A Book of Abstracts for the

2019 Underwater Acoustic Signal Processing Workshop

October 16-18, 2019

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Welcome to the 2019 IEEE workshop on Underwater Acoustic Signal Processing. This year the special session, organized by Zoi-Heleni Michalopoulou and Paul Gendron, is titled *Ocean Acoustic Source Localization: Signal Processing Algorithms and Applications*.

The organizing committee would like to thank and acknowledge the continued support of the Office of Naval Research, Raytheon Integrated Defense Systems, and the Acoustical Society of America. Finally, we are proud to announce that this year's recipient of the Donald W. Tufts UASP Award is Prof. Leon Cohen.

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The 2019 Donald W. Tufts UASP Award is presented to Prof. Leon Cohen

for notable contributions in time-frequency distribution theory and forging deep connections between underwater scattering and the statistics of sonar signals

Leon Cohen's work epitomizes the statistical part of statistical signal processing, having incorporated distribution theory into his research long before it was common to do so. A deep dive into Leon's original papers in the field of physics suggests why he may have been ideally suited to this effort. It reveals the insight to include randomness in historically deterministic subjects such as gravitation (1972) as well as to clarify the subtle use of probability in quantum mechanics (1966). He has applied his mathematical insights to the identification of appropriate non-Gaussian statistical families for specific reverberant and clutter-filled undersea environments at past UASP workshops.

Professor Leon Cohen received the B.S. in physics from the City University of New York in 1958, and the M.S. and Ph.D. degrees in theoretical physics from Yale University, New Haven CT in 1964 and 1966 respectively. His pioneering work in 1966 on generalized phase-space distributions in quantum mechanics has had a significant impact on the field of signal processing with applications in sonar signal processing, speech processing, biomedical signal processing, and machine monitoring via vibration signal analysis. Eventually emerging from this work was the Cohen class of time-frequency distributions (TFDs) covariant with respect to time and frequency shifts. The Cohen class defines an infinite set of functions that may be used for signal analysis in the joint time-frequency plane and includes well-known TFDs such as the Wigner distribution and the spectrogram. Lofargrams, essentially spectrograms of array beam time-series, remain a critical tool in sonar signal processing. Many regard Leon as the "godfather" of time-frequency analysis for nonstationary signal processing with citations in the thousands.

Although it is easily overshadowed by his technical achievements, one of Leon's most valuable skills is his ability to imbue scientific curiosity to the next generation of researchers with his joie de vivre and his engaging presentation style. He is, and has long been, a wonderful mentor to young colleagues in the field.

For his many contributions, the underwater acoustic signal processing community is pleased to present the 2019 Donald W. Tufts UASP Award to Professor Leon Cohen.

Contributed by Keith Davidson, Patrick Loughlin, Doug Abraham, Geoff Edelson and Ashwin Sarma

Schedule at a glance

Wednesday October 16, 2019		**Thursday October 17, 2019**		**Friday October 18, 2019**	
		8:30–9:55	Session B Loc. I *Laurel*	8:30–10:15	Session F Waveforms & Proc. *Laurel*
		9:55–10:25	Break *Laurel*	10:15–10:40	Break *Laurel*
		10:25–11:50	Session C Loc. II & Mach. Lrn *Laurel*	10:40–12:00	Session G Loc. IV *Laurel*
		12:00–1:00	Lunch *Whisp. Pines*	12:00–1:00	Lunch *Whisp. Pines*
		1:00–2:25	Session D Loc. III *Laurel*	1:00–2:30	Session H Sys and Proc *Laurel*
		2:25–2:55	Break *Laurel*		
		2:55–5:00	Session E Array Proc. & conn. *Laurel*		
5:00–6:00	Welcome Reception *Whisp. Pines*	5:00–6:00	Hors d'ouvres de Capon *Whisp. Pines*		
6:00–8:00	Dinner *Whisp. Pines*	6:00–8:00	Dinner *Whisp. Pines*		
8:00–9:00	Session A Plenary *Laurel*	8:00–?	SOB Session *Laurel*		

Sessions: Titles and presenters

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

Plenary presentation by the 2019 Donald W. Tufts UASP Award recipient

A–1 *The History of Noise*
Leon Cohen, City University of New York

Session B: Thursday Morning, 8:30am–9:55am

Localization I

B–1 *Using Blind Deconvolution for Sound Source Localization*
Shima Abadi, University of Washington

B–2 *Source localization: matched field processing, waveguide/array invariant and blind deconvolution*
Hee-Chun Song, Scripps Institution of Oceanography

B–3 *Passive Acoustic Localization with Adaptive Back-Propagation*
Ying-Tsong Lin, Woods Hole Oceanographic Institution

B–4 *Simple and accurate beacon-based acoustic ranging in complex ocean environments*
Ashwin Sarma, BAE Systems/URI

Session C: Thursday Morning, 10:25am–11:50pm

Localization II and Machine learning

C–1 *Convolutional neural networks applied to source and environmental characterization with mid-frequency data on two vertical line arrays*
David Knobles, Knobles Scientific and Analysis, LLC

C–2 *Source-range estimation in a modeled Arctic environment with a convolutional neural network*
Rui Chen, MIT

C–3 *Direction-of-arrival estimation using supervised machine learning*
Emma Ozanich, Scripps Institution of Oceanography

C–4 *Sediment classification of rough multilayer seafloors using localized finite element modeling and machine learning*
Christina Frederick, NJIT

Session D: Thursday Afternoon, 1:00pm–2:25pm

Localization III

D–1 *Passive source depth discrimination in deep-water using a horizontal line array*
Julien Bonnel, ENSTA Bretagne

D–2 *A computational Bayesian processor for active sonar target localization from sub-Rayleigh resolvable multipath arrivals in refractive environments*
Abner Barros, University of Massachusetts Dartmouth

D–3 *Water column sound speed and sediment property inversion via Bayesian filtering and linearization with SBCEX linear frequency modulated signals*
Zoi-Heleni Michalopoulou, Department of Mathematical Sciences, New Jersey Institute of Technology

D–4 *Long Range Source Localization in the Philippine Sea using the Frequency-Difference Autoproduct with Cross-Term Corrections*
David Geroski, Applied Physics Program, University of Michigan

Session E: Thursday Afternoon, 2:55pm–5:00pm

Array processing and connections

E–1 *Designing universal adaptive beamformers for moving interferers*
John Buck, UMass Dartmouth

E–2 *Performance Weighted Blended Power Spectrum Estimation*
Jeff Tucker, George Mason University

E–3 *Estimating frequency-wavenumber spectra using a universal dominant mode rejection processor*
Kathleen Wage, George Mason University

E–4 *Improving the Robustness of the Dominant Mode Rejection Beamformer with Median Filtering*
David Campos Anchieta, University of Massachusetts Dartmouth

E–5 *DOA Estimation in the Presence of Local Scattering and Correlated Noise Using Co-prime Arrays*
Tongdi Zhou, Temple University

E–6 *Clock drift estimation and correction across multiple asynchronous sonobuoy arrays*
Anil Ganti, Duke University

Session F: Friday Morning, 8:30am–10:15am

Waveforms and processing concepts

F–1 *Measurements of matched-filter loss from sea-surface reflection*
Stefan Murphy, Defence Research and Development Canada

F–2 *Comb Waveform Design using Multi-Tone Sinusoidal Frequency Modulation*
David Hague, Naval Undersea Warfare Center

