne TWR

H–2 *Observations of Broadband High Frequency Coherence Degradation in Shallow Water Environments*
Paul Gendron, University of Massachusetts Dartmouth

H–3 *Field Tests of Adaptive Modulation and Coding for Underwater Acoustic OFDM*
Shengli Zhou, University of Connecticut

Session I: Friday Afternoon, 1:00pm–3:30pm

Detection, Classification, and Tracking

I–1 *Target versus Clutter Discrimination in Terms of the Wigner Spectrum and the (Symmetric) Ambiguity Function*
Patrick Loughlin, University of Pittsburgh

I–2 *Target Detection and Classification Against Non-stationary Interference Using Dynamic Feature Clustering*
Ivars Kirsteins, Naval Undersea Warfare Center

I–3 *Applying a Featureless Classifier in Non-Stationary Generalized Pareto Distributed Clutter*
Bruce Newhall, The Johns Hopkins U. Appl. Physics Lab.

I–4 *Feature-Aided Tree-Search Tracking on the CLUTTER09 Dataset*
Jill K. Nelson, George Mason University

I–5 *Detection Performance of Coprime Sensor Arrays*
Kaushallya Adhikari, Electrical and Computer Engineering Department

I–6 *A Track Before Detect Method with Motion Model*
Daniel Park, Applied Research Laboratory Penn State

Abstract Listings

On the (Somewhat) Obscure Origins of the UASP Workshop

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This talk will review some of the history of the Underwater Acoustic Signal Processing Workshop and provide an introduction to the Tufts Memorial Session.

Compressive Sensing and Threshold Effects in Signal Processing

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The source of performance breakdown in parameter estimation may be attributed to a subspace swap, wherein one or more modes of a noise subspace better approximate a measurement than one or more modes of a signal subspace. This was the insight of the late Prof. Donald W. Tufts. In this talk we address the effect of compressive sensing on the probability of a subspace swap. We derive an analytical bound on this probability when the parameters to be estimated modulate a rank-deficient signal covariance matrix. This bound predicts quite well the threshold SNR at which performance breaks down when estimating DOAs for closely-space planewaves, using compressed or uncompressed measurements in a linear array.

Acknowledgment: This talk is based on collaborations with Pooria Pakrooh, Ali Pezeshki, and Yuejie Chi.

[This work is supported by the NSF under grant CCF-1018472.]

Rank Reduction: A Journey from Principal Components to Conjugate Directions for Sparsity Exploitation

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In this presentation, we discuss ideas and related solutions for rank reduction developed over the years under the inspirational influence of Dr. Donald Tufts. The impetus is that underwater acoustic signals, e.g., radiated noise from vessel machinery, propeller cavitation noise, active sonar echoes and clutter etc. are typically non-stationary both temporally and spatially. In array processing for example, the intrinsic non-stationarity of the interference and signal often results in sample deficiency (relative to data dimension). Consequently the estimated cross-spectral density matrices are frequently ill-conditioned, making it difficult to apply standard adaptive beamforming methods based on matrix inverse methods.

Motivated by these challenges, we review and trace the history of rank reduction methods in underwater acoustic signal processing starting with the principal component inverse (PCI) method to suppress interference and enhance signals in sample deficient situations. We also discuss its connections to sparse polynomial linear prediction. Principal component methods perform rank reduction on the basis of energy in the particular eigen-component. The need for knowledge-driven, e.g., signal-dependent, rank reduction solutions lead to the development of conjugate gradient and Krylov space-based methods, which potentially provided greater dimensionality reduction and a faster rate of adaption than principal components under certain conditions. Finally we conclude with a brief discussion of how these rank reduction methods are philosophically related to the methodologies used by the compressive sensing and sparse reconstruction techniques in vogue today.

*New Jersey Institute of Technology, Electrical and Computer Engineering Department

The Reasonable Effectiveness of Nonparametric Methods in Signal Processing

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This talk summarizes an area of interest of Don Tufts and his collaborators - the application of nonparametric philosophies to critical signal processing tasks that remain unresolved through historical approaches.

Such critical signal processing tasks include: 1) maximizing detection performance under nominal null and alternate hypotheses while simultaneously controlling the false alarm rate in arbitrary non-Gaussian noise, 2) in-situ adaptive estimation of stationarity lengths to increase performance of subsequently applied adaptive detectors and estimators, 3) the reduction of a large set of classification features to a smaller, equally informative set and 4) methods to eliminate overtraining artifacts from a fixed classification decision rule and accurately bound its performance.

We hope to show that approaching such problems from a nonparametric viewpoint is certainly worthwhile, perhaps even unreasonably effective à la Wigner.

Continuous Active and Pulsed Active Sonar Measurements in Littoral Waters

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Military sonars must detect, localize, classify, and track submarine threats from distances safely outside their circle of attack. However, conventional pulsed active sonars (PAS) have duty cycles on the order of one percent which means that 99% of the time, the track is out of date. In contrast, continuous active sonars (CAS) have a 100% duty cycle which enables continuous updates to the track. If one can overcome technical challenges such as the high dynamic range required by the receiver, then CAS should significantly improve tracking performance in the free-field environment which one encounters (approximately) in the deep ocean; however, improvements in tracking performance in shallow water are not assured since both targets and clutter will be tracked continuously and CAS may increase false tracks to an unacceptably high level essentially continuously tracking the clutter. Theoretical predictions of performance are challenging since the reverberation background for shallow water CAS has not been accurately modeled. To compare performance of CAS with conventional PAS in the littorals, a set of experiments were conducted as part of the Target and Reverberation Experiment (TREX) in May 2013. In this paper preliminary results from the field trial will be presented.

[Partial funding from the U.S. Office of Naval Research Grant N62909-12-1-7130 is gratefully acknowledged.]

Operational Deployable Continuous Active Sonar Processing Chain

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Existing in service underwater Sonar equipment has exploited pulsed sonars, generally frequency modulated with increasing bandwidths to provide improved detection, clutter reduction and classification in both deep water and littoral water operations. The same equipment is adaptable to transmitting continuous swept waveforms however the inboard processing for a continuous active sonar (CAS) system is very different. This paper provides an overview of the CAS processing chain used by Thales in the UK to investigate the processing challenges and identify solutions for a deployable processing chain. Specifically the paper will discuss options for integration within the CAS processing chain, normalisation for robustness against narrowband interferences and data processing issues for auto detection and tracking on CAS data using track before detect concepts and presenting results from recorded data.

On the Environmental Impact of Continuous Active Sonar

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Continuous Active Sonar (CAS) is an emerging technology in anti-submarine warfare operations. The feasibility of the technology has been demonstrated and it has been shown that CAS has a potential to reduce false alarm rates in reverberation-limited conditions. Now that the feasibility has been shown and further use of the technology is considered, it becomes important to consider the environmental impact. There is a special concern for marine mammals, since typical anti-submarine warfare (ASW) sonar has the potential for introducing physical effects and behavioral responses in particular. Compared to conventional pulsed sonar (CPS), there are two main differences: (i) CAS can operate at lower source levels compared to CPS, (ii) CPS operates at low duty cycles, whereas CAS uses a full duty cycle. For the assessment of the influence on marine mammals, the received sound exposure level (a measure of received sound energy level) is usually considered in addition to the peak pressure level. The sound exposure and peak pressure levels are thought to determine the potential of the sonar to induce hearing threshold shifts, where received sound pressure levels, that may be lower for CAS compared to CPS, are thought to be relevant for behavioral responses. However, masking of marine mammal hearing by CAS might be a new issue. Our conclusion therefore is that a detailed assessment of the environmental impact of CAS, and especially the influence on marine mammals, is required.

*The Netherlands Ministry of Defence

Sound Speed Estimation and Source Localization in the Ocean via Linearization and Particle Filtering

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A new approach for source localization in the ocean and estimation of the sound speed profile in the water column is presented. The method is based on linearization of the relationship that links multipath arrivals to geometry and environmental parameters. Arrival times of multiple paths within time-series recorded at an array of phones are here connected to the unknown parameters via a quasi-linear system. The Jacobian matrix required for the solution of the inverse problem consists of partial derivatives of time with respect to the unknown quantities. The Jacobian terms include derivatives with respect to Empirical Orthogonal Function (EOF) coefficients instead of sound speed directly. These coefficients are obtained from a number of in situ measurements. Working with EOFs facilitates the solution of the problem, because modeling sound speed in such a manner simplifies calculations and provides flexibility, as will be demonstrated. The method is initially tested on synthetic data generated following the assumption of Gaussian behavior of multipath arrival times and is shown to provide accurate results. Moreover, the approach is shown to require a small number of iterations to converge. Subsequently, the technique is applied to Shallow Water 06 data, where arrival times and their probability densities at a vertical line array are extracted using particle filtering. The probability densities consist of particles representing arrivals along with their associated weights/probabilities and quantify the uncertainty in arrival time estimation from the data, obviating the need for the assumption of normality. In addition to regular filtering, a smoother is employed for tighter arrival time extraction. The inversion estimates and probability density functions of the unknown parameters using the resulting particles as input to the inverse problem agree closely with ground truth information and estimates obtained via computationally intensive global optimization techniques.

[This work was funded by the Office of Naval Research.]

Application of Time Frequency Analysis Techniques for Long Range Sediment Tomography

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Long range sediment tomography inversion technique requires accurate estimation of the arrival times of acoustic normal modes. The inversion technique is based on minimizing the difference between the forward model predictions and the data using a global optimization scheme. Predictions are computed using a trial parameter set which is iteratively modified until the algorithm converges. The modal arrival times are calculated from the time-frequency analysis of broadband acoustic data collected on a single hydrophone. During the initial stages of the development of the inversion scheme, Fourier-based spectrograms and wavelet-based scalograms were used. Different acoustic normal modes correspond to different frequency bands and they sample the sediment at different depths. Hence achieving good time and frequency resolution over a large frequency band is important to extract maximum number of acoustic normal modes. The sediment tomography technique, as described above, was developed in the late 1990s and is getting more attention currently due to its robustness. In recent years we also have been focusing on extracting shear information from the modal dispersion and this requires a better identification of the arrivals near the late arrival corresponding to the group speed minimum (Airy phase region). Taking advantage of some of the recent developments in time-frequency analysis, dispersion based short time Fourier transform (DSTFT) and warping transform techniques were used for the identification of late arrivals near the Airy phase. This paper will summarize and compare the performance of these time frequency techniques in the context of long range sediment tomography. Broadband data from some of the recent field tests (Shelfbreak Primer and Shallow Water-06 experiments) will be analyzed for this study. Finally, time-frequency analysis performance of another new techniques, such as Modified S transform, will be examined using the field data.

[Work supported by Office of Naval Research.]

The Evolution of Nonstationary Noise and an Application to Reverberation

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The fundamental ideas of noise have been developed for stationary noise. However, there exist many situations where the noise is non stationary both in space and time. Indeed the simplest noises, the Wiener, Langevin and Brownian motion processes are nonstationary. We discuss the mathematical characterization of nonstationary noise and we consider how, under some circumstances, nonstationary noise evolves to quasi-stationary noise and eventually to stationary noise. Of particular interest is the evolution of noise in dispersive wave guides where there may also be attenuation. We develop the random pulse train model wherein the pulses have width and are not necessarily uniformly distributed. We explicitly calculate the space and time correlation functions and their evolution in a wave guide. This model is used to describe reverberation and to calculate the probability distribution of intensity of the noise.

[Work supported by ONR.]

Effects of Environmental Variation and Uncertainty on a Physics-Based Depth Discriminator

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We examine the effects of environmental variation and uncertainty on the performance of a physics-based depth discriminator. The discriminator was developed by Vince Premus [1-3]. It uses the principles of modal propagation in a shallow water waveguide to passively label a source as either surface or submerged. The discriminator requires an accurate picture of the local ocean environment to function properly. Operationally, it is important to understand how accurately the environment must be known and modeled for such a system to operate properly. We examine the effect imperfect knowledge of the environment has on the performance of the processor through both simulation and application to data from the REP11 experiment. The REP11 data includes both spatio-temporal environmental measurements and towed acoustic source recordings. Using REP11 combined with simulations of similar environments, we characterize performance in the presence of both simple mismatches and the complex behavior of the real ocean environment to gather insight into the performance of the discriminator in a real-world scenario.

[1] V. Premus, J. Ward, and C. D. Richmond, "Mode Filtering approaches to acoustic source depth discrimination," in Proc. of the Thirty-Eighth Asilomar Conference on Signals, Systems and Computers, Vol. 2, pp. 1415-1420, November 2004.

[2] V.E. Premus and D. Backman, "A matched subspace approach to depth discrimination in a shallow water waveguide," in Proc. of the Forty-First Asilomar Conference on Signals, Systems and Computers, pp. 1272-1276, November 2007.

[3] V. E. Premus and M. N. Helfrick, "Use of mode subspace projections for depth discrimination with a horizontal line array: Theory and experimental results," J. Acoust. Soc. Am., vol. 133, no. 6, pp. 4019-4031, 2013.

[This research was supported by the Applied Research Laboratory at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

Fractionally Spaced Passive Synthetic Aperture Sonar for Operation Above the Design Frequency

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The frequency band in which a uniformly spaced linear array can operate is limited by the temporal and spatial sampling rates. The spatial equivalent of the temporal Nyquist frequency is the spatial design frequency, which is determined by the array element spacing. Processing above the spatial design frequency incurs the penalty of grating lobes in the array beampattern due to spatial aliasing, in which signal energy at one angle appears incorrectly at another angle, possibly obscuring other signals. For a signal arriving at a given angle, the mainlobe of the beampattern occurs at nearly the same angle in all frequency bins, while the grating lobe locations vary with the temporal frequency. Towed arrays provide the opportunity for creating synthetic elements located a fraction of the distance between the actual physical elements, thus sampling the spatial field at a higher rate and potentially allowing for operation above the design frequency without grating lobes, provided there is adequate temporal sampling. However, the ability of a passive synthetic aperture sonar (PSAS) towed array system to achieve the performance of the equivalent fully populated array is limited by the lack of temporal coherence of typical underwater acoustic broadband signals over the time intervals required to obtain synthetic aperture samples.

In this work, we have developed a broadband signal processing algorithm for generating bearing-time record (BTR) displays of signal energy at frequencies both above and below the design frequency using a fractionally spaced synthetic array. We use a synthetic array response model which exploits any degree of temporal coherence present in the signals to reduce grating lobes, combined with a basis pursuit algorithm to iteratively identify sources of consistent signal energy across all frequency bins. Performance is demonstrated on simulated signals and noise exhibiting realistic broadband and narrowband characteristics. The algorithm shows significant performance advantages over traditional broadband processing using the physical array, and is robust to lack of signal coherence as well as uncertainty in the tow velocity.

[This work is supported by NAVSEA under contract N00024-11-C-4186.]

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Rank-Adaptive Dominant Mode Rejection Beamforming

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The Dominant Mode Rejection (DMR) adaptive beamformer (ABF) is a subspace approach to making the Minimum Variance Distortionless Response ABF more robust. The performance of the DMR ABF relies on an accurate estimate of the correct subspace dimension. The proposed Rank-Adaptive DMR (RADMR) ABF computes its array weights as a weighted mixture of the DMR array weights from fixed dimension DMR ABFs. Simulations indicate that the RADMR pays a relatively small penalty in signal to interference and noise ratio relative to the fixed-dimension DMR ABF with prior knowledge of the correct subspace dimension.

[Work supported by ONR Code 321US.]

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Convergence Rate of the Dominant Mode Rejection Beamformer

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In passive sonar, adaptive beamformers (ABFs) facilitate the detection of quiet sources by removing interference and noise. Since the underwater environment is typically non-stationary, it is important to consider the convergence rate of the adaptive weights when selecting an ABF algorithm. A common metric used to quantify the convergence rate is the signal-to-interference-and-noise ratio (SINR) loss. SINR loss is the ratio of the SINR for an ABF designed using estimated statistics to the SINR for a clairvoyant ABF designed using ensemble statistics. In a classic paper, Reed, Mallett, and Brennan show that the SINR loss of the Minimum Variance Distortionless Response (MVDR) beamformer is beta-distributed [IEEE Trans. Aerosp. Electron. Syst., 1974]. Based on this result, MVDR requires twice as many snapshots as sensors for the SINR to be within 3 dB of the SINR achievable with clairvoyant weights. When the environment fluctuates rapidly, it is not possible to obtain the number of snapshots required for convergence, thus alternative approaches are necessary. One standard approach to improving the convergence rate is to diagonally load the sample covariance matrix (SCM) used to design the MVDR weights. Interestingly, Cheremisin has shown that the SINR loss for a diagonally-loaded MVDR ABF is also characterized by a beta distribution [Radio Eng. Electron. Phys., 1982]. Cheremisin's results indicate that the number of snapshots required is proportional to the number of interferers, rather than the number of sensors. Kirsteins and Tufts [IEEE Int'l. Conf. Syst. Eng., 1989] and Geirull [IEE Proc.-Radar, Sonar Navig., 1997] obtain similar results for a class of reduced-rank beamformers based on the eigendecomposition of the SCM. Specifically, these authors show that SINR loss is beta-distributed for the reduced-rank eigenvector beamformers. This talk investigates the SINR loss of another class of eigenvector ABFs, known as Dominant Mode Rejection (DMR) beamformers [Abraham/Owsley, IEEE Oceans, 1990]. Instead of projecting into a reduced-rank subspace, the DMR ABF uses a structured estimate of the covariance matrix, derived from the eigendecomposition. Similar to previous work, our analysis shows that the SINR loss for the DMR ABF is beta-distributed when the rank of the interference subspace is known. The talk will relate these results to our previous work analyzing the DMR ABF using random matrix theory [Buck/Wage, IEEE SSP Workshop, 2012].

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A Random Matrix Theory Model for the Mean Notch Depth of a Diagonally Loaded MVDR Beamformer

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Adaptive beamformers (ABF) improve the SINR performance over the conventional beamformer in the presence of interferers. ABFs place a notch in the interferer direction which suppresses the interferer power at the output. Therefore, the notch depth (ND) in the interferer direction plays a crucial role in determining the gain in SINR. Prior work derived an analytical expression for the mean beampattern of MVDR beamformer, [Richmond, IEEE SAM 2000]. However the derivation does not consider the case of diagonal loading (DL) applied to the sample covariance matrix. This research derives a model for the mean ND of a DL-MVDR assuming a single discrete interferer. The model describes the mean ND as a function of number of snapshots, number of sensors in the array, interferer to noise ratio (INR) level, interferer location relative to the look direction and the diagonal loading level. The model derived is similar in spirit to the ND model for the DMR beamformer [Buck and Wage, IEEE SSP 2012]. The derivation uses the random matrix theory results on the behavior of sample eigenvectors of a 'spiked covariance matrix,' i.e., a low rank perturbed Wishart matrix. The ND predicted by the model is shown to be in close agreement with simulation results over a range of INRs and snapshots.

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Implementing a Minimum Variance Distortionless Response Beamformer - DSP Chips versus Graphical Processing Units (GPUs)

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This paper will compare two implementations of an adaptive beamformer (ABF), one on DSP chips and one on graphical processing units or GPUs.

We will first present the linear algebraic formulation that is the basis of our implementation, in which we project the steering vectors onto the eigenvector basis of the estimated cross spectral density matrix, so that we can perform all subsequent operations using vector-vector operations, avoiding more costly matrix-vector operations.

An important aspect of the implementation is finding the diagonal loading factor that realizes a user-specified white noise gain constraint at all beams and all frequencies. We have compared several root finders for this non-linear inversion problem, including Brent's and Ridder's methods. These root finders require a bracketing interval before they can start narrowing down their search, which we find starting with a nominal initial value. Obviously, if we start with a value close to the root we are seeking, we do less processing to find a solution. Indeed, if the starting value satisfies the desired white noise gain constraint, then we do not have to do ANY processing. This suggests that we should try to exploit any persistence or continuity in the diagonal loading factors across beams and frequencies. Indeed, we will discuss several strategies for doing so, and show the results of applying such strategies to several datasets and in aid of two different processing configurations: 1) keeping all beams fixed and updating with a 50% overlap, and 2) varying the beams to follow a bearing track and updating the ABF output at every snapshot.

After reviewing our basic formulation of an MVDR ABF, we will discuss how we have adapted our implementation to two different processing architectures, one a Texas Instruments floating point DSP chip, and the other a graphical processing unit or GPU.

On Designing the Transmission and Reception of Multistatic Continuous Active Sonar Systems

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We focus on the transmission and reception of multiple continuous probing sequences in multistatic continuous active sonar (MCAS) systems. The main contributions of this paper are: i) spectrally-contained continuous sequence sets with low correlation sidelobe levels are designed for the MCAS transmission so that the so-generated sequences meet the spectral containment restrictions and the weak correlations among the received echoes can be exploited to improve the target detection performance; and ii) data-adaptive receiver filters are used to form accurate range-Doppler images, which facilitate the determination of target range and Doppler measurements as well as the estimation of the target parameters (position and velocity). Comparing with the pulsed counterpart, the transmission of continuous sequences effectively alleviate the negative impact of the man-made noise on marine life. More importantly, the new continuous scheme has a wider passband as well as both higher range and Doppler resolutions, and thus can achieve significantly enhanced performance of target detection and parameter estimation through exploiting continuous illumination and spatial diversity.

Composite Continuous Active Sonar

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This presentation will discuss the use of composite signals in continuous active sonar (CAS) systems. Conventional CAS transmissions include continuous wave (CW) and frequency modulation (FM) sweeps. CW echoes provide direct information about the target speed, while FM echoes provide direct information regarding the target position. Using composite waveforms that are a coherent combination of CW and FM waveforms can provide both types of target information simultaneously, and lead to better sonar system performance. A relatively simple model of the signal excess reveals the complementary nature of the CW and FM waveforms and enables direct performance comparisons between conventional CAS systems and those using composite transmissions. Multistatic fields of source and receive array sonobuoys are the focus of this effort. Consequently, the acoustic energy consumed by active transmissions plays an important role in performance comparisons. Data from real-world acoustic experiments shows the potential benefits of composite CAS technology.

The Generalized SFM Waveform for Continuous Active Sonar

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The Generalized Sinusoidal FM (GSFM) waveform [Hague, Buck, Asilomar (2012)] modifies the Sinusoidal FM (SFM) waveform to use an Instantaneous Frequency (IF) function that resembles a FM chirp waveform. As a result of this the GSFM possesses a thumbtack Auto-Ambiguity Function (AAF). This is a drastic improvement over the SFM which suffers from poor range resolution as the Auto-Correlation Function (ACF) contains many ambiguous peaks generated by the periodicity of the SFMs IF. The GSFM waveform achieves minimal range-Doppler coupling for single target measurements which in turn minimizes the error in jointly estimating target range and velocity. The GSFM's AF Peak Sidelobe Level (PSL) and Integrated Sidelobe Ratio (ISLR) are comparable to or better than that of other waveforms with a thumbtack AAF. Additionally, the GSFM is a promising candidate waveform for Continuous Active Sonar (CAS) applications. Modifying the GSFM's parameters produces a family of nearly orthogonal waveforms which can be combined as a collection of an even integer N of sub-pulses resulting in a long duration pulse train waveform that possesses a thumbtack AAF. The pulse train can be decomposed into an even integer D of contiguous pulse train waveforms containing N/D sub-pulses which also possess a thumbtack AAF and are nearly orthogonal to each other. These variable length GSFM pulse train waveform families facilitate adaptive Coherent Processing Intervals (CPI) and Target Revisit Rates (TRR) in exchange for Doppler resolution.

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Wideband Slow-time Costas Continuous Active Sonar in a Shallow-Water Channel

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This paper addresses the performance of wideband slow-time Costas (SLO-CO) continuous active sonar (CAS) in a shallow-water environment. Conventional continuous active sonar enables high revisit rates at the expense of reduce per-pulse bandwidth which could result in lower signal-to- clutter-plus-noise gain, particularly highly reverberant shallow-water channels. Wideband SLO-CO CAS, which uses non-recurrent wideband linear FM signals with circular Costas frequency- staggering across chirp repetition interval, has been proposed to achieve high revisit rates without sacrificing per-pulse bandwidth. SLO-CO processing generates updated unambiguous range- velocity estimates every pulse repetition interval with the potential to achieve better clutter suppression than conventional CAS. This paper studies the reverberation and direct-blast suppression performance of the SLO-CO waveform in a shallow-water environment. A realistic simulation of data from the Target and Reverberation Experiment 2013 (TREX-13) experiment, which took place in the Gulf of Mexico near Panama City Beach, was obtained using the Sonar Simulation Toolbox (SST) developed by APL-UW. A 48-element horizontal array and bi-static mid-frequency acoustic source was considered in an channel with average water depth of 60 m. Real reverberation plus direct-blast data from the actual TREX-13 experiment is evaluated with signal-to-clutter-plus-noise gain (SCNR) with injected target data. Results with real data compare favorably with SST-simulated data and suggest that at least 60 dB suppression of the direct-blast can be obtained using the SLO-CO waveform with conventional beamforming of the horizontal array.

[Special thanks to APL-UW for the use of SST Supported by ONR 321US.]

Multistatic Tracking for Continuous Active Sonar Using Doppler-Bearing Measurements

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Unlike conventional Pulsed Active Sonar (PAS) which listens for echoes in between short-burst transmissions, Continuous Active Sonar (CAS) attempts to detect echoes amidst the continual interference from source(s) transmitting with nearly 100% duty cycle. The potential advantage of CAS is an increased number of continuous detection opportunities, leading to improved target detection, localization, tracking, and classification. The challenge is detecting the target echoes in the presence of continuous interference. CAS transmission waveforms may be of several types, i.e., frequency modulated waveforms (FMs) which provide good range measurements, continuous waveforms (CWs) which provide good Doppler measurements, or sophisticated broadband waveforms which attempt to provide both good range and Doppler measurements simultaneously. In order to mitigate against the multi-source mutual interference problem, it may be preferred to only use continuous CW transmissions, rather than continuously repeating broadband waveforms. This paper develops a tracking approach based on the Gaussian Mixture Probability Hypothesis Density (GMPHD) filter for multistatic sonar configurations using Doppler-bearing measurements from continuous CW transmissions. From a single fixed source-receiver node, such measurements do not enable precise target state estimates. However, when data fusion and target tracking is performed amongst multiple source-receiver nodes, it is shown that good localization estimates and target tracking may be obtained.

Range and Velocity Estimation Methods for Continuous Active Sonar with a Fast Moving Target

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Continuous active sonar (CAS) transmits a continuous predesigned signal, simultaneously receiving echoes which carry information on the target range and radial velocity. Echo information can be processed over the entire transmission cycle. Thus, CAS does not need to transmit at high power, yet it can obtain a high average radiated power when compared to traditional pulsed sonar. Transmission at lower power levels is believed to have reduced impact on marine mammals. Also CAS can deliver a much higher data update rate than conventional pulsed sonar, which has the potential to provide more robust tracking solutions. Therefore, CAS has attracted a lot of interest from sonar waveform designers. Since conventional CAS does not consider Doppler shift, this paper will discuss how to estimate range and velocity in a CAS system. In the paper, we consider two estimation methods: heterodyne processing and matched filter methods. The heterodyne method uses a Continuous Transmission Frequency Modulated (CTFM) waveform. Traditional CTFM processing assumes a stationary or slow moving target. However, for a fast moving target, the Doppler shift creates distortions in the estimated range. We propose a method whereby multiple Doppler shift channels are applied and then we look for minimal variance across all Doppler channels. The matched filter method is based around the use of Costas codes. The whole CAS transmission cycle is divided into eight different Costas waveforms. Each Costas waveform consists of coherently processed frequency hopped pulse trains. Its center frequency is calculated based on Linear Frequency Modulated (LFM) relation. The output has eight channels with each channel signal passed through a predesigned matched filter. The advantages and disadvantages of these two methods will be discussed and compared.

A Multiuser Framework for Sonar Watermarking

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Sonar watermarking is defined as the embedding of a secure, discreet digital signature in the sonar waveform at the source so that authorized receivers are able to authenticate the transmission. The embedding is modeled after spread spectrum where each source uses a dedicated spreading sequence. The existing algorithms work for a single source/receiver configuration. However, real world conditions are a multiuser environment with multiple sources broadcasting waveforms that overlap in both time and frequency. The receiver can be configured as a filter bank where each bank is matched to a specific watermark. However, a filter bank is subject to mutual interference as multiple sonar waveforms may be simultaneously present at the detector input. To mitigate mutual interference, a multiuser watermark detector is formulated as a decorrelating detector that decouples detection among watermark signatures. The acoustic channel is simulated in software using Sonar Simulation Toolset (SST). SST provides a rigorous model for multipath, ocean bottom sediments, surface properties, wave heights, wind speed and Doppler. A channel estimation stage is used to compensate for the degradation of spreading sequences used for watermark embedding. Ambient ocean noise is modeled after Wenz curves controlled by varying sea states. The test statistic generated at the output of the decorrelating detector is used in a joint maximum likelihood ratio detector to establish the presence or absence of the watermark in each received sonar waveform. ROC curves for multiple sources that are broadcasting simultaneously and positioned at varying ranges are produced. Results show that for 30dB signal to watermark ratio and 180dB re 1μPa source level, 90% watermark detection probability is achievable at 5% false alarm rate at ranges up to 7500m.

[A version of this work was presented at the SPIE Defense, Security and Sensing, Baltimore, MD, April 2013.]

Environmentally Sensitive AUV Behaviors for Collaborative Multistatic Surveillance Networks

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The Cooperative ASW Programme at the NATO Centre for Maritime Research and Experimentation (CMRE) is developing multistatic littoral surveillance network concepts that depend on distributed sensing and decision making by autonomous underwater vehicles (AUVs). Since the location of the AUVs dramatically affects the performance of the multistatic autonomous sensor network, optimizing the sensor positions for detection, localization, classification, and/or tracking performance is of primary importance. Signal processing and modeling which account for range-dependent environmental parameters, anisotropic reverberation, signal/array parameters, and hypothesized target depths can give the vehicle an estimate of its expected performance as a function of position and heading and facilitate environmentally-sensitive autonomous vehicle navigation. In general, autonomous behaviors are desired which calculate a given objective function over a large area, then make vehicle navigation decisions which maximize the objective function, taking into account the position and contacts of any available collaborating vehicles. In this work, the expected probability of detection is calculated as a function of the hypothesized target and receiver position as well as all available environmental parameters using the adiabatic normal mode propagation program ARTEMIS. These data can be combined in a Bayesian framework with in-situ detections in the observed acoustic data to update a hypothesized target probability density function. Vehicle navigation decisions are reached using information- and mission-based objective functions. Collaborative strategies are desired which optimize performance over the network rather than a single node. Collaboration is limited by bandwidth available in underwater communication networks. For this reason, globally optimal results are desired which require minimal information sharing between vehicles. A simple but suboptimal collaborative solution is presented here using the position, heading, and five contacts with the highest signal-to-noise ratio (SNR) from collaborating vehicles. Results are presented from the COLLAB13 cruise conducted by the CMRE, as well as simulation results showing the benefit of collaborative autonomous navigation based on the presented method over conventional predefined paths.

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Measuring the Directionality of the Ambient Noise Field from an Autonomous Underwater Vehicle

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For autonomous underwater vehicles (AUVs), the primary method of sensing the local environment is through acoustics. The local noise field contains a wealth of information the AUV uses - from target tracking to communication to general understanding of the environment. An assessment of the spatial composition of the ambient noise field can provide details about the physical environment as well as information for the AUV to incorporate into its control decisions. Measuring the ocean ambient noise field from a towed line array has been done for decades (Wagstaff, 1978). However, for a measure of the ambient noise field to be of use for the AUV it must be measured - and updated - in real-time. The challenge therefore is in quickly and accurately measuring the directionality of the noise field from a single moving line array. Additional complications occur when AUV motion is limited - particularly in pitch and yaw. This talk presents the current successes (and complications) in continuously assessing the directionality of ocean ambient noise measured by an AUV with a towed hydrophone array.

Optimal Joint Channel Estimation and Data Decoding, and Problems with Sequential Analysis in Underwater Communications

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The state-of-the-art in high-rate, single-carrier wideband signaling for underwater acoustic communications is represented by the decision feedback equalizer (DFE). Though often effective, its performance is far from the optimal obtained from soft decision maximum a posteriori probability (MAP) and maximum-likelihood sequence detectors (MLSD). While these algorithms have complexity increasing exponentially with the duration of the channel impulse response, such methods can be used as part of communication system or in combination with other signal processing methods. There are examples of joint channel estimation and data decoding applications for multicarrier modulation, for direct-sequence spread spectrum transmissions, for systems with mode filtering or time-spatial pre-processing equalization applied for time response shortening. We have derived the optimal MAP joint channel estimation and data decoding algorithm in a Gaussian approximation of a posteriori probability and exploit Monte-Carlo numerical simulation for its analysis. The combination of pre-processing, channel response shortening equalization, and joint channel and data recovery have shown excellent performance in shallow water acoustic communications experiments. A sequential estimation alternative to MAP-based decoding is almost as effective in a probability sense, given a modest increase in signal-to-noise ratio (SNR). That approach promises to combine very high performance with a small computational burden relative to the MAP approach. The numerical simulations of simplified variant of sequential estimation has shown that direct application of FANO sequential analysis has poor performance and is too sensitive to modest errors in channel estimator at the front end. When the initial portions of the impulse response fades, the metrics are just too similar and the algorithm needs considerable computation to discover a right branch. To improve performance we suggest combining a MAP decoder for the initial arriving signal portions followed by the (FANO) sequential algorithm applied to the remainder of the message.

Observations of Broadband High Frequency Coherence Degradation in Shallow Water Environments

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We are interested in quantifying the effects of coherence degradation on broadband high frequency acoustic signals through shallow water environments. For this purpose we consider a few canonical models of source-receiver motion on angle delay-Doppler spread channels and compare the coherence degradation that is implied with that observed in real ocean experiments under drifting source-receiver scenarios. It is useful to consider two categories of channel variation, first are acoustic paths whose time-variation rates allow for tracking/estimation. The second category are those rapidly time-varying paths that impart an effective incoherent self noise on the channel. We consider two statistical models of the acoustic Greens function dynamics to quantify the effects of coherence loss on practical underwater acoustic processing systems where the explicit delay Doppler profile is not known apriori; the classic Gauss-Markov model and a recently proposed Gaussian mixture model [Canadian Acous. Vol. 40 No.1 & JASA Vol. 133 No.5]. The models and their associated MMSE estimators are applied to drifting bi-static acoustic recording data and to modeled pressure fields. We provide expressions for the added reverberation noise associated with coherence degradation under an uncertain delay-Doppler structure for each of these models. We discuss the importance of Doppler/dilation estimation and compensation in acoustic processing for the coherent reception of broadband signals and further demonstrate that an upper bound on effective signal to noise ratio is present for these dynamic channels. We show that in the case of a drifting source-receiver scenario in the downward refracting St. Margarets Bay environment of Nova Scotia that upper bound on effective SNR is about 15dB.

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Field Tests of Adaptive Modulation and Coding for Underwater Acoustic OFDM

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Underwater acoustic channels are fast varying and hence the achievable data rates are variable. Adaptive modulation and coding (AMC) technique, which has been widely used in radio communication systems, is applied in underwater acoustic channels. This presentation explores the performance of AMC with an OFDM commercial modem. Five transmission modes corresponding to BPSK, QPSK or 16QAM modulation, and either rate 1/2 or rate 3/4 nonbinary low-density-parity-check (LDPC) channel coding have been implemented, having data rates 1.5, 3.0, 4.5, 6.0, and 9.0 kilo bits per second, respectively. To better predict the data decoding performance, we suggest to use the concept of effective SNR (ESNR). Real time sea trial have been conducted in May 2013, and the performance results will be presented to demonstrate the benefits brought by AMC technique in underwater acoustic communication systems.

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Target versus Clutter Discrimination in Terms of the Wigner Spectrum and the (Symmetric) Ambiguity Function

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Classical signal detection and classification typically models the problem as identifying a signal buried in undesired independent additive noise. Automatic target recognition in active sonar operations has the added difficulty of contending with noise induced by the reflections of the transmitted sonar signal from non-target objects in the channel (i.e., reverberation and clutter). While increased transmit energy will improve performance against independent additive noise, it will not help with induced noise. Moreover, the induced noise is inherently nonstationary. Accordingly, framing the problem in terms of a time-varying spectrum may provide insights into detection and classification of targets in clutter. We examine a standard clutter model in terms of the Wigner spectrum and the ambiguity function, and show how the nonstationarities are manifest. We formulate a target v. clutter classification problem in terms of the ambiguity function and solve for the optimal 2-D filter for the target class. The solution is reminiscent of a Wiener filter, but in the ambiguity domain.

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Target Detection and Classification Against Non-stationary Interference Using Dynamic Feature Clustering

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Target detection and identification is challenging in active sonar due to severe reverberation interference caused by the transmitted signal reflecting off the sea surface, sea bottom, and volumetric scatterers like fish. Other issues include target echo distortion by multipath channels, and Doppler effects. Nearly all active sonar systems employ matched filter processing followed by normalization, which not only ignores non-stationarity issues but also performs poorly when the target has a complicated impulse response convolved with a multipath channel. Our objective is to track the sonar channel in real time as the interferers, environmental, anthropogenic or biological, shift in their characteristics in unpredictable non-stationary directions.

Although the target echo itself may exhibit a sparse representation, sparse optimization techniques typically perform poorly in target detection against non-stationary interference due to four overlapping factors: (i) Rapidly time-varying non-target interference due to non-stationary reflections from the environment and other non-target surfaces, (ii) Time-varying sparseness of energy of these reflected components, (iii) Localization uncertainty of non-stationary interference, (iv) Failure of any sparse optimization technique to detect localized information beyond the main peaks. From a target classification perspective, the last factor is a key motivator, as smaller components that cluster around the principal components record the "rise" and "fall" of target echoes thereby, play an important role in fingerprinting target reflections against non-target interference.

We employ a combination of optimization, feature localization, interference cancellation and non-uniform sampling techniques to enable target detection and classification beyond traditional matched filter processing. In particular, we propose dynamic feature localization techniques that can adapt to varying levels of sparseness of a physical process and thus detect smaller components that cluster around key significant components. By capturing clustered features, we move beyond popular sparse sensing techniques and address sparseness more generally as a changing property of a time-varying distribution rather than optimizing towards only the principal components. Furthermore, we propose an adaptive subspace navigation algorithm for dynamic target tracking and classification. We also investigate interference mitigation techniques to distinguish between feature clusters from target and non-target sources, and thus enable target detection in low SNR scenarios. Numerical results based on simulated environments built upon ground truths provided by shallow water acoustic field data are presented.

[James Preisig (Woods Hole Oceanographic Institution) provided the shallow water acoustic field data.]

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Applying a Featureless Classifier in Non-Stationary Generalized Pareto Distributed Clutter

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Active sonar clutter is generally non-stationary and non-Gaussian. The non-Gaussian behavior is most apparent in the heavy upper tail of the clutter distribution. There are typically more high level threshold crossings than would be expected in the Gaussian case. Recent experience has shown that this upper tail of the clutter level distribution is often well modeled by a power law, such as found in the Generalized Pareto Distribution (GPD). A featureless classifier has previously been applied to active sonar. The featureless classifier uses an adaptive coherence estimator (ACE) applied to snippets of data around each potential high level target. The ACE is an optimal process, however, that optimality is based on the assumption that data within each snippet has a Gaussian distribution. Approaches were developed to modify the ACE to perform optimally in the presence of non-Gaussian clutter. We investigated a non-linear transformation of sonar clutter data to Gaussian, so that the featureless classifier can be optimally applied. The non-linear transformation taking GPD data to Gaussian requires that the GPD parameters be accurately estimated. The conventional maximum likelihood method for parameter estimation assumes that clutter points are independent, but data often show significant correlation. We developed and investigated a new method of log moments for the GPD, to better estimate parameters in correlated clutter. The estimation method is applied to clutter examples taken from the NATO Base 04 exercise.

[Supported by ONR.]

Feature-Aided Tree-Search Tracking on the CLUTTER09 Dataset

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When active sonar systems are used in shallow water environments, a large number of clutter contacts are produced in addition to the desired target contacts. Heavy clutter in combination with a nonlinear observation model and low target detection probability makes tracking multiple targets in these environments particularly challenging. Recently, physical features of the return signals have been considered as additional information to improve tracker performance by discriminating between target-generated and clutter-generated returns. In our work, we incorporate physical features within the Bayesian tree-search tracker. We evaluate the effectiveness of these features in classifying returns, as well as the robustness of the features across scans and environments.

The tree-search tracker uses a search tree to represent the target state space; the tracker navigates the tree to identify the most likely sequence of states visited by the target(s). The search for new targets and the tracking of the existing targets are governed by a path metric that is proportional to the posterior probability density of the target state given the observed contacts. Our approach to using features in tree-search tracking is to incorporate feature information in the calculation of the path metric by augmenting the observation vector. When features help classify targets vs. clutter, this approach boosts the metrics of paths that follow target returns. The ability of the features to discriminate between target and clutter returns is studied through principal component analysis (PCA) and linear discriminant analysis (LDA). Using the PCA or/and LDA projection matrix, the given features are transformed to a new feature space with significantly reduced dimensionality and increased robustness across scans. The statistics of the projected features are approximated and used in the calculation of the observation likelihood element of the path metric.

The performance of the proposed technique is evaluated on data from the CLUTTER09 experiment, which was performed by the NATO Undersea Research Center (NURC) in the Malta Plateau on May 7-8, 2009. Results show that the PCA projection matrix derived from all contacts in the first two scans of the May 7 experiment provides significant separation between the clutter and target-generated contacts throughout the May 7 experiment. Results also show that while the LDA approach provides improved performance when extensive training is available, PCA is considerably more robust to reduced and/or imperfect training. As a final step, we present an algorithm that combines PCA for reducing clutter with LDA for discriminating among targets; this approach successfully prevents track coalescence between adjacent targets in the May 7 experiment.

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Detection Performance of Coprime Sensor Arrays

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Coprime sensor arrays (CSAs) achieve the resolution of a fully populated uniform linear array (ULA) with the same aperture using fewer sensors. However, since a CSA has fewer sensors than a fully populated ULA with the same aperture, the CSA's ability to attenuate white noise is also less. Consequently a CSA has less array gain than a full ULA with equivalent aperture. Previous publications on CSA arrays [Vaidyanathan and Pal, IEEE Signal Processing, 2011] have emphasized the important benefits of high resolution from fewer sensors but have not considered the possible performance costs when operating in a noisy environment. This research compares the detection performance of a CSA against a full ULA when the arrays are operating in an environment with spatially uncorrelated noise. Assuming a narrowband signal independent of spatially white noise, and that both the signal and noise are modeled by zero-mean, circular complex Gaussian distributions, the detection statistic for the ULA is the sum of the squares of the absolute values of the real and imaginary parts of the array output. Hence, the distribution for the ULA has a Chi-square distribution with two degrees of freedom. Since the real and imaginary parts of the array output are independent with equal variances, the detection statistic distribution simplifies to an exponential distribution with its parameter dependent on the input SNR. The detection statistic for the CSA is the product of the output of one subarray with the complex conjugate of the output of the second subarray. Being the product of two independent complex Gaussian random variables, the CSA detection statistic has a distribution proportional to a modified zeroth order Bessel function of the second kind [O'Donoughue and Moura, IEEE Signal Processing, 2012] with its argument dependent on the input SNR. Manipulating the distributions for the detection statistics provides analytical expressions for probabilities of detection and false alarm and also mean discriminating information for both ULA and CSA. These expressions simplify to exponential functions for ULA and they can be evaluated directly. The probabilities of detection and false alarm can be computed using numerical integration techniques and the mean discriminating information can be calculated using simulations for the CSA. Evaluating these expressions confirms that a CSA pays a detection gain penalty of about 5 dB over a wide range of SNRs and array sizes.

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A Track Before Detect Method with Motion Model

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A Track Before Detect (TBD) method using motion model is proposed for detecting very low SNR targets. This method is dierentiated from Dynamic Programming (DP) or Particle Filter (PF) in that it evaluates the likelihood for the entire hypothesized motion of the target using a motion model in contrast to the sequential state update in DP and PF. Instead of the recursive state updates used in DP and PF, this method employs a random search of the motion of the target for the entire duration of the observation. For each trial, a random motion of the target is hypothesized and the motion is then used to process the raw observation to coherently integrate the target echo energy. Using multiple transmissions, the coherent processing can be thought of as beamforming an irregular array with delays due to the hypothesized motion. This motion and the peak value of the processed time series is used the compute the likelihood. The hypothesized motion trial that yields the maximum likelihood is chosen as the estimated motion of the target. The likelihood value is compared to a threshold and the range to the target is inferred from the position of the peak. Therefore, track detection and track estimation is done simultaneously. The motion model allows the random search to be computationally practical. With brute force motion search, the search space grows exponentially with the number of transmissions. Here, a simple motion assumption is made using the Hidden Markov Model. With motions with relatively small variation in speed and heading represented by sequence of states, we show that the number of trials required to try the correct motion is computationally viable when the assumed motion model is similar to the true model that governs the motion. Comparison of the proposed method to existing TBD methods is shown in terms of the probability of detection of track against SNR of transmission. While the recursive method fails to accumulate enough energy to make a reliable state estimates for very low SNR targets, the proposed method is able to coherently integrate the target echo energy and detect the true track at lower SNRs.

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Wednesday **October 16, 2013**		**Thursday** **October 17, 2013**		**Friday** **October 18, 2013**	
		8:30–9:45	Session B CAS I Laurel	8:15–9:30	Session F CAS III Laurel
				9:30–10:20	Session G AUV Signal Proc. Laurel
		9:45–10:15	Break Laurel	10:20–10:45	Break Laurel
		10:15–11:55	Session C Sensing & Noise Laurel	10:45–12:00	Session H UW Acoustic Comm. Laurel
		12:00–1:00	ASA Lunch Whisp. Pines	12:00–1:00	Lunch Whisp. Pines
		1:00–3:05	Session D Array Proc. Laurel	1:00–3:30	Session I Detection, Classification, and Tracking Laurel
		3:05–3:30	Break Laurel		
		3:30–5:10	Session E CAS II 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		
8:00–9:45	Session A Tufts Memorial Laurel	8:00–?	SOB Session Laurel		