A Book of Abstracts for the

2017 Underwater Acoustic Signal Processing Workshop

October 4-6, 2017

Alton Jones Campus

University of Rhode Island

West Greenwich, RI, USA

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Welcome to the 2017 IEEE workshop on Underwater Acoustic Signal Processing. This year the special session, organized by Ashwin Sarma, is titled **Revisiting Assumptions and Approximations in Structured Signal Processing**.

The organizing committee thanks and acknowledges the continued support of our promotional partners, the Office of Naval Research, Raytheon Integrated Defense Systems, and the Acoustical Society of America. We thank Michael Janik for his efforts in arranging for Raytheon Integrated Defense Systems to sponsor our Wednesday evening awards dinner. The Acoustical Society of America is sponsoring Thursday evening's dinner and an award for the best student presentation. The winner of the student contest will receive $500 towards travel to a future ASA meeting. Finally, we are proud to announce that this year's recipient of the Donald W. Tufts UASP Award is Dr. Bruce K. Newhall.

The Organizing Committee

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Special Session Organizer

Ashwin Sarma
BAE Systems/URI

The 2017 Donald W. Tufts UASP Award is presented to Dr. Bruce K. Newhall

for contributions to underwater acoustic signal processing, noise and reverberation modelling, and scientific leadership in sonar experimentation

Bruce K. Newhall received the B.S., M.S., and Ph.D. degrees in mathematics from Rensselaer Polytechnic Institute, culminating in 1980 with research on the effect of ocean current fluctuations on sound. In 1980, Dr. Newhall joined the Johns Hopkins University Applied Physics Laboratory (JHU-APL) where he remained until retiring in 2017. Throughout his career, Dr. Newhall contributed to the full scope of research and development in underwater acoustic signal processing, including theoretical modeling, algorithm development, and at-sea scientific experimentation. He is known and has been recognized in the broader scientific and technical community for his contributions to array shape estimation, ambient noise and reverberation modeling, and active waveform design. His expertise on the theoretical side is complemented by his significant contributions in scientific experimentation, which undoubtedly explains why his work is so highly relevant. As the chief scientist of a number of large, multiple-institution, multiple-platform, Navy experiments, Dr. Newhall's efforts were instrumental in the advancement of undersea surveillance sonar. Dr. Newhall has long been a scientific leader, mentoring the next generation of scientists and engineers at JHU-APL and lending his trusted voice to the needs of science management both internal and external. Dr. Newhall's leadership also extended to several line-management positions at JHU-APL and as an associate editor with the IEEE Journal of Oceanic Engineering.

For his many contributions to the field, the underwater acoustic signal processing community is pleased to present the 2017 Donald W. Tufts UASP award to Dr. Bruce K. Newhall.

Schedule at a glance

Wednesday October 4, 2017		Thursday October 5, 2017		Friday October 6, 2017	
		8:15–9:30	Session C Spec. Sess. II *Laurel*	8:40–9:30	Session I Acous. Inf. Proc. *Laurel*
		9:30–10:20	Session D Non-Coherent Detect. *Laurel*	9:30–10:20	Session J Adaptive Proc. *Laurel*
		10:20–10:45	Break *Laurel*	10:20–10:45	Break *Laurel*
		10:45–12:00	Session E Adapt. Beamforming *Laurel*	10:45–12:00	Session K Spec. Sess. IV *Laurel*
		12:00–1:00	Lunch *Whisp. Pines*	12:00–1:00	Lunch *Whisp. Pines*
		1:00–2:15	Session F Spec. Sess. III *Laurel*	1:00–1:50	Session L Acoustic Comms. *Laurel*
		2:15–3:05	Session G Waveform Design *Laurel*	1:50–2:05	Break *Laurel*
		3:05–3:30	Break *Laurel*	2:05–3:20	Session M Sparse Arrays *Laurel*
		3:30–4:45	Session H Tracking & Positioning *Laurel*		
5:00–6:00	Welcome Reception *Whisp. Pines*	5:00–6:00	Reception *Whisp. Pines*		
6:00–8:00	Dinner *Whisp. Pines*	6:00–8:00	Dinner *Whisp. Pines*		
8:00–9:30	Session A Plenary Session B Spec. Sess. I *Laurel*	8:00–?	SOB Session *Laurel*		

Sessions: Titles and presenters

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

Plenary presentation by the 2017 Donald W. Tufts UASP Award recipient

A–1 *Revisiting Assumptions in Underwater Acoustic Signal Processing: A Personal Historic Perspective*
Bruce K. Newhall

Session B: Wednesday Evening, 9:00pm–9:30pm

Special Session I: Revisiting Assumptions - Modeling

B–1 *Recent Results in Parameter Estimation under Model Mismatch/Misspecification*
Christ Richmond, Arizona State University

Session C: Thursday Morning, 8:15am–9:30am

Special Session II: Revisiting Assumptions - SVD Usefulness

C–1 *Maximal Invariants and the Singular Value Decomposition in Detection Theory*
Louis Scharf, Colorado State University

C–2 *A Differential Geometry Based Framework for Constrained Subspace Estimation*
Thomas Palka, Raytheon Integrated Defense Systems

C–3 *The Role of Subspace Estimation in Sensor Array Signal Processing*
Richard J. Vaccaro, University of Rhode Island

Session D: Thursday Morning, 9:30am–10:20am

Non-Coherent Detection

D–1 *Performance Analysis of Constant-False-Alarm-Rate Detectors using Characteristic Functions*
Douglas Abraham, CausaSci LLC

D–2 *Generalization of the Edgeworth and Gram-Charlier Series and Application to Underwater Acoustics*
Leon Cohen, City University of New York

<u>Session E</u>: Thursday Morning, 10:45am–12:00noon

Approximate Adaptive Beamforming

E–1 *Unit Circle Constrained Beamforming for Structured Covariance Matrices*
Colin Ryan, MIT Lincoln Laboratory

E–2 *Phase Detections with Low Complexity Adaptive Beamforming on Swath Sonars*
Tor Inge Birkenes Lønmo, Universitetet i Oslo

E–3 *Use of Basis Pursuit for Detecting Weak Sources Among Strong Interferers*
Tom Northardt, MIKEL Inc.