F–3 *Optimal Target Detection for Cognitive MIMO Radar/Sonar*
Christ Richmond, Arizona State University

F–4 *Phase space representation of acoustic signals*
J. Daniel Park, Applied Research Laboratory Penn State University

F–5 *Sonar Signal Representation Using Sparse Gabor Dictionaries*
Ananya Sen Gupta, University of Iowa

Session G: Friday Morning, 10:40am–12:00pm

Localization IV

G–1 *Shallow Water Localization Estimates Using a Matched-Field Backpropagation Algorithm*
Carolyn Binder, Defence Research and Development Canada

G–2 *Acoustic source localization: A non-Euclidean geometric approach*
Steven Finette, Naval Research Laboratory

G–3 *New Method for Source Depth Discrimination Using Information Geometry*
Daniel Brooker, Naval Research Laboratory

G–4 *An Algebraic Geometry Approach to Passive Source Localization*
Margaret Cheney, Colorado State University

Session H: Friday Afternoon, 1:00pm–2:30pm

System and processing concepts

H–1 *Development of a Distributed MIMO Bathymetric Mapping System*
Joseph Edwards, MIT Lincoln Laboratory

H–2 *Doppler-based single-hydrophone relative navigation for low-cost underwater vehicle swarming*
Erin Fischell, Woods Hole Oceanographic Institution

H–3 *Analyzing models for goal management with operator input in intelligent active sonar*
Jill Nelson, George Mason University

H–4 *Adaptive and Compressive Sensing Beamformers Compared*
Paul Hursky, Sonar-synesthetics LLC

Abstract Listings

The History of Noise

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"Noise" had a glorious birth. While there were rumblings before 1905 it was Einstein's explanation of Brownian motion that started the field. His motivation was not the mere explanation of the erratic movement of pollen but much bigger aims: that noise could establish the existence of atoms. Immediately after Einstein, there was an incredible flurry of ideas of the most profound kind that continues to this day. Within three years Langevin started the field of stochastic differential equations, although that was not his motivation. Einstein continued to make contributions to noise till about 2015. Concurrently there were major developments by great physicists, engineers, mathematicians, astronomers, who laid both the foundations and applications of noise to many fields. Indeed, many fields have claimed to be the originators of noise. The historical twists are fascinating.

But noise, considered by many as unwanted, and mistakenly defined as such by some, has little respectability. The term itself conjures up images of rejection. Everyone builds filters to stop it and few have love affairs with it. Yet, it is an idea that has served mankind in the most profound ways.

The story of noise is a fascinating one and while in its early stages its story was clearly seen, its subsequent divergence into many sub-fields has often resulted in a lack of understanding of its historical development. We will discuss who did what, when, and why, the historical misconceptions, and the reasons for the impact on so many fields. But most importantly, to show that the history of noise is worth telling.

Using Blind Deconvolution for Sound Source Localization

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Sound from a remote underwater source is commonly distorted by multipath propagation. Blind deconvolution is the task of estimating the unknown waveforms for the original source signal and the source-to-receiver impulse response(s), from signal(s) recorded in an unknown acoustic environment. Synthetic time reversal (STR) is a blind deconvolution technique that relies on generic features of the propagating modes or ray paths that lead to multipath sound propagation. STR can be used to estimate simultaneously the source signal and impulse response waveforms from a remote unknown source. However, it does not recover absolute timing or amplitude information from the recorded signal, so elementary source ranging estimates based on propagation timing or amplitude decay are not possible. Fortunately, relative amplitude and timing information can be recovered. This presentation describes how mode-based and ray-based STR can be extended for sound source localization using the relative amplitudes, relative timing, and cross correlations between different acoustic ray or mode arrivals at the array. Results are provided from two experiments. In the first experiment, ray-based STR is used to locate the source of a 50-millisecond chirp signal with a bandwidth of 1.5 to 4.0 kHz recorded by a 32- element receiver array in 60-m-deep water. In the second experiment, mode-based STR is used to estimate the range of bowhead whale calls with a bandwidth of 50 to 500 Hz recorded with a 12- element vertical array in a 55-m-deep dispersive waveguide. The localization results are compared with conventional techniques such as the matched-field processing (MFP) and the conventional mode filtering (CMF) in these two propagation environments, respectively.

[Work supported by the Office of Naval Research]

Source localization: matched field processing, waveguide/array invariant and blind deconvolution

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Over the last four decades, a de facto approach to source localization in underwater acoustics has been matched-field processing (MFP). MFP compares the measured field at an array with replicas of the expected field for all possible source locations, requiring accurate knowledge of the environment and/or array geometry (e.g., array tilt). As a result, MFP is inherently sensitive to modeling mismatch and has been successful mostly at low frequencies below 1 kHz. In this talk, we present a robust, physics-based approach that requires minimal/no environmental information, called the array invariant (AI). The AI based on the beam-time migration is built upon the elegant 'waveguide invariant' theory that characterizes the dispersion characteristics of ocean waveguides. In addition, the AI has the capability of self-calibrating the vertical array tilt. While AI requires either a broadband impulsive source (e.g., an explosive) or Green's function, we can extract the Green's function from unknown sources (e.g., ships) using a ray-based blind deconvolution. The cascade of blind deconvolution and AI has been successfully demonstrated to localize and track a surface ship using both a vertical or horizontal array in shallow water. Further, the approach has been extended to localize a weak submerged target in the presence of surface ship.

Passive Acoustic Localization with Adaptive Back-Propagation

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This paper is presenting an adaptive back-propagation method for passive acoustic localization by tracing signals received on multiple hydrophones back to sound sources. This method is generalized from an adaptive normal mode back-propagation method by Lin et al. (J. Acoust. Soc. Am. vol. 131, pp. 1798-1813, 2012), which required a normal mode filter with a vertical hydrophone array. The generalization is to directly back-propagate received signals without passing through a mode filter. The principle of this new Adaptive Back-Propagation method is close to the classic time reversal technique, and the advancement is on the adaptive Gauss-Markov processing to balance the resolution and robustness of localization according to the signal-to-noise ratio. Numerical examples of line array end-fire and distributed arrays will be shown in the presentation. Experimental data collected in complex oceanic environments will be utilized to demonstrate the advantage of this localization approach in the real world.

[The work is supported by the Office of Naval Research, USA.]

Simple and accurate beacon-based acoustic ranging in complex ocean environments

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Driven by the problem of self-localization of an underwater node via acoustic means, we discuss a method to arrive at accurate range estimates from a beacon source and refine these estimates over time. The method is shown to successfully operate when the beacon source is transmitting low probability of detection coded waveforms. Continuous transmission of such waveforms are used for repeated estimation of time-varying ranges. The method accounts for expected relative node-beacon movement via a motion model in a Kalman framework, thereby allowing continuous refinement over time. Deliberately avoiding the less reliable acoustic predictions related to amplitude and phase, the method incorporates travel-time predictions from possible ray trajectories determined via ray-theoretic modeling and in-situ sound speed field estimates to achieve highly accurate individual ranging estimates. Ambiguities that arise from the complexities of limited connecting ray trajectories or uncertainties in sound speed field estimates are handled over time by viewing the results as a sequence that must obey the Kalman motion model. These estimates are therefore further refined in the Kalman framework, leading to a continually improving self-localization estimate with errors eventually converging to much less than single-ping levels. We show the applicability of the method over a range of acoustic frequencies. A statistically significant set of at-sea results are provided showing the performance of the estimator over time. In addition, we suggest methods for combining results from two or more beacons for 3D self-localization.

[The work is supported by DARPA. Distribution Statement "A" (Approved for Public Release, Distribution Unlimited).]

Convolutional neural networks applied to source and environmental characterization with mid-frequency data on two vertical line arrays

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Acoustic time series data were recorded on March 24, 2017 on two 16-element vertical line arrays (VLAs) spaced about 6.7 km apart and deployed on the New England continental shelf in about 75 m of water. The acoustic field was due to the RV Endeavor towing an International Transducer Corporation (ITC) 2015 source moving at about 2.7 knots and emitting continuous wave (CW) sound radiation at 1503, 2003, 2503, 3003, 3503, and 4003 Hz. The source track formed a rectangle, about 17 km long and about 0.6 km wide with the two VLAs on a line passing through the center of the rectangle. A convolutional neural network (CNN) is trained with a broadband normal mode model for parameters that describe both the track and the seabed. The seabed is modeled as an N-layer sediment where the frequency-dependent sound speed and attenuations are modeled by the viscous grain shearing (VGS) model in each layer. The feature time series X are the received level versus time for each channel and frequency. The trained network is applied to about 7 hours of data. Predicted labels include CPA range, ship speed, source depth, and several parameters of the VGS theory that characterize a broad spectrum of seabed types ranging from clays to sands. The predicted labels are compared to the ground truth labels for source track parameters and direct measurements made for the sediment sound speeds.

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Source-range estimation in a modeled Arctic environment with a convolutional neural network

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Conventionally, the range of an underwater source to a receiver array can be estimated using matched field processing (MFP), a method that relies on precise environmental modeling to achieve high accuracy. In this study, we take a convolutional neural network (CNN) approach to source-range estimation and compare its effectiveness to MFP in a modeled Arctic propagation environment. The covariance matrices of simulated signal upon a vertical line array are used as input data into the CNN and output estimates through both classification and regression approaches are examined. The training data consist of a near-surface monopole source placed at discrete increments between 3 - 50 Km away from the receiving array while test data is generated by placing the source at randomly selected ranges within the training interval. The CNN's architecture is designed to be light-weight. Its parameters are optimized through cross-validation, and regularization is implemented to prevent overfitting. Robustness of CNN estimates to sound speed profile (SSP) variability is tested. Results suggest that the CNN approach is more tolerant of SSP mismatch compared to MFP at the cost of worse range resolution in its estimates when the SSP is accurately modeled. By examining our CNN model's filter weights and intermediate layer outputs, we present insights into how it generates its estimates and achieves its robustness. Potential benefits and drawbacks to this approach compared to MFP will be discussed as well.

[Work supported by ONR and NSF.]

Direction-of-arrival estimation using supervised machine learning

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Supervised machine learning, a pattern recognition framework, has been shown to accurately predict underwater acoustic source range and depth for broadband sources of opportunity. In previous studies, machine learning has been used to infer the relationship between source range or depth and the hydrophone array sample covariance matrix (SCM), its eigenvalues, or the received pressure amplitude across frequency.

In this study, we use the framework based on the complex SCM to examine direction-of-arrival (DOA) estimation as a machine learning problem. First, the complex SCM is reformulated as a real vector of concatenated real and imaginary components, $\mathbf{p}_{\mathrm{eff}}$. Conventional beamforming is formulated as a real, linear inverse problem in the weight space, $\mathbf{w}_{\mathrm{eff}}$,

$$B(\theta_m) = \mathbf{w}_{\mathrm{eff}}^{\top}(\theta_m)\mathbf{p}_{\mathrm{eff}} \tag{1}$$

$$\hat{w}_{\mathrm{eff}}(\theta_m) = \underset{\mathbf{w}_{\mathrm{eff}}(\theta_m)}{\arg\min}\{-\mathbf{w}_{\mathrm{eff}}^{\top}(\theta_m)\mathbf{p}_{\mathrm{eff}} + \lambda(||\mathbf{w}_{\mathrm{eff}}(\theta_m)||_2^2 - 1)\} \tag{2}$$

where θ_m is the candidate arrival angle, $\mathbf{B}(\theta_m)$ is the conventional beamformer output, and λ is a constant. $||\cdot||_2^2$ is the Frobenius norm.

This inverse problem may be solved by algorithms implemented in the feed-forward neural network or support vector machine methods. We show that the beamforming weights for a realistic perturbed array scenario can be quickly determined in this machine learning framework, allowing direction-of-arrival to be accurately estimated. This application demonstrates how machine learning software can be utilized to supplement model-based approaches, where training data is available.

Then, the direction-of-arrival problem is solved by developing a deep feedforward network with SCM inputs. The arrival angles $\theta \in [-90°, 90°)$ are discretized into a set of M labels, θ_m, $m = 1, \ldots, M$. Noiseless, plane wave simulations of two sources impinging on a linear array are used to train the deep network. To improve performance in noisy scenarios, a regularizing Gaussian noise layer is included in the deep network. A softmax output estimates the likelihood of each arrival angle, or output class. Test data is generated for two-source arrivals across 1000 random angle combinations and Gaussian noise is added at varying SNRs. The effect of source incoherence on the deep network is discussed. We demonstrate that directional-of-arrival estimation of two incoherent sources may be improved by increasing the amount of training samples. Finally, the direction-of-arrival estimation performance of the deep network is compared to conventional beamforming and Sparse Bayesian Learning for simulation data and for experiment data collected during the Swellex96 experiment.

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

Sediment classification of rough multilayer seafloors using localized finite element modeling and machine learning

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The key to a successful recovery of seafloor characteristics from measured backscatter data generated from SONAR systems is finding a balance between the computational cost of forward modeling and the desired resolution. In the high frequency regime, many propagation models are often limited by costly simulations or unrealistic environmental assumptions. To enable a rapid, remote, and accurate recovery of parameters such as sediment type, roughness, and layer thickness, we propose a combination of localized forward modeling and machine learning.

First, a reference library of acoustic seafloor responses from rough, multilayer seafloors are generated using finite element simulations of high frequency Helmholtz equations. A main challenge of feature generation is addressed with the wavelet scattering transform. We compare classification tools including supervised machine learning techniques and deeper architectures such as convolutional neural nets for improved accuracy. Our results shed light on the use of signal processing tools to design a machine learning framework for inversion that incorporates physics-based training libraries.

Passive source depth discrimination in deep-water using a horizontal line array

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When considering propagation of low-frequency ($f < 500$ Hz) acoustic waves, the underwater environment acts as a dispersive waveguide. In this context, propagation is adequately described using normal mode theory, and one should not use classical source localization methods based on plane waves or any other simplistic wave model. In this presentation, we focus on the problem of source depth discrimination using a horizontal line array (HLA) in deep water ($D > 1000$ m). The scope is further restricted to low-frequency broadband signals and distant source (range greater than several km) at the endfire position.

In this context, we propose a physics-based method that relies on the range-frequency intensity, as measured by the HLA. To perform discrimination, we propose to compute energy ratio between groups of modal interferences. Such groups are defined based on their respective waveguide invariant values, which in turns depend on source depth. This idea is formalized to propose a source depth discrimination method, which is performed as a binary classification problem. The proposed method is validated on simulated data. It is notably robust to environmental mismatch, and is shown to outperform existing depth discrimination methods if the environment and/or the array geometry are not well known.

[Work supported by Délégation Générale de l'Armement and Thalès Defense Mission Systems]

A computational Bayesian processor for active sonar target localization from sub-Rayleigh resolvable multipath arrivals in refractive environments

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Tracking of submerged targets in an ocean waveguide is challenging due to the deleterious effects of acoustic boundary interactions and the refractive nature of the ocean volume. When this challenge must be met with limited vertical aperture strains are placed on the signal processors ability to resolve closely-spaced phase fronts in angle and frequency as well as infer the range, depth, and speed. The solution must take into account both ocean refraction and differences in propagation path Doppler. A fairly efficient computational Bayesian scheme is presented here to address localization of such an underwater target by resolving these closely- spaced acoustic arrivals in a horizontally stratified ocean. A Gibbs sampling scheme is employed that provides a numerical solution to the joint posterior probability density (PPD) of the angles of arrival and the Doppler frequencies. This scheme takes full advantage of the tractable conditional densities of the complex amplitudes and the ambient acoustic noise power. However, conditional densities of the angles of arrival and Doppler frequencies are not tractable, nevertheless, their constrained a-priori domain admits the use of a fast 2-dimensional quantile sampling approach. To infer the location and state of motion of the submerged target, the PPD of the two dominant, scattered, refracted, and Doppler-shifted arrivals are mapped to the joint range, depth, and speed PPD. This is accomplished using a computationally efficient inversion approach based on acoustic eigenray interpolation. This posterior provides confidence intervals essential to submersible localization and tracking applications. A rich set of environments of varying complexity are considered using at-sea measured sound speed profiles. Simulations are presented to lend credence to the approach by exploring algorithm performance as a function of SNR and aperture. This algorithm can be employed recursively with successive PPDs serving as inputs to a Kalman-like algorithm for higher-level tracking applications.

Water column sound speed and sediment property inversion via Bayesian filtering and linearization with SBCEX linear frequency modulated signals

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Broadband receptions at a vertical line array (VLA) in the Sea-Bed Characterization Experiment (SBCEX) are considered for source localization and geoacoustic inversion. We focus on signals resulting from lfm transmissions that travel short distances and carry information between 1.5 and 4 kHz. Because of the short-range propagation, distinct path arrivals are identifiable such as the direct, surface reflection, bottom reflection, and first-second sediment interface reflection. Accurate, high-resolution estimation of arrival times of such paths leads to successful inversion for source location and water column depth and sound speed and, subsequently, estimation of sediment sound speed and thickness. Amplitude estimation leads to inversion for sediment attenuation. To achieve accurate inversion, Bayesian sequential (particle) filtering is applied to the received signals at the VLA phones. The filter works by constraining arrival times at a given phone to vary within specified intervals around arrival times at neighboring phones. Complete probability densities for arrival times are computed. Amplitudes are included in the state equation of the particle filter. Using linearization of the inverse problem, arrival time probability density functions are related in a straightforward manner to source and receiver location, water column depth, and water column sound-speed. Because probability densities of arrival times are propagated backwards through the linearized sound propagation model, posterior densities for the unknown parameters are calculated in addition to Maximum a Posteriori estimates. The densities are then used for grid-based sediment sound speed and thickness estimation. The results are consistent with prior information on the geometry of the experiment and the physics of the site.

[Work supported by ONR.]

Long Range Source Localization in the Philippine Sea using the Frequency-Difference Autoproduct with Cross-Term Corrections

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Frequency Difference Source Localization (FDSL) [Geroski and Dowling, 2019] has recently been applied to localize four sources based on signals measured during the PhilSea10 experiment [Worcester et. al., 2013] with 90or more snapshots. These sources were positioned near the sound speed minimum, approximately 1 km below the surface, of the ocean channel with source ranges varying from 129 to 380 km from a nearly water-column spanning vertical receiving array. The ocean sound channel varied in depth from 4 to 6.5 km over the experimental area. The transmitted signals were linear frequency modulated chirps with 100 Hz bandwidths and center frequencies between 250 and 275 Hz with an approximate signal duration of 135 s. Signals were compressed to a length of 1 to 3 seconds before localization processing was conducted. FDSL localized these sources by correlating the cross spectral density matrix (CSDM) of the frequency-difference autoproduct of the measured signals to replica autoproducts computed at out of band difference frequencies between 1 and 5 Hz, with a modeled environment represented by a single depth-varying sound speed profile with a perfectly rigid ocean floor at a constant depth of 6.3 km and a flat ocean surface. At such low difference frequencies, FDSL localizes distant sources despite challenging levels of environmental mismatch that cause in-the-signal-band matched field algorithms to fail. This study extends the previous work by analyzing undesired cross terms in the CSDM of the frequency- difference autoproduct that are generated as artifacts of the signal processing used to form this CSDM. These cross terms adversely affect the dynamic range, peak-to- sidelobe ratio, and noise rejection capabilities of FDSL; and can provide erroneous localization results even when using ideal data. These undesired cross terms are estimated and removed from a CSDM using several methods, including Monte Carlo estimation and diagonal subtraction methods, and updated source localization results based on both simulated and measured deep-ocean acoustic fields are shown. [Sponsored by ONR code 32.]

[The authors would like to thank Drs. Peter Worcester and Matthew Dzieciuch of the Scripps Institute of Oceanography for offering and sharing PhilSea10 propagation data, as well as for their useful insights and discussions regarding these measurements. The authors would also like to acknowledge useful discussions with Dr. Brian Worthmann in the development of this study.]

Designing universal adaptive beamformers for moving interferers

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Adaptive beamformers (ABFs) often struggle to attenuate loud moving interferers while attempting to detect quiet sources. ABFs must balance competing demands in choosing the sample covariance matrix (SCM) window length. An SCM averaging window that is too long will split interferers across multiple beams, reducing the ABF's performance in nulling the interferers [Baggeroer & Cox, 1999]. A window that is too short, the beampattern null will not be accurately steered toward the interferer. Multirate adaptive beamforming [Cox, 2000] and Null-Broadening [Song et al, 2003] both proposed approaches to address these challenges. A new adaptive beamformer proposed in this talk exploits techniques from the online machine learning community to find the best balance in SCM window length. This new universal ABF blends the array weight vectors across a set of ABFs with different SCM averaging window lengths. The recent performance of each fixed window ABF controls how much it contributes to the blended array weight vector for the universal ABF. The performance of the universal ABF provably converges to that of the best fixed-length window ABF for every finite power sequence. This convergence in the individual sequence sense does not require the assumption of a specific probability distribution. The universal ABF may perform better than all of the fixed-window ABFs in nonstationary environments where interferers change speed during the observation period.

[ONR Code 321 US]

Performance Weighted Blended Power Spectrum Estimation

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In classical spectrum estimation, windowing a time series before computing the periodogram improves the estimator's ability to detect a weak signal in the presence of a strong signal. This improvement is achieved at the cost of degraded resolution, i.e., the ability of the estimator to distinguish two signals that are close in frequency. Since many windows are available [Harris, Proc. IEEE, 1978] a fundamental question in spectral estimation is how to choose a window that provides the best trade off between resolution and dynamic range for a particular time series? In practice, analysts often synthesize information from periodograms computed with different windows to detect both loud and quiet sources. This presentation proposes a spectral estimator that automates that process by creating a blend of an ensemble of unbiased linear estimators.

The proposed estimator is an adaptation of the performance weighted blended beamformer developed by Buck and Singer [IEEE SAM 2018]. The blended beamformer computes its array weights as a mixture of the weights from an ensemble of adaptive beamformers, where the mixture coefficients are determined based on past performance. Buck and Singer observed that when beamformers are constrained to have unity gain in the look direction, the only difference in their outputs is due to noise and interference from other directions. Therefore, the accumulated output power is a good performance metric for the beamformers in the ensemble. The blended beamformer is guaranteed to perform as well or better than the best beamformer in the ensemble as the number of snapshots goes to infinity. Planewave beamforming is analogous to estimating the temporal spectrum of complex exponentials in noise. Thus the same approach can blend periodograms computed with different windows to estimate power spectra for time series, finding the best trade-off between resolution and sidelobes for each frequency of interest. The proposed estimator asymptotically converges, at each frequency, to the estimator with the best performance, i.e., lowest accumulated power in the ensemble.

Simulations demonstrate the performance of the proposed method using Chebychev windows, which are chosen because they provide the best possible resolution for a fixed sidelobe level. This property allows the windows to outperform the rectangular window in terms of resolution at the cost of severely high sidelobes, or to achieve extremely low sidelobes at the cost of very poor resolution. Complex exponentials with uniformly distributed random phase in Gaussian white noise were simulated. Stationary processes were tested as well as non-stationary processes. Non-stationary tests created a situation where no single estimator in the ensemble was the best performer throughout the simulation. Results show that improved resolution can be achieved in regions of the spectrum where there are loud closely spaced signals, while preserving the ability of the processor to detect quiet signals in other regions of the spectrum.

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Estimating frequency-wavenumber spectra using a universal dominant mode rejection processor

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Frequency-wavenumber spectra are useful for analyzing propagating waves and noise in the ocean. While conventional windowed Fourier transform estimators produce useful spectra, complicated environments often require high resolution estimators capable of isolating loud sources to preserve quieter parts of the spectrum. The minimum variance distortionless response (MVDR) estimator [Capon, Proc. IEEE, 1969] provides better resolution than the conventional Fourier estimator. This presentation focuses on a modified MVDR approach called dominant mode rejection (DMR) [Abraham and Owsley, IEEE Oceans, 1990]. DMR uses a structured estimate of the sample covariance matrix (SCM) to design the weights used to compute the spectrum. The structured SCM consists of a low-rank interferer subspace plus an orthogonal noise subspace. The DMR algorithm requires that the rank of the interference subspace is known or can be determined from the data. Buck and Singer recently proposed a universal DMR (uDMR) processor that uses a weight vector that is a linear combination of fixed-rank DMR processors. For stationary environments, the uDMR processor achieves performance equal to the best fixed-rank processor in its ensemble, and for non-stationary environments, uDMR can outperform all fixed-rank processors. This talk applies the universal DMR approach to estimate frequency-wavenumber spectra for experimental data from underwater vertical arrays. The discussion will focus on practical issues, including how to efficiently implement the uDMR spectral estimator for equally spaced arrays using fast Fourier transforms and how parameter choices affect convergence for short data receptions. [Work supported by ONR.]

[ONR 321US]

Improving the Robustness of the Dominant Mode Rejection Beamformer with Median Filtering

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Adaptive beamformers (ABFs) outperform conventional beamformers in detecting weak sources while attenuating background noise and strong interferers. An ABF such as MVDR relies on the inverse of the sample covariance matrix (SCM) compute the array of weights. However, the number of snapshots used to average the SCM must be greater than the number of sensors in order to generate an invertible (full rank) matrix. Fast-changing environments limit the number of snapshots that an ABF can average to estimate the SCM. The dominant mode rejection (DMR) beamformer [Abraham & Owsley, Oceans, 1990] overcomes rank-deficient SCMs by imposing a structured covariance matrix. For the DMR beamformer, the eigenvalues of the noise subspace of the SCM are replaced with an estimate of the noise power, traditionally an average of those eigenvalues. If the dimension for dominant subspace is underestimated, one or more loud interferers will not be attenuated and will contaminate the output. To avoid that, the DMR beamformer often overestimates the dominant subspace dimension. However, the overestimation introduces a negative bias and increased variance to the background noise power estimate, since eigenvalues averaged are smaller and fewer. This bias impairs the array gain and output signal to interferer and noise ratio (SINR) of the beamformer.

This talk proposes a modification to the DMR beamformer, replacing the average of the noise eigenvalues with an estimate of background noise power derived from the median of all eigenvalues. As long as the number of interferers is less than half of the number of sensors, the median-based noise power estimate does not vary when the estimated dominant subspace dimension changes. This makes the noise estimate robust against incorrect estimates of the noise subspace dimension. Simulations demonstrated that the median-based estimator of the noise power is more accurate than the average in scenarios which conservatively overestimated the dominant subspace dimension and suffered from a rank-deficient SCM. Also, this new median-DMR beamformer improves the output SINR by up to 0.9 dB compared to the standard DMR beamformer in the same conditions.

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DOA Estimation in the Presence of Local Scattering and Correlated Noise Using Co-prime Arrays

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Co-prime sparse arrays offer the capability to resolve a much higher number of sources as compared to a conventional uniform half-wavelength-spaced array. In this paper, we present a method for source direction-of-arrival (DOA) estimation using co-prime arrays in the presence of local scatterings and spatially correlated noise. We consider local scatterings which follow specific deterministic or statistical scattering models and assume the second-order noise statistics to be unknown. The proposed method simultaneously estimates the noise covariance matrix, the source power, and source direction by solving a non-convex optimization problem using a global optimization algorithm called covariance matrix adaptation evolution strategy. In order to further improve practicability in realistic scenarios, sensor positioning errors are introduced and an extension of the proposed method that is robust to positioning errors is discussed. Supporting numerical results are provided which demonstrate the effectiveness of the proposed DOA estimation approach.

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Clock drift estimation and correction across multiple asynchronous sonobuoy arrays

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Large sonar arrays are desirable because of their ability to detect and localize weak targets in complex and noisy underwater environments. These arrays achieve high spatial resolution with a narrow beamwidth and can adaptively suppress several sources of interference. In practice, array size is often limited to what can be reasonably deployed and maintained. Forming distributed arrays composed of multiple sonobuoy arrays offers a flexible and easily deployable way to achieve a large array without the physical challenge of deploying a fully- populated long baseline array. Processing distributed sonobuoy arrays present their own set of challenges. Inter-array calibration is required for joint, coherent processing of the received data. This requires estimating the relative sonobuoy locations, orientations and sampling differences due to asynchronous timing references. It is the sampling differences that this work focuses on.

Since the individual sonobuoy arrays do not share a common clock there is a fixed phase offset due to differences in the start recording times. Additionally, imperfect on-board oscillators have small deviations (on the order of parts per million) from the nominal sampling frequency which introduces a time-varying phase offset between data received on different sonobuoys. This causes stationary targets to appear oscillating when the full distributed array is beamformed. Sampling rate offsets, referred to as clock drift, have not been well investigated in the literature and their estimation and compensation preclude any kind of distributed array beamforming.

This work develops a narrowband signal model and estimation procedure for determining the clock drift between multiple sonobuoys in the presence of non-cooperative, discrete far-field noise sources. Sonobuoys are modeled based on air-deployable active receiver (ADAR) arrays with on-board GPS and heading sensors. Transiting cargo ships are modeled as noise sources of opportunity used to estimate the relative clock drifts. This work begins with a single stationary source with an unknown bearing. A Cramer-Rao lower bound (CRLB) is analytically computed on clock drift estimation for this sub-problem and shown to match the Maximum Likelihood Estimator at high SNR. For this problem we also show analytically that the clock drift parameter may be estimated without prior knowledge of the sonobuoys positions and start recording times, but only with prior knowledge of the orientations. This model is extended to the case of multiple, slowly moving sources. Finally, in-lab experiments using commercially available circular microphone arrays validate the signal model, drift estimation and compensation procedure.

[Office of Naval Research]

Measurements of matched-filter loss from sea-surface reflection

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Continuous active sonar (CAS) involves the transmission of long-duration, large-bandwidth waveforms. The coherence of these signals as they propagate underwater is a topic of interest. The first scientific measurements on shallow-water CAS were made during the TREX13 sea trial in 2013. The TREX13 data were subsequently analyzed to show coherence loss (matched-filter loss) that was larger for longer-duration signals and also for rougher sea surfaces. The success of TREX13 motivated the Littoral Continuous Active Sonar (LCAS) Multi-National Joint Research Project and beginning in 2015 there has been a large experimental effort led by the NATO Centre for Maritime Research and Experimentation (CMRE). One of the fundamental topics of the project is determining how to best process CAS that employs linear frequency modulation (LFM) waveforms. Processing with short-duration, low-bandwidth sub-bands can increase the sonar update rate (but with reduced signal excess), while processing with long-duration, large-bandwidth sub-bands potentially maximizes the signal excess (but with risk of matched-filter loss). The presented work includes results from LCAS experiments designed to measure matched-filter loss incurred in reflections from the sea surface as a function of the duration and bandwidth of the transmitted LFM signals. Limitations of processing duration and bandwidth must be known to optimally configure sonar settings for a given environment and sea state.

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Comb Waveform Design using Multi-Tone Sinusoidal Frequency Modulation

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Comb waveforms possess the ability to strongly suppress reverberation over a controllable range of Doppler shifts. This ability is measured by the Q-function, the Doppler marginal of the waveform's Ambiguity Function. In addition to Q-function shape, it is generally desirable for Comb waveforms to possess a low Peak to Average Power Ratio (PAPR) and an Auto-Correlation Function (ACF) with low sidelobes. These three design considerations conflict with one another and prevent a single waveform from optimally achieving all three simultaneously. Newhall train waveforms [Newhall, IEEE JOE 2006] possess a desirable Q-function and a low PAPR but suffer from high ACF sidelobes. Geometric comb waveforms [Cox, Asilomar Conf, 1994] have desirable Q-function and ACF properties but a higher PAPR. This research addresses whether the Multi-Tone Sinusoidal Frequency Modulated (MTSFM) waveform [Hague, IEEE RadarConf 2017] can smoothly trade-off between these three design criteria. The MTSFM waveform's modulation function is represented as a finite Fourier series. The Fourier coefficients are a discrete set of parameters that are adjusted to design waveforms with desirable properties. Analysis shows the MTSFM can naturally possess an optimally low PAPR and its design coefficients allow for smoothly trading off Q-function shape and ACF sidelobe levels.

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Optimal Target Detection for Cognitive MIMO Radar/Sonar

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Optimal linear filtering has been explored extensively for multi-input multi-output (MIMO) radar/sonar and its cognitive cousins. This is primarily due to its simplicity, practical implementation and computational advantages. Optimal waveform transmission and concomitant optimal receive linear filtering has been considered in [1,2], for example. Recent research efforts explore use of stochastic transfer functions (channel matrices) to capture the joint dependence of the target return/echoes and clutter/reverberation on the transmit waveform, where the optimal transmit waveform derives from a generalized eigenvector solution involving these channel matrices [3]. Detection theory, however, often suggests signal processing architectures that are not necessarily strictly linear. Indeed, radar/sonar literature is replete with examples of optimal nonlinear processing, e.g. see [4]. Thus, we explore the implications of a stochastic channel matrix on the resulting structure of optimal signal processing for a MIMO cognitive radar/sonar system. We focus on the commonly used Ricean channel model that contains both direct (discretes) and diffuse components, and can reduce to the Rayleigh channel model as a special case. Analysis of the optimal average likelihood ratio test demonstrates that a combination of matched subspace linear filtering and pre-whitened energy detection define the overall processing structure for the optimal detector. The matched filtering is defined by the direct components of the channel matrices, and the pre-whitening and energy detection is essentially determined by the diffuse components. Furthermore, the rank of the subspaces associated with these channel matrices translate into the required complexity of the receiver structure.

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[3] J. R. Guerci, J. S. Bergin, R. J. Guerci, M. Khanin, and M. Rangaswamy, "A new MIMO clutter model for cognitive radar," in 2016 IEEE Radar Conference (RadarConf), May 2016, pp. 1–6.

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Phase space representation of acoustic signals

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Phase space representation of time series has been developed as a tool to analyze dynamical systems and recent work have shown its utility in various real world applications, including vibrational diagnostics for engine fatigue detection and EEG signal analysis for seizure event prediction. Symbolized time series analysis using phase space representation has been used to characterize the evolution of states of dynamical systems and can be used to discriminate between different types of observed phenomena. For example, symbolized time series is modeled as a Markov chain and the estimated stochastic matrices from different events are used to evaluate the likelihood of the observed symbol sequence. In sonar, the data we record for underwater applications are acoustic pressure. These signals may be emanated from complex dynamical systems or scattered by objects with complicated geometries and a variety of elastic behavior. It is believed that using alternative representations, such as the phase space approach, for acoustic signals may allow us to access information embedded in the data in a more interpretable way, and features extracted from these representations will be more useful for detection and classification tasks. Due to the prohibitive cost of collecting underwater data and the need for well-controlled collection geometry and object configuration, we have devised a benchtop in-air acoustic data collection setup. We analyze the scattering responses from various objects with different material properties, with an emphasis on characterizing their elastic and inelastic responses.

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Sonar Signal Representation Using Sparse Gabor Dictionaries

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The primary motivation of this work is to utilize the interconnections between sonar acoustic color features towards robust sonar signal representations. The engineering goal is to achieve signal representation in an appropriate domain that renders high-precision low false alarm detection possible using popular signal processing techniques. Robust feature disambiguation has been a bottleneck to automatic target recognition for decades. In particular, feature disambiguation in real-ocean scenarios is challenging due to: (i) statistically significant but largely diffused interference due to environmental clutter, and (ii) more structured interference between acoustic features generated by different aspects of target geometry. We design active sonar signal representations for different targets that encode the scattering physics unique to a given target onto sparse dictionary representations. The proposed technique is highly appropriate for battlespace scenarios, e.g. for detecting manmade targets of unknown geometry, discovering moving covert targets that may exhibit transient albeit morphable spectral features, etc.

Technical approach:

(i) Design adaptive sparse dictionaries using the Gabor wavelet as a seed basis. We will use the Gabor wavelet as the seed model to capture unique target features that manifest as wave orbits in the time-frequency domain. The mathematical characterization of the Gabor wavelet aligns well with the scattering physics of sonar targets, and adapts to smooth morphological deformations. In the full paper, we will detail how the seed Gabor wavelet can be adapted through local diffeomorphisms (smooth invertible transformations) to depict the same target feature across different aspect angles.

(ii) Develop representations of individual target features using signal dictionaries based on the (morphable) Gabor basis. These dictionaries, by design, will typically exhibit a sparse distribution across a single target feature, and provide the building blocks of the proposed sonar signaling scheme. Specifically, the Gabor basis of sonar signals can be constructed based on signal models of known target response and used to probe unknown targets for potential match for specific characteristics such as target composition (aluminum, steel, etc.) and geometry (cylindrical UXO vs. others).

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Shallow Water Localization Estimates Using a Matched-Field Backpropagation Algorithm

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A typical matched field processing (MFP) approach performs source localization by correlating the fields measured on a hydrophone array with replicas of the modelled field for all candidate source locations. The replicas are generated using a full-field propagation model, like a parabolic equation (PE) model. The range-depth coordinate that produces the highest correlation between the measured and replica fields is used to estimate the source location. In this presentation we discuss results from a matched-field backpropagation algorithm. The backpropagated processor employed by this MFP algorithm is unique in that it combines the principles of acoustic reciprocity and linear superposition to simultaneously backpropagate data from all N sensors of a vertical line array (VLA). A correlation surface is generated in a single run of a PE propagation model by propagating the ambiguity surface outward from the array towards potential target locations. Using the backpropagated processor, calculation of the ambiguity surface is N times faster relative to a traditional MFP approach.

Environmental and acoustic data collected during an at-sea experiment were used to assess localization estimates from the matched-field backpropagation algorithm. The trials were conducted in a shallow-water environment (water depths ranging from 65 m to 155 m) using a towed acoustic source and moored Distributed Underwater Sensor Network (DUSN) nodes. Each DUSN node contains a nine-element VLA with a 40 m aperture and a hydrophone spacing of 4 m, resulting in a nominal design frequency of 178 Hz. A combination of transmitted low frequency signals from the towed source and opportunistic boat traffic will be considered as the source signals. Localization estimates from the matched-field backpropagation method will be compared to the true range and depth of the sources to assess the accuracy of each estimate.

Acoustic source localization: A non-Euclidean geometric approach

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Passive localization of acoustic sources is explored by comparing a cross-spectral density estimate of received data on a vertical array with a set of stochastic replica steering matrices, rather than comparing with replica steering vectors. A signal processing scheme involving matrix-matrix comparisons is considered where replica steering matrices, as functions of the replica source coordinates, naturally incorporate environmental variability or environmental uncertainty and provides a rather general framework for considering the acoustic inverse source problem in an ocean waveguide. Within this context a novel subset of matched-field processors has been developed, based on recent advances in the application of non-Euclidean geometry to pattern recognition of data feature clusters. The traditional Euclidean distance is not necessarily an appropriate distance measure because cross-spectral density matrices (CSDMs) are not arbitrary points in a vector space; rather, they form a Riemannian manifold constrained by the facts that CSDMs are both Hermitian and positive semi-definite. These properties naturally lead to the interpretation of geodesic distance between CSDMs as a measure of similarity between acoustic fields. This minimum distance, parametrized by replica source location, establishes an estimate of the true source position and defines a Riemannian-based matched-field localization processor where the matrices are interpreted abstractly as points in a Riemannian manifold. Acoustic simulations at a frequency of 300 Hz were performed for a waveguide comprising both a depth-dependent sound-speed profile perturbed by linear internal gravity waves and a depth-correlated surface noise field, providing an example of the viability of this approach to passive source localization in the presence of sound speed variability. An extension to include environmental uncertainty in the waveguide parameters or fields is also discussed, based on a polynomial chaos representation of the steering matrices. Work supported by the Office of Naval Research.

New Method for Source Depth Discrimination Using Information Geometry

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A new method for discriminating between surface and submerged sources based on the Riemannian geometry of covariance matrices is presented. The new technique is executed in two steps. First, replica covariances are generated for a known environment and then received signals are separated using simple clustering algorithms based on Riemannian distance measures between the sample and replica covariances. The novelty of this approach is that it works in environments which do not exhibit the phenomenon of 'mode trapping' that existing techniques rely on. Results are presented for a simulated environment with a sloping bottom and Gaussian sound speed perturbations in both the winter and summer at varying SNR. Comparisons with existing techniques and avenues for further refinement are also discussed. Work supported by the Office of Naval Research.

An Algebraic Geometry Approach to Passive Source Localization

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We examine the application of algebraic geometry-based localization methods to 3D underwater acoustic source localization. An approach to passive source localization that is commonly used in the electromagnetics community is to first determine the time difference of arrivals (TDOAs) by cross-correlating pairs of received signals, and then from the TDOAs use algebraic geometric methods to determine the source location. In particular, a TDOA for a receiver pair determines a hyperboloid on which the source must lie. Consequently the source must lie at the intersection of the TDOA hyperboloids. Because hyperboloids are examples of polynomials, systems of hyperboloids can be solved with numerical algebraic geometry software such as Bertini [1] or *Macaulay2*. Such software typically works by starting with a reference system of polynomials whose exact solution is known, and tracking the solution while gradually deforming the reference system into the desired one.

This approach is less well-known in the sonar community because the sonar wave propagation environment is more complicated. We begin by applying this algebraic geometry approach to the case of an iso-velocity range-independent sound speed environment. In this case the method of images can be used to convert a single-receiver problem to an equivalent free-space problem with multiple virtual receivers. The autocorrelation of the signal measured at the single receiver provides TDOAs for signals measured at pairs of the virtual receivers. With enough multiple-scattering paths and consequently TDOAs for sufficiently many pairs of virtual receivers, the source location can theoretically be determined from a single data stream. We also investigate extensions of this approach to underwater waveguides with non- homogeneous sound speed profiles.

[1] D.J. Bates et al., Numerically solving polynomial systems with Bertini, Society for Industrial and Applied Mathematics, 2013

Development of a Distributed MIMO Bathymetric Mapping System

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If the global ocean seafloors were mapped to a 1-m resolution, surface ships and undersea vehicles could potentially navigate in GPS-denied areas using topological features, or recognize new submerged manmade objects in a surveyed area. Less than 0.05% of the 360 million km^2 of ocean floor is currently mapped at this desired resolution. The two techniques that can achieve 1-m resolution, high frequency imaging sonar and airborne LIDAR, are both limited to a short standoff distance (10s to 100s of meters) from the ocean floor. Due to acoustic attenuation, a mapping sonar operated at the surface of the ocean in deep regions would need to be at frequencies less than or equal to 12 kHz. Between 10% and 15% of the ocean floor has been mapped using 12 kHz depth sounders in combination with large hydrophone arrays installed on surface vessels. This technique can achieve 100-m resolution in the deep ocean, limited by the available aperture on the ship hull. To alleviate the aperture limitation, we propose a sparse aperture distributed multiple-input and multiple-output array designed for deployment on a fleet of unmanned surface vessels. Our proposed array has an aperture 100 times greater than existing surface ship arrays, giving us the theoretical ability to achieve 1-m resolution maps of the ocean floor. We have built and tested a scaled model of this sparse aperture system, operating at 200 kHz, in a water tank facility and developed a mapping signal processing chain that creates topographic maps of the tank bottom covered by various submerged objects. Some of the challenges in this signal processing development include contending with platform motion and positional error, applying appropriate scattering model(s) of the bottom to allow for both specular and diffuse scattering within the downward-looking aperture, and in situ estimation of the propagating sound speed. We discuss the signal processing development, lessons learned and results from these tank tests, and plans to augment the signal processing chain to contend with the additional challenges presented by an upcoming ocean-going demonstration.

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Doppler-based single-hydrophone relative navigation for low-cost underwater vehicle swarming

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Underwater robot swarming has not yet been realized for a reason of basic physics: the attenuation of light in water is 10 orders of magnitude greater than that in air, which makes autonomous underwater vehicle (AUV) navigation either expensive or unwieldy, requiring high-power sensors such as inertial navigation sensors (INS) that cost hundreds of thousands of dollars, frequent surfacing for GPS, or the deployment of geo-located acoustic beacons. To do swarms of underwater vehicles will require inexpensive vehicles, a navigation and communication scheme that allows vehicles to remain together in an area while prosecuting a mission, and a multi-vehicle command and control scheme. The first technology is emergent, but the challenge remains to find a low-cost solution that keeps scalably large numbers of AUVs together in some kind of swarm or formation while collecting oceanographic data, and with an intuitive user interface that makes it possible for an operator to deploy, command, and recover large numbers of vehicles together. A possible solution to this problem in an integrated system that uses single-transducer, Doppler-based and multi-frequency attenuation-based acoustic navigation without time-synchronization for multi-vehicle swarming. In this system, a low-cost custom acoustic package provides the 'follower' autonomous vehicles with an estimate of the bearing and approximate range to a transmitting 'leader' vehicle. When a Doppler shift is present the vehicle maneuvers maximize Doppler shift, and therefore convergence between the following AUV and leader AUV with the pinger. As the vehicle turns, the solution converges. Approximate range is estimated based on attenuation difference between different frequencies. Absolute location may be reconstructed in post-processing, using Doppler shift caused by relative speed between the vehicles to estimate the angle between the vehicles and attenuation difference to estimate range. Preliminary results show the efficacy of this technique under different simulation conditions and in the field.

Analyzing models for goal management with operator input in intelligent active sonar

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Active sonar systems have a variety of transmit, receive, and processing parameters that can be dynamically modified to better achieve high-level surveillance goals. However, operators are tasked with interpreting large quantities of data, making it impractical for them to simultaneously adjust low-level system parameters, e.g., waveform, illumination sector, ping rate, etc. Hence, most parameters remain relatively constant, and the full capability of the sonar system is not exploited. Intelligent, or cognitive, sonar provides a framework through which the sonar system selects actions (parameter settings) to address one or more goals identified by the system and/or the operator. By translating high-level tasks to explicit actions, cognitive sonar allows the operator to take full advantage of system capabilities.

It is critical that an intelligent sonar system be able to seamlessly integrate input from the sonar operator. Unlike fully autonomous systems, intelligent sonar is designed to ease operator burden and help the operator make full use of system capabilities, not to remove human oversight from the sonar surveillance process. In particular, when multiple goals are active simultaneously, the intelligent system's mechanism for arbitrating among those goals must allow for operator input and feedback. A key design element for goal management in intelligent active sonar is constructing action selection criteria that incorporate both the relative urgency of each goal and the desire to achieve balance in addressing multiple active goals.

In this talk, we describe an approach to intelligent goal management that integrates operator input and facilitates joint consideration of multiple goals; we present simulation results to demonstrate its behavior. For each possible action a score is computed relative to each system goal. This score reflects how well each action makes progress toward each goal. The priority of each goal is then computed as a function of the best action score for that goal, the urgency of the goal, and the best action's effect on any global motivators (e.g., minimizing changes to the system setup). Goal urgency incorporates both system-computed risk (e.g., the nearness of a target in track) and priority information provided by the operator. Using Monte Carlo simulation of a simple scenario including search and track tasks and with waveform, illumination sector, and ping interval as action parameters, we study the effect of various priority functions on the behavior of the system with particular focus on the impact of operator input. In addition, we use simulation to study various approaches to goal management that address multiple goals simultaneously. These approaches include maximizing the overall gain (score weighted by priority), minimizing the largest loss, and searching for a Pareto optimal solution.

Adaptive and Compressive Sensing Beamformers Compared

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Adaptive (ABF) and compressive sensing (CS) beamformers solve different optimization problems. We will present some simulations comparing sensitivity to SNR and calibration errors, and then look at an interesting configuration which we suggest ABF handles better, presenting simulation and experiment results to demonstrate our claim.

The adaptive MVDR beamformer solves for the weights that produce the minimum beam power output, while constrained to maintain unit response in the beam direction. We must model the signal of interest for ABF to form this look direction constraint. Adjusting the weights to minimize the beam output power under such a constraint forces the weights to simultaneously maintain a unit response to the modeled signal of interest while at the same time adapting to all other signals. If a loud interferer is present, and is distinct from the modeled signal of interest, the optimization will adjust the weights so that a null is steered in the direction of the loud interferer.

The compressive sensing (CS) beamformer requires us to construct a "dictionary" of signal replicas and solves for the weights that minimize the squared error between the observed signal and a weighted sum of signal replicas from our "dictionary", with a regularization term consisting of the sum of absolute values of the weights. The genesis of this approach is to seek the sparsest set of weights possible. The strict solution to this problem yields no practical algorithm, since it entails testing all combinations of the weight vector (the l0 problem). The regularization is "relaxed" to minimizing the sum of absolute values of the weights (the l1 problem). Fortunately, practical algorithms exist for this modified problem.

In setting up ABF, note that we do not have to provide a model for the interference for ABF to cancel it. In CS, we will show what happens when our "dictionary" has a model for the quiet signal of interest, but not the loud interference.

An example of this is when we are looking for a quiet source in a flat part of the ocean, where we can readily model the signal, but we have loud interferers in an upslope region, where the bathymetry and seabed features are so variable that we cannot provide models for sources in this region.

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Wednesday October 16, 2019		**Thursday October 17, 2019**		**Friday October 18, 2019**	
		8:30–9:55	Session B Loc. I *Laurel*	8:30–10:15	Session F Waveforms & Proc. *Laurel*
		9:55–10:25	Break *Laurel*	10:15–10:40	Break *Laurel*
		10:25–11:50	Session C Loc. II & Mach. Lrn *Laurel*	10:40–12:00	Session G Loc. IV *Laurel*
		12:00–1:00	Lunch *Whisp. Pines*	12:00–1:00	Lunch *Whisp. Pines*
		1:00–2:25	Session D Loc. III *Laurel*	1:00–2:30	Session H Sys and Proc *Laurel*
		2:25–2:55	Break *Laurel*		
		2:55–5:00	Session E Array Proc. & conn. *Laurel*		
5:00–6:00	Welcome Reception *Whisp. Pines*	5:00–6:00	Hors d'ouvres de Capon *Whisp. Pines*		
6:00–8:00	Dinner *Whisp. Pines*	6:00–8:00	Dinner *Whisp. Pines*		
8:00–9:00	Session A Plenary *Laurel*	8:00–?	SOB Session *Laurel*		