A Book of Abstracts for the

2021 Underwater Acoustic Signal Processing Workshop

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Welcome to the 2021 IEEE workshop on Underwater Acoustic Signal Processing.

The organizing committee would like to thank and acknowledge the continued support of the Office of Naval Research. We are proud to announce that this year's recipient of the Donald W. Tufts UASP Award is Prof. Leon Sibul.

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The 2021 Donald W. Tufts UASP