<u>Session F</u>: Thursday Afternoon, 1:00pm–2:15pm

Special Session III: Revisiting Assumptions - Environmental Realities

F–1 *The Occurrence of Lucky Scintillations in HLA data from the Shallow Water 2006 Experiment (SW06) and Optimum Array Processing Strategies*
Ivars Kirsteins, Naval Undersea Warfare Center

F–2 *The Validity of the Frozen-Surface Approximation for Large Time-Bandwidth Signals*
Paul Hines, Dalhousie University

F–3 *Internal Wave Limits to Matched Field Processing*
Arthur Baggeroer, Massachusetts Inst. of Technology

<u>Session G</u>: Thursday Afternoon, 2:15pm–3:05pm

Waveform Design

G–1 *A Steganographic Approach for Active Sonar Track Detection*
Daniel Park, Applied Research Laboratory, Penn State University

G–2 *Active Sonar Waveform Design using Multi-Tone Sinusoidal Frequency Modulation*
David Hague, Naval Undersea Warfare Center

Session H: Thursday Afternoon, 3:30pm–4:45pm

Tracking and Positioning

H–1 *Multipath Tracking in a Shallow Water Environment with ML-PMHT*
Sean Walstead, NUWC Newport

H–2 *Inference Regarding the State of Mobile Underwater Object from a Small Aperture Vertical Array*
Abner Barros, University of Massachusetts Dartmouth

H–3 *Passive Acoustic Navigation and Command of a Low-Cost Autonomous Underwater Vehicle*
Nicholas Rypkema, Massachusetts Institute of Technology

Session I: Friday Morning, 8:40am–9:30am

Acoustically Informed Processing

I–1 *Decoupling Depth and Range Estimation using Acoustic Invariant Theory*
Paul Hursky, Sonar-synesthetics LLC

I–2 *Model-Based Signal Processing for Sediment Property Inversion using Group Velocities*
Zoi-Heleni Michalopoulou, Department of Mathematical Sciences, New Jersey Institute of Technology

Session J: Friday Morning, 9:30am–10:20am

Adaptive Processing

J–1 *Adaptive Doppler-Azimuth Processing for Comb Signals using Steered Covariance Matrices*
Jonathan Soli, Duke University

J–2 *Adaptive Range Processing for Active Sonar*
Travis Cuprak, L3 Adaptive Methods

Session K: Friday Morning, 10:45pm–12:00noon

Special Session IV: - Revisiting Assumptions - Comms and Systems

K–1 *Characterization of Communication Channel Conditions over Varying Ranges as Long as 10 km*
Jean-Francois Bousquet, Dalhousie University

K–2 *Relative Impacts of Signal-to-Noise Ratio and Propagation-Induced Distortion on Passive Acoustic Monitoring of Cetaceans*
Carolyn Binder, Defence Research and Development Canada

K–3 *A Goal Reasoning Based Architecture for Intelligent Active Sonar*
Jill Nelson, George Mason University

Session L: Friday Afternoon, 1:00pm–1:50pm

Underwater Acoustic Communications

L–1 *M-ary orthogonal spread spectrum signaling for very low signal to noise ratio acoustic communications*
Paul Gendron, University of Massachusetts Dartmouth

L–2 *Coherent OFDM Receiver for High Data Rate Acoustic Communication Systems*
Sayedamirhossein Tadayon, Northeastern University

Session M: Friday Afternoon, 2:05pm–3:20pm

Sparse Arrays and Processing

M–1 *Analysis of Linear Co-prime Array Geometries in a Continental-Shelf Ocean Environment*
Juan Ramirez Jr., ASEE-NRL Post-Doctoral Fellow

M–2 *Spatial Correlation Resampling for Wideband Source Enumeration and Direction-of-Arrival Estimation on Sparse Arrays*
Yang Liu, University of Massachusetts Dartmouth

M–3 *Performance Prediction of Coprime Sampled Arrays in Spatially Correlated Noise*
Radienxe Bautista

Abstract Listings

Revisiting Assumptions in Underwater Acoustic Signal Processing: A Personal Historic Perspective

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Selected topics in underwater acoustic signal processing from the author's 40 year career will be discussed, with a primary focus on ONR funded efforts in towed array performance and active sonar. Fortuitous events which led the author to work in signal processing are described. Early towed array performance was assumed to be limited by propagation through ocean inhomogeneities, however later efforts showed that the varying non-linear shapes of towed arrays had a much larger impact. Beamforming that compensated for those array shapes for SURTASS 3X and earlier towed arrays was developed and led to significant gains in performance. Matched field processing is based on an assumption of improved signal modeling. Reasons for the failure of matched field to achieve its potential are given. Topics from early low-frequency active acoustics will include waveform design for improved Doppler response from bottom reverberation. More recent work in mid-frequency active clutter simulation and classification are included. The Gaussian assumption for clutter distribution is revisited.

Recent Results in Parameter Estimation under Model Mismatch/Misspecification

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Ground zero for signal processing efforts is typically adopting a statistical model for the acquired data that conveys some structured relationship between the data measurement and a set of parameters carrying desired information. When the adopted model is correct, then fundamental limits exist for detection and parameter estimation performance that establish what signal processing can and cannot accomplish with the data. Due to the practicalities of science and engineering, however, assumed models are rarely perfectly specified. Thus, mismatch is often present in some form, and the model is said to be misspecified. This talk will (i) review some challenges in specifying and exploiting structured data models, (ii) remind of the value of sensitivity analysis, (iii) discuss new results for bounding parameter estimation under model misspecification, and (iv) outline possible paths forward toward robustness. Specific examples in array processing including direction-of-arrival estimation will be discussed.

Maximal Invariants and the Singular Value Decomposition in Detection Theory

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In this talk I will frame a number of multi-pulse, or multi-snapshot, detection problems which are relevant to underwater acoustic signal processing. For each problem I will establish the transformation group which leaves the detection problem invariant, and establish its maximal invariant statistics. Sometimes these are singular values of a data matrix, and sometimes they are not. In any case, the principal of ordinary likelihood, termed generalized likelihood in the engineering literature, will be used to derive a detector statistic that is a function of these maximal invariants, and is therefore also invariant to the transformation group.

One large class of detection problems may be framed as tests for space-time *patterning* in a mean-value matrix or covariance matrix. These include null hypothesis tests for patterning and factor analysis for low-rank structure. For these, singular values of a data matrix or singular values of a coherence matrix typically emerge as maximal invariants. But the resulting detectors, which apply to coherence detection in space-time fields, are more complicated functions of the maximal invariants than you might imagine. Another large class of detection problems may be framed as tests for space-time *parameterization* of a mean-value matrix or covariance matrix. For these I will try to establish the transformation group that leaves the detection problem invariant, and to establish its invariant statistics. I will speculate that recent work by Palka and Vaccaro on the geometry of second-order perturbations for parameter estimation may be adapted to the problem of constructing ordinary likelihoods in *parametric* models for propagating waves.

[This work is supported by the National Science Foundation under contract CCF-1712788, and by the Air Force Office of Scientific Research under contract FA9550-14-1-0185.]

A Differential Geometry Based Framework for Constrained Subspace Estimation

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Subspace estimation appears in a wide variety of signal processing applications such as radar, communications, and underwater acoustics, often as a prelude to high resolution parameter estimation. The parameter space in general subspace/basis estimation problems is naturally described as a Riemannian quotient manifold. This identification permits the well-developed tools of differential geometry to be brought to bear on the analysis of subspace/basis estimation problems. Using this differential geometric framework, we derive an intrinsic CRB on subspace estimation accuracy applicable to signal processing applications. The relationship between this new bound on constrained subspace estimation accuracy and the existing bounds developed for system parameters is established and its relevance to the high resolution parameter estimation problem is explored in detail. This development naturally suggests an asymptotically ML subspace estimation approach for the constrained model that leverages the sub-optimal subspace estimate provided by the standard EVD. By the maximum likelihood (ML) invariance principle, this subspace estimator yields an asymptotically ML estimator of the desired parameters of the usual signal model. An efficient implementation of this general approach is developed for the important special case of the uniform multi-dimensional array and complex exponential waveform model. Utilizing the shift-invariant properties that define the constraint in this setting, we derive a closed-formed estimator of the signal subspace. This new estimation approach is applied to the challenging 2-D Direction-of-Arrival (DOA) estimation problem and a set of scenarios drawn from the literature are evaluated to demonstrate performance.

The Role of Subspace Estimation in Sensor Array Signal Processing

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Subspace estimation is an implicit or explicit part of signal processing tasks such as adaptive beamforming or direction-of-arrival estimation. The signal subspace for such problems is usually estimated by calculating the eigenvector decomposition (EVD) of the sample covariance matrix. The justification for using the eigenvectors is that, as the number of snapshots grows large, the span of the principal eigenvectors converges to the true signal subspace. That is, the eigenvectors are a consistent estimator of the signal subspace. However, a more useful property than consistency is finite-sample accuracy, as measured by statistical efficiency. That is, for a finite number of snapshots, how accurately can the signal subspace be estimated? A bound on subspace accuracy has recently been derived and the sample eigenvector subspace does not achieve it. A new, closed-form, optimal subspace estimation (OSE) algorithm that achieves the bound for 1-D and 2-D uniform arrays has been derived. Signal processing tasks that use the OSE subspace inherit the benefits of its best achievable accuracy. However, the OSE calculations are prohibitive for arrays with a large number of sensors and the OSE subspace estimate is sensitive to perturbations such as array calibration errors. Both of these drawbacks are addressed by a newly proposed estimation method based on subspace averaging (SSA). The SSA subspace estimate is suboptimal with respect to accuracy but is better than EVD and can approach that of OSE with less computation. This talk will present the EVD, OSE, and SSA subspace estimators and compare them with respect to accuracy, computation, and sensitivity to perturbations.

Performance Analysis of Constant-False-Alarm-Rate Detectors using Characteristic Functions

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Constant false alarm rate (CFAR) detectors in sonar are typically implemented by forming the ratio of an unnormalized decision statistic from a test cell to an estimate of the noise background power and comparing it to a decision threshold. Evaluation of the performance in terms of the detection and false alarm probabilities is commonly done through the use of the probability density function (PDF) of the background power estimate and the cumulative distribution function (CDF) of the unnormalized test cell. An alternative approach is presented where the PDF, CDF, and other properties of the CFAR decision statistic are obtained as a function of the characteristic functions (CFs) of the unnormalized test cell and the background power estimate. The CF-based approach is appealing because of its generally simpler form compared to the PDFs and CDFs of both the background power estimator and standard signal models (deterministic, Gaussian fluctuating, Rician, and gamma-fluctuating intensity). The technique is demonstrated through several examples including background power estimators formed using a weighted average (e.g., an exponential averager), an order-statistic filter, and correlated Gaussian data. The broad utility of the approach is illustrated by evaluating false alarm and detection performance of a normalizer for a Gamma-fluctuating-intensity signal in heavy-tailed clutter and by assessing how much non-stationarity in the background noise power is required for normalizing before incoherent integration to perform better than normalizing after.

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Generalization of the Edgeworth and Gram-Charlier Series and Application to Underwater Acoustics

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The main historical purpose of the Edgeworth and Gram-Charlier series is to correct a Gaussian distribution when new information is given, such as moments, which do not match the Gaussian. These methods relate a probability density to a Gaussian by way of an operator transformation that is a function of the differentiation operator. We generalize the standard series in two ways. First, we relate any two probability densities using the differentiation operator. More generally, we show that any two probability densities may be related by an operator transformation, which is a function of a Hermitian operator. In particular, we show that any two probability distributions $P_1(x)$ and $P_2(x)$ are related by

$$P_2(x) = \Omega(\mathbf{A}) P_1(x)$$

where $\mathbf{A}$ is *any* Hermitian operator and Ω is a function whose explicit form is known and depends on the eigenfunctions of the operator.

We apply these methods to the reverberation problem by showing how to correct for the Rayleigh distribution. We also show that these methods can be used for pulse and noise propagation. For these cases, the density, $P(x,t)$, is a function of space and time, and we show that it is related to the density at time zero, $P(x,0)$ by

$$P(x,t) = \Omega(\mathbf{A}) P(x,0)$$

Unit Circle Constrained Beamforming for Structured Covariance Matrices

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Adaptive beamforming is an array processing technique that enhances the detection of signals by using data covariance to null interference. In time-varying environments, insufficient sample support for the covariance estimate limits performance. The recently proposed Unit Circle Diagonally Loaded beamformer improves performance by leveraging properties of the optimal array weights to use as constraints in a post-adaptive-beamformer, processing stage. However, the improved performance comes at a high computational cost. This paper revisits methods to enforce such constraints directly on the covariance matrix and their relation to unit circle analysis. Superior interference rejection is demonstrated on simulated data for low computational cost.

Phase Detections with Low Complexity Adaptive Beamforming on Swath Sonars

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Low Complexity Adaptive (LCA) beamforming is a restricted Minimum Variance Distortionless Response (MVDR) beamformer with the same optimality criteria and a discrete search space. It runs multiple weighted delay and sum (DAS) beamformers, some *microsteered*, and selects the one that minimizes the output power. The weight selection is inspired by the weights MVDR finds optimal in various cases. It was first suggested as a robust and low computational cost version of MVDR in medical ultrasound, and has since then shown promising results for sonars. The reduced search space makes it possible to get good results with single sample covariance matrix estimates, even without spatial smoothing and diagonal loading. Although to preserve speckle statistics, temporal averaging is needed.

We have applied LCA to a seabed mapping swath sonar to improve the accuracy of bottom detections and the quality of the water column image. Current results show that LCA gives a large increase in edge definition, leading to smaller points and sharper edges, and smaller improvements in resolution and sidelobe suppression. This translates to up to large improvements in the accuracy of amplitude detections, one of the two main detection methods for a swath sonar. It uses the center of gravity of the received envelope. Phase detection is the other method. It looks for the zero phase difference instant between two subarrays. LCA currently reduces the phase detection accuracy compared to DAS. This is because the phase difference becomes more irregular with LCA.

This talk will discuss why phase detections deteriorate with LCA and possible corrections. One possible explanation is a footprint shift effect. The microsteered weights have a non-symmetric response around the steering axis, which may lead to a shift in the phase center of the footprint. This is consistent with the problem disappearing for point targets and increasing with longer pulses. Initial results indicate that this effect also applies to MVDR.

Use of Basis Pursuit for Detecting Weak Sources Among Strong Interferers

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Detecting weak sources in the presence of strong interferers is a common objective in sonar surveillance and warfare scenarios. Often the direction-of-arrival (DOA) of strong sources is easy to estimate even without advanced beamforming techniques and is readily available information. Adaptive beamformers such as the linear constraint minimum variance (LCMV) are commonly employed to steer multiple nulls to the location of the known interferers. But LCMV, has a few limitations; the number of nulls is restricted to the number of array elements, the required number of array observation samples is high, and sidelobe levels near nulls can be quite high compared to the weak sources. In contrast, popular Basis Pursuit-based (BP) algorithms for DOA estimation offer highly resolved spatial spectrums and require very few array observations. However, BP-based algorithms, by design, only estimate the strongest contributors to the array observations. In this work, the array data is pre-conditioned using a simple orthogonal projection and a Basis Pursuit De-Noising (BPDN) algorithm is used to subsequently estimate the spatial spectrum after data conditioning. Spatial spectrum estimation performance is evaluated with simulated scenarios and compared to LCMV using pre-conditioned and unconditioned array observations. Though simple, the approach produces good results and is pragmatic.

The Occurrence of Lucky Scintillations in HLA data from the Shallow Water 2006 Experiment (SW06) and Optimum Array Processing Strategies

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A commonly held assumption in underwater acoustic array processing is that increasing sample support, i.e. the number of data snapshots, to calculate sample covariance matrices always improves performance in detection and localization. That is, each extra data snapshot must add more information to the problem to be solved. In Gaussian case, this is true when the signal and noise data snapshots are independently distributed and temporally stationary. However, this may no longer be the case when the signal wave fronts are distorted by time-varying random medium effects like internal waves. Wave front distortions can cause large estimation errors, biases, and loss of array gain for algorithms derived under ideal assumptions. Under these circumstances increasing sample support generally does not improve estimation performance since the signal wave fronts are still distorted on average.

In this paper we present an analysis of horizontal line array (HLA) data from the Shallow Water 2006 experiment provided by the Woods Hole Oceanographic Institute. Our analysis results give strong evidence that lucky scintillations occur regularly at shorter time scales even during periods of internal wave activity. By lucky scintillations we mean short moments when the instantaneous signal wave front is relatively undistorted even though at longer time scales on average the wave fronts appear to be distorted. This suggests an alternative array processing strategy where each data snapshot is collected over a time interval matched to the ocean random medium time scales and first ranked in quality, i.e., level of signal wave front distortion, with only the highest ranked snap shots utilized by the estimator. In addition, an adaptive iteration scheme is developed for estimating the steering vectors associated with a data coherence matrix across pairs of subarrays.

The Validity of the Frozen-Surface Approximation for Large Time-Bandwidth Signals

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The physical modeling of underwater acoustic propagation, scattering, reflection, and reverberation, and the signal processing associated with these processes, usually rely on the so-called frozen-surface approximation. This longstanding approximation, which dates back to Carl Eckart's seminal paper (The Scattering of Sound from the Sea Surface, 1953), assumes that the ocean surface can be safely modeled as if it is frozen in time throughout the entire pulse duration. As he states it, *"Thus far, it has been tacitly assumed that the (surface roughness) is independent of the time t. Presumably, if the sea surface does not change too much during the time...so that (surface roughness) is a slowly varying function of t, then (the model) will remain valid to the degree of approximation here achieved."* The approximation is valid for the short-duration pulses typically used by sonars in the decades following that paper; however, its applicability to present-day high duty cycle sonars has not been confirmed. In fact, there seems to be few papers in the peer-reviewed literature that have explicitly examined its accuracy even for short-duration pulses. Although the assumption is based on the physics of the problem, it can have a profound effect on the signal processing, especially for large time-bandwidth signals. Put another way, the signal processing can inform how accurately the surface must be modeled. In this paper some experimental results will be presented to provide examples of how the approximation fails for a long duration, wideband pulse. Using this as motivation, the authors will explore some areas where incorrectly employing the approximation can introduce errors in expected signal processing gains in underwater acoustic systems. While no attempt will be made to correct shortcomings in the approximation, it is hoped that the discussion may motivate the community to examine these issues in its future research.

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Internal Wave Limits to Matched Field Processing

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Matched Field Processing (MFP) depends upon using a coherent replica across the aperture. Internal waves (IW) lead to coherence loss which limits useful apertures using rank one processing, so aperture size must be limited or high rank processing must be used. Essentially, the mean signal rapidly decorrelates into a set of random orthogonal components propagating in a cone. Simulations and path integrals evaluations have been used to establish these aperture limits, which are functions of source/receiver depths, frequency and range. [Baggeroer and Colosi, *JASA* 140(4,Pt 2), 3024, 10/16] All scenarios may be described to be in "unsaturated" realm. The frequency dependence has the expected trend towards "saturation" as it increases. Deep sources also trend towards saturation compared to shallower one. Earlier papers on MFP established three types of mismatch: environmental (SVP errors), system (equipment locations and calibrations), and stochastic (snapshot) limitations. [Baggeroer, *et al*, *JASA*, 82(2), 571-589, 2/88] IW represent a fourth, a combination of environmental and stochastic mismatch. This presentation examines performance degradation due to IW's and the gains achievable by higher order processing paralleling concepts where the parameter is embedded in a random signal. Subsequent work intends to apply Cramér-Rao Bounds. [Baggeroer, *JASA*, 141(5), 3430-3451, 5/17]

[Work supported by ONR Codes: Undersea Signal Processing and Ocean Acoustics]

A Steganographic Approach for Active Sonar Track Detection

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Conventional active sonar for tracking is done by transmitting an acoustic pulse and processing the return echo to make detections, repeated for multiple pulses. The sequential connections of the detections and associated range estimates form a track. Under steganographic constraints, the return echoes are so low that individual detections cannot be reliably made without incurring more false detections. A batch track detection approach using a long timescale motion model is used to detect tracks with low SNR echoes that conventional tracking and track-before-detect methods fail to detect. For transmission waveforms, linear frequency modulation is typically used for conventional active sonar, which generates sounds that are conspicuous in the underwater ambient environment. Steganographic security of transmission waveforms is estimated by comparing the statistical models of the ambient sound and the sound perturbed by the transmission waveform. A sound modeling approach using hidden Markov model with symbolized phase space representation of time series is used for this comparison for candidate waveforms with different spectral characteristics.

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

Active Sonar Waveform Design using Multi-Tone Sinusoidal Frequency Modulation

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Over the past 60 years there has been a wealth of research in choosing the optimal waveform to transmit for radar/sonar systems. Efforts in the published literature have largely focused on designing waveforms that possess a desired Ambiguity Function (AF), Auto-Correlation Function (ACF), or Cross-Correlation Function (CCF) [Wilcox: Math. Res. Center Rpt. 157 (1960), Li et al : Camb. Univ. Press (2012)]. Waveforms for active sonar must additionally be well suited for transmission on piezoelectric transducers. This typically requires the waveform to possess a constant amplitude and high Spectral Efficiency (SE) where the vast majority of the waveform's energy is contained in the band of operational frequencies. Designing constant amplitude waveforms that are jointly optimized to possess a desired AF/ACF shape and high SE is a problem that has only recently been addressed in the published literature for radar waveforms [Blunt et al: IEEE Trans. Aero. Sys. (2014)]. This research explores this challenging design problem for active sonar using Multi-Tone Sinusoidal Frequency Modulation (MTSFM). The MTSFM waveform s modulation function is a finite sum of weighted sinusoidal functions expressed as a Fourier series. The Fourier coefficients act as a tunable set of parameters that may be adjusted to synthesize a waveform with desired AF/ACF shapes. Utilizing a finite number of Fourier coefficients produces a modulation function that is smooth and continuous in time which results in a waveform with high SE. Simulations demonstrate that the MTSFM waveform s design coefficients can be adjusted to finely control the waveform s performance characteristics while maintaining the constant amplitude and high SE properties necessary for transmission on practical piezoelectric devices.

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Multipath Tracking in a Shallow Water Environment with ML-PMHT

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The Maximum Likelihood Probabilistic Multi-Hypothesis Tracker (ML-PMHT) is a powerful deterministic estimation algorithm that has been successfully implemented in recent work as a multistatic active tracker. Its target-measurement generation model is different from most other tracking algorithms in that it allows any number of measurements to originate from a target in a single scan. This target-measurement generation model makes ML-PMHT a natural fit for use in multipath environments where it is physically realistic and possible that more than one target measurement exists in a scan.

In this work, ML-PMHT is implemented as a shallow water passive tracker. Measurements are generated from a target/clutter model that includes first order (bottom and surface) boundary reflections. Acoustic receivers are distributed along the water surface and represent a distributed buoy field with the capability of transmitting passive acoustic data for land based processing. If the multipath structure of the channel is known precisely, ML-PMHT should function very well. In reality, exact details of the multipath structure of the acoustic channel may be uncertain. Tracker performance is assessed in the face of this multipath measurement uncertainty in a variety of scenarios including surface and submerged targets, uncertain bathymetry, and receiver sensor capability.

The work being presented is a first step in modifying ML-PMHT to track successfully in a shallow water environment that features multipath propagation. As such, performance is measured via Monte Carlo simulation with measurements generated from a passive acoustic simulator that assumes 'straight-line' isovelocity multipath propagation paths. Subsequent work will leverage ongoing testing within NUWC's instrumented ranges department which feature raytracing and propagation loss models to calculate multipath receptions. The long term goal is to develop a robust algorithm capable of tracking targets in shallow water using multiple arrival paths concurrently in single measurement scans.

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Inference Regarding the State of Mobile Underwater Object from a Small Aperture Vertical Array

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A narrowband source ensonifies an area of interest in a refractive undersea environment. A small aperture vertical array is employed to infer the depth, range and speed of the acoustic scattering target by resolving the scattered direct path and one surface interacting path. Tracking the target by means of a continuous wave transmission is challenging due to the difficulty of inferring the frequencies and angles of the two returned closely spaced wave vectors. The propagation of sound in a refractive medium presents additional challenges for inversion of the wave vectors to range, depth and speed. A Gibbs sampler is employed to construct the posterior joint density of all parameters taking full advantage of the analytic tractability of the conditional densities of the received amplitudes and phases and of the ambient acoustic noise power. The conditional densities of the ordered wave vectors however are constructed numerically by 2-dimensional inverse quantile sampling. The inferred joint posterior density of the target state is obtained by constructing a numerical inverse transformation of the acoustic propagation model. Simulation results demonstrate the approach at received signal to noise ratios below -3dB and display the limits of depth and speed estimation as a function system parameters.

Passive Acoustic Navigation and Command of a Low-Cost Autonomous Underwater Vehicle

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Improvements in the reliability of Autonomous Underwater Vehicles (AUVs) over the past few decades has resulted in their widespread use for various oceanographic, defense and industry applications. A long-held goal of AUV research has been the coordinated use of multiple vehicles to enable novel applications, such as effective sampling of spatiotemporal ocean processes, or coherent acoustic processing over many AUVs. Unfortunately, this improved reliability is associated with a high cost that has prevented the realization of this goal: sophisticated and expensive navigational sensors including a Doppler velocity log (DVL) and a high-grade inertial navigation sensor (INS) are necessary for accurate vehicle navigation, driving up vehicle price, size, and power use, and reducing operator willingness to experiment with multi-vehicle behaviors. However, the recent emergence of a new class of very low-cost, miniature AUVs has once again opened up the possibility of multi-AUV deployments. Unfortunately, the absence of a DVL-aided INS on these vehicles results in rapid accumulation of navigation error.

To address this problem, this work presents an inexpensive acoustic localization system to enable accurate and robust navigation for these low-cost, miniature AUVs. A single acoustic transmitter periodically transmits a LFM chirp that is synchronously recorded by a passive nose- mounted ultra-short baseline (USBL) array on the AUV. Phased-array beamforming and matched filtering, combined with AUV pitch-roll-heading from a MEMS inertial measurement unit (IMU), provides an instantaneous estimate of range, azimuth and inclination between the vehicle and the transmitter. By closely coupling beamforming and sequential Monte-Carlo filtering, a consistent estimate of the transmitter position relative to the AUV is generated on- board the vehicle in real-time. This enables absolute localization of the AUV when the transmitter is fixed at a known position, as well as relative navigation behaviors such as homing when the transmitter is moving. The passive nature of the acoustic receiver reduces power use, and allows multiple vehicles to navigate using a single transmitter. An extension to this system enables rudimentary command of the AUV through the transmission of various LFM chirps, with each chirp commanding the vehicle to perform a specific pre-programmed behavior. Results from field experiments are presented, demonstrating the described acoustic localization system and behavioral mode-switching. [Work supported by ONR, DARPA and Battelle]

[Work supported by ONR, DARPA and Battelle]

Decoupling Depth and Range Estimation using Acoustic Invariant Theory

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Multipath vertical angles of arrival (AOA) and time differences of arrival (TDOA) can be measured on vertical or horizontal apertures from broadband signatures, either using a matched filter when the source waveform is known, or by cross-correlating beams when processing an uncooperative target. Such processing produces reasonably compact arrivals on a 2D plane with TDOA on the x-axis and sin(AOA) on the y-axis. A remarkable consequence of acoustic invariant theory is that such arrivals are confined to a 1D curve. The shape of this curve is a function of two parameters: beta, the scalar parameter characterizing acoustic invariance for a given waveguide; and range. This means that the arrivals for a source at a particular range will be restricted to lie on this single curve, even if the source is moved up and down in depth. The parameter beta is roughly 1 in shallow water, and roughly -3 in a canonical deep water environment, but it can be more precisely calculated using wave propagation models, measured directly using a source moving along a known track, or even jointly estimated with range if some assumptions can be made about the source motion. Once the range has been recovered, the depth can be estimated by matching against patterns of arrivals along the range-specific curve. These patterns can be pre-calculated using wave propagation models, or simply measured using cooperative own sources, such as acoustic modems on vehicles.

We will review how the relationship between TDOAs and AOAs described above arises as a natural consequence of acoustic invariant theory. Next, we will present the processing for estimating TDOAs and AOAs, fitting them to the closest curve from the set of range-specific curves to estimate the range, and, finally, matching them to expected patterns associated with specific depths. We will show results from the RADAR 2007 field experiment in Portugal.

Model-Based Signal Processing for Sediment Property Inversion using Group Velocities

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Dispersion curve tracking is successfully carried out using particle filtering applied to a Short Time Fourier Transform of an acoustic signal that has propagated a long distance in an underwater waveguide. The filter estimates probability density functions of modal frequencies at specific time 'slices' of the computed spectrogram. The observation equation, a building block of the sequential filter, is obtained using the stationary phase approximation applied to the frequency-domain pressure field of distinct modes. The transition (or state) equation describes the evolution of modal frequencies with time. The filter provides 'clouds' of modal frequency particles that represent probability densities. Once these are available, they are propagated backwards through an acoustic model to estimate water column depth and sediment sound speed and thickness. This is achieved by calculating replica arrival times using group velocities computed with normal modes, which are compared to the arrival times corresponding to the clouds of frequency particles forming the probability density functions. We obtain clouds of values for the unknown parameters that form probability densities for those. In parallel, we calculate densities for arrival times rather than frequencies exploiting the results of the filter. We proceed in a similar fashion to estimate the propagation medium properties (water column depth and sediment sound speed and thickness) and we compare the two inversion techniques (using frequency and time densities). The approaches are tested with synthetic data as well as data collected in the Gulf of Mexico, where the environment consists of a shallow water waveguide with a thin sandy sediment over limestone. Interestingly, the parameter that is most accurately estimated is the water column depth for both synthetic and real data. However, sediment sound speed and thickness are resolved even when the sediment is very thin. The most interesting and revealing information is provided by the joint density of these two parameters.

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Adaptive Doppler-Azimuth Processing for Comb Signals using Steered Covariance Matrices

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This paper presents an adaptive Doppler-azimuth (ADA) processing technique that exploits the spectral degrees of freedom offered by frequency comb signals for active sonar. It is shown adaptive ADA processing using steered covariance matrices (ADA-STCM) results in improved spatial resolution and increased suppression of nearfield direct blast and reverberation over conventional methods. ADA-STCM requires fewer temporal snapshots than standard adaptive techniques because STCMs enable combining narrowband snapshots across time and comb tone frequencies. Performance was verified using simulations as well as real data from the Littoral Continuous Active Sonar 2015 (LCAS15) experiment which took place in the Mediterranean off the coast of La Spezia, Italy. ADA-STCM processing of both the simulated data matching the LCAS-15 experiment and also the real LCAS-15 data with an injected target was performed. Processed results confirm that ADA- STCM improves spatial resolution and suppresses nearfield direct blast and reverberation, while requiring fewer temporal snapshots than standard adaptive techniques.

[We acknowledge the NATO CMRE and LCAS Multinational Joint Research Project participants for their resources and support in the LCAS-15 sea trial. This work was supported by the U.S. Office of Naval Research (ONR 321US) under grant no. N0014-13-1020-1.]

Adaptive Range Processing for Active Sonar

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Signal detection in active sonar is conventionally performed with the matched filter. The matched filter is simple, efficient, and well understood. Unfortunately, it is also susceptible to poor sidelobe performance in the presence of loud signals. This can lead to signal masking and missed detections. The matched filter may not be able to resolve the arrival time of multiple, closely-spaced signals. This information can be useful for analyzing the acoustic environment and understanding the underlying scattering function.

In this work, the application of an adaptive pulse compression (APC) technique from the radar community will be presented for active sonar. In particular, the reiterative minimum mean squared error (RMMSE) algorithm will be shown. This algorithm relies on a structured covariance estimate to enable single pulse adaptation. Some sensitivities associated with the assumptions implicit in the structured estimate will be presented, along with mitigation strategies.

Characterization of Communication Channel Conditions over Varying Ranges as Long as 10 km

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In this work, a set of experiments are run over a period of 4 days to demonstrate long range communication between a 2-kHz transmitter and a remote receiver equipped with a 5-element vertical line array. The communication link is deployed at approximately 10 km from the southern coast of Nova Scotia, near St-Margaret's Bay. In this area, local bathymetry indicates a depth between 70 and 90 meters. The receiver is fixed at approximately 40 meters from the surface, and the transmitter is deployed from an anchored vessel at distances between 1 km and 10 km. The acoustic channel impulse response is characterized as a function of time over transmission periods of ten minutes. The transmission loss, Doppler spread and multipath delay spread is measured. Different sounding signatures, including short pulses, chirps, and spread spectrum signatures are utilized for hardware calibration and to obtain complimentary information about the acoustic channel. Also, during the trial, sensors are utilized to monitor the environment: sound speed profiles are obtained using a handheld CTD sensor, an ADCP is mounted on the hull of the vessel to obtain a transect of the currents, and an ultra-sonic ranger serves to measure the sea state. These sensors are deployed and their outputs are fed in a software model that predict acoustic propagation. It is found that the transmission loss, Doppler spread and multipath statistics are well predicted by the models.

Relative Impacts of Signal-to-Noise Ratio and Propagation-Induced Distortion on Passive Acoustic Monitoring of Cetaceans

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Automated passive acoustic monitoring (PAM) is widely used to study cetaceans in their natural habitats. The utility of automated PAM systems depends on their ability to operate across a wide range of ocean acoustic environments. Additionally, the performance of a PAM system must be carefully characterized in order to be able to transform single-vocalization detections into estimates of the number of animals present. A critical step in obtaining an accurate density estimate is to determine the detection function. Detection functions are used to map the signal-to-noise ratio (SNR) of a detected vocalization to the probability of detection and correct classification. In doing so, there is an implicit assumption that environment-dependent propagation has negligible impact on a PAM system's performance. In this presentation, a counter example will be provided to demonstrate how this assumption biases performance estimates in terms of a classifier's accuracy and area under the ROC curve.

Variability in acoustic propagation characteristics leads to differences in how each cetacean vocalization is distorted as it propagates along the source-receiver path. Unless these differences are accounted for, the acoustic environment will bias estimates of a PAM system's performance. The impact of environment-dependent propagation on an aural classifier is demonstrated using a pulse propagation model. Simulations were conducted to disentangle the relative impacts of SNR and signal distortion caused by multipath addition — the effects of SNR and distortion were considered separately, as well as in combination. Shannon information theory predicts a linear relationship between SNR and classifier performance; deviations from this linear trend may be attributed to propagation-induced signal distortion. Liner regression was performed to assess the relationship between performance and SNR, and the value of R^2 was used to indicate the relative proportion of performance change explained by SNR. It was found that in many environments, propagation could have a significant impact on performance. In such environments, the standard assumption that a PAM system's performance scales with SNR significantly biases performance estimates. Simulation results will be presented for sample bowhead and humpback whale vocalizations propagated through a model of an Arctic environment, to demonstrate the potential risks of overlooking propagation-induced distortion when characterizing system performance.

A Goal Reasoning Based Architecture for Intelligent Active Sonar

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Active sonar systems include a variety of parameters, both in transmission and in receive signal processing, that can be tuned to improve search and track performance in various scenarios. Historically, parameters have remained largely fixed in active systems, in part because adding parameter tuning to sonar operator's task list imposes an impractical cognitive load. We propose adopting a goal reasoning architecture to develop intelligent active sonar systems that perform dynamic parameter adaptation. Originating in the artificial intelligence community, goal reasoning architectures support systems that not only adapt to meet a fixed set of goals but also reason about those goals; goal reasoning systems are designed to modify and/or augment goal sets based on information gained in real time.

To build a cognitive sonar model, we adopt the goal-driven autonomy (GDA) framework proposed in [Klenk et al., Computational Intelligence, May 2013]. Interacting with a planner that generates sequences of actions and a set of sensors that observe the environment, the GDA controller identifies discrepancies between predicted and actual observations, generates possible reasons for discrepancies, and adjusts goals and goal priorities based on newly acquired information. Based on the GDA framework, we have developed a model for an intelligent active sonar system that tunes parameters at both the transmitter and receiver to improve its ability to track a maneuvering target in clutter. In its current implementation, the system tunes the transmitted waveform and the detection threshold. The waveform is adapted based on observed Doppler from the target, and the threshold is adapted based on a rough estimate of local clutter density. In order to operate in a cognitive fashion, the system must be able to both predict performance as a function of its actions and to evaluate performance based on its observations. In this presentation, we will describe the development of metrics for performance prediction and evaluation. We will also describe the techniques used to tune the transmit waveform and detection threshold and present simulations that illustrate the performance gain achieved by the cognitive system.

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M-ary orthogonal spread spectrum signaling for very low signal to noise ratio acoustic communications

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Acoustic communications through a doubly spread shallow water ocean environment is challenging as it typically requires the joint estimation of the acoustic response and the symbols that are sent. However if the symbols are appropriately orthogonal and judiciously constructed joint estimation of the acoustic response can be accomplished in a computationally efficient manner without pilot symbols. We consider an M-ary orthogonal spread spectrum signal set for the purpose of coherent or non-coherent symbol decisions in multi-path environments at very low signal to noise ratios (SNR). The M symbols have a more general feature of orthogonality over the entire multi-path delay spread of the channel operator and this generalized orthogonality feature allows for the minimum mean square error (MMSE) estimation of the acoustic response function with greatly reduced computational effort at the receiver. These signals with an assumed sparsity prior are employed for joint symbol and broadband channel estimation scheme without the use of intra-packet training symbols of any kind. The approach allows for the compensation for the shared time varying Doppler process of the various coherent arrivals. Demonstrations with at-sea recordings at extremely low received SNR are presented.

Coherent OFDM Receiver for High Data Rate Acoustic Communication Systems

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High data rate coherent transmission over acoustic channels is a challenging problem due to the combined effects of long multipath and Doppler fluctuations. To account for these effects, we design a coherent receiver based on multicarrier modulation in the form of orthogonal frequency division multiplexing (OFDM). OFDM is an attractive method for data transmission over frequency-selective channels due to its ability to achieve high bit rates at reasonably low computational loads. This fact motivates the use of OFDM in mobile acoustic communications where the channel exhibits long multipath delays but each narrowband carrier only experiences flat fading, thus eliminating the need for time-domain equalizers.

The major problem in applying OFDM to acoustic channels is the Doppler distortion caused by the relative motion between the transmitter and receiver, which causes non-uniform frequency shifting across the acoustic signal bandwidth. In highly mobile scenarios, Doppler frequency scaling is effectively seen as a time-varying channel distortion which adversely affects the performance of OFDM systems as it causes loss of orthogonality between the carriers. To mitigate the resulting distortion, resampling must be performed at the receiver front-end. Coarse resampling is typically performed on an entire frame of OFDM blocks, and may leave individual blocks within a frame exposed to different frequency offsets. These offsets, if left uncompensated, can have a detrimental impact on data detection. To compensate for these offsets, we use a hypothesis testing approach proposed in [A. Tadayon and M. Stojanovic, *Proc. IEEE Oceans'17 Conf.*, 2017]. The approach is based on differentially coherent detection which keeps the complexity low and requires only a negligible pilot overhead.

Reliable coherent data detection requires the knowledge of channel state information (CSI) at the receiver. Conventional approaches to channel estimation, such as least-square (LS) estimation, are based on sample-spaced channel models. Consequently, their performance suffers when the channel is not sample-spaced, as is the case in most practical situations. Capitalizing on the physics of multipath propagation which reflects the sparse nature of channel, we propose a path-based sparse channel estimator. The resulting path identification (PI) algorithm targets a continuum of path delays, eliminating the dependence on a sample-spaced model and focusing instead on processing a transformed version of the signal observed over all the carriers spanning the system bandwidth [M. Stojanovic and S. Tadayon, *Proc. 52nd Annual Allerton Conf. Commun. Control Comput.*, 2014]. Unlike the sparse identification methods with over-complete dictionaries such as orthogonal matching pursuit (OMP) [W. Li and J. C. Preisig, *IEEE J. Ocean. Eng.*, 2007], the resolution and delay coverage that the PI algorithm provides can be increased without a prohibitive cost in complexity.

Block-by-block receivers obtain the necessary CSI using the standard pilot-assisted channel estimation method which reduces bandwidth efficiency of the system as it requires assigning pilot carriers in each block. Assuming that the channel is slowly time-varying, we propose a receiver algorithm that makes channel predictions from one block to the next, thus reducing the pilot overhead and increasing the data throughput. The algorithm is demonstrated on experimental data from the Mobile Acoustic Communication Experiment (MACE 2010) where the transmitter moves at a relative speed on the order of 0.5-1.5 m/s with respect to the receiver and the OFDM blocks containing up to 2048 QPSK modulated carriers occupy the acoustic frequency range between 10.5 and 15.5 kHz. Very good performance is obtained, which we discuss in terms of data detection mean-squared error (MSE) and symbol error rate (SER).

Analysis of Linear Co-prime Array Geometries in a Continental-Shelf Ocean Environment

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Co-prime array geometries have received a lot of attention due to their ability to discriminate $O(MN)$ sources with only $O(M+N)$ sensors. This has been demonstrated both theoretically and in simulation. However, there are many practical limitations that make it difficult to realize the enhanced degrees of freedom when applying co-prime geometries to real acoustic data taken on a horizontal line array. For instance, co-prime sampling leads to increased sidelobe levels compared to that of a uniform linear array (ULA). These increased sidelobes can obscure and mask lower signal-to-noise-ratio acoustic sources, making them difficult to detect. In this work, we present a synthetic aperture method for filling in holes and increasing redundancy in the difference co-array by exploiting array motion. The synthetic aperture method is applied to acoustic data collected off the Southeastern shore of Florida on a fixed large aperture horizontal ULA. The array motion is simulated by taking a co-prime sampled sub-array and virtually moving it along the horizontal aperture of the fixed array. We demonstrate that the synthetic aperture processing on real acoustic data results in reduced sidelobe levels compared to that of the physical co-prime aperture. This work was supported by ONR.

[This work was supported by ONR.]

Spatial Correlation Resampling for Wideband Source Enumeration and Direction-of-Arrival Estimation on Sparse Arrays

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Sparse arrays, such as minimum redundancy, coprime, and nested arrays, often exploit the second-order spatial statistics of the propagating field to localize more sources than sensors by constructing an augmented covariance matrix (ACM) from spatial correlation estimates [Pillai & Haber 1987, Vaidyanathan & Pal, 2011]. The source localization performance largely depends on the number of snapshots available, which are often limited in underwater environments due to the slow propagation speed, large array aperture and non-stationary field. However, many natural and manmade sound sources span significant bandwidth. Combining these wideband snapshot data coherently across multiple frequency bands provides processing gains for subspace-based source localization [Wang & Kaveh, 1985]. Nearly all previous wideband processing algorithms focused on uniform linear arrays (ULAs) and very few consider processing wideband signals on sparse arrays.

Our algorithm extends Krolik & Swingler's [1990] spatial resampling technique from ULAs to sparse array geometries. The proposed algorithm applies to any sparse array geometry with a contiguous region in the difference coarrays and requires no preliminary estimates on the array manifold for wideband focusing. The proposed algorithm estimates the spatial correlation function for each frequency band. Each narrowband spatial correlation estimate is spatially resampled to account for the impact of changing temporal frequency on spatial wavenumber, aligning the correlation estimates at a nominal frequency. The aligned resampled spatial correlation functions are averaged across all frequency bands to estimate the 'focused' correlation function, which then populates the diagonals of a Hermitian Toeplitz ACM. A modified minimum description length (MDL) criteria termed MDL-gap and another information criteria termed Second-ORder sTatistic of Eigenvalues (SORTE) estimate the number of sources using the eigenvalues of the ACM. Narrowband spectral MUSIC estimates the source DOAs using the eigenvectors of the ACM. Numerical simulations show that the proposed spatial correlation resampling algorithm achieves improved performance over the incoherent subspace processing approach [Han & Nehorai, 2013] for estimating both the number of wideband sources and their corresponding DOAs in under-determined scenarios with more sources than sensors. The proposed focusing algorithm approaches the performance of narrowband algorithms with increased snapshots to provide comparable time-bandwidth products, especially in low SNR and snapshot poor scenarios.

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Performance Prediction of Coprime Sampled Arrays in Spatially Correlated Noise

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Prior analysis on coprime sampled arrays (CSAs) made the assumption of spatially uncorrelated noise. This work predicts the performances of CSAs in the presence of spatially correlated noise. The noise is modeled as a first-order auto-regressive process to enable introducing inter-sensor noise correlation that decays exponentially with length. The analysis on the CSA geometry considers both the subarray product processor (CSA_{pp}) [Vaidyanathan & Pal, 2011] and the conventional beamformer (CBF) on the CSA (CSA_{cbf}). Coarray processing allows for comparing the two CSA processors to the baseline of the densely populated CBF uniform line array (ULA). The beamformers are compared in terms of detection and estimation performance.

In terms of detection, the array gains of all processors are derived through the deflection statistic [Cox, 1973]. The deflection is generalized for processors that are either incoherent or involve non-linear processing of the array measurements. This statistic is a metric to evaluating the noise variance reduction achieved at the beamformer output by quantifying the separation of the binary hypothesis probability distribution functions (PDFs). These PDFs are used to evaluate the receiver operating characteristics of each processor to analyze and compare the beamformer detection sensitivities.

In terms of estimation, the implicit Fourier relations with coarray processing and power spectral density (PSD) estimation shows the estimate produced at the beamformer output is biased. The processing of the measurements inherently smears the "true" spatial characteristics of a random acoustic field. For the same measurements, the CSA_{pp} and CSA_{cbf} will smear the true PSD differently. This work considers both aspects of performance metrics to analyze and compare the two CSA processors against the baseline of the ULA in spatially correlated noise.

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*Now at the Naval Undersea Warfare Center Division Newport

Attendees

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2	Bacon, Joshua	BBN Technologies
3	Baggeroer, Arthur	MIT
4	Barros, Abner	University of Massachusetts Dartmouth
5	Bautista, Radienxe	Naval Undersea Warfare Center Divison Newport
6	Binder, Carolyn	Defence Research and Development Canada
7	Bissinger, Brett	Applied Research Laboratory, Penn State University
8	Blair, Ballard	MITRE
9	Bockman, Alexander	MIT Lincoln Laboratory
10	Bongiovanni, Kevin	Raytheon Integrated Defense Systems
11	Bousquet, Jean-Francois	Dalhousie University
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15	Cox, Harry	Lockheed Martin
16	Cuprak, Travis	L3 Adaptive Methods, Inc.
17	Daily, Matthew	BBN Technologies
18	Davidson, Keith	Office of Naval Research
19	DeMaio, Martin	Alion Science
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34	Nelson, Jill	George Mason University
35	Newhall, Bruce	
36	Northardt, Tom	MIKEL, Inc.
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40	Richmond, Christ	Arizona State University
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43	Ryan, Colin	MIT Lincoln Laboratory
44	Rypkema, Nick	MIT
45	Sarma, Ashwin	BAE Systems/University of Rhode Island
46	Scharf, Louis	Colorado State University
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48	Soli, Jonathan	Duke University
49	Tadayon, Amir	Northeastern University
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52	Vaccaro, Rick	University of Rhode Island
53	Wage, Kathleen	George Mason University
54	Walstead, Sean	Naval Undersea Warfare Center Division Newport
55	Wilmott, Dan	Department of National Defence

Wednesday October 4, 2017		**Thursday October 5, 2017**		**Friday October 6, 2017**	
		8:15–9:30	Session C Spec. Sess. II *Laurel*	8:40–9:30	Session I Acous. Inf. Proc. *Laurel*
		9:30–10:20	Session D Non-Coherent Detect. *Laurel*	9:30–10:20	Session J Adaptive Proc. *Laurel*
		10:20–10:45	Break *Laurel*	10:20–10:45	Break *Laurel*
		10:45–12:00	Session E Adapt. Beamforming *Laurel*	10:45–12:00	Session K Spec. Sess. IV *Laurel*
		12:00–1:00	Lunch *Whisp. Pines*	12:00–1:00	Lunch *Whisp. Pines*
		1:00–2:15	Session F Spec. Sess. III *Laurel*	1:00–1:50	Session L Acoustic Comms. *Laurel*
		2:15–3:05	Session G Waveform Design *Laurel*	1:50–2:05	Break *Laurel*
		3:05–3:30	Break *Laurel*	2:05–3:20	Session M Sparse Arrays *Laurel*
		3:30–4:45	Session H Tracking & Positioning *Laurel*		
5:00–6:00	Welcome Reception *Whisp. Pines*	5:00–6:00	Reception *Whisp. Pines*		
6:00–8:00	Dinner *Whisp. Pines*	6:00–8:00	Dinner *Whisp. Pines*		
8:00–9:30	Session A Plenary Session B Spec. Sess. I *Laurel*	8:00–?	SOB Session *Laurel*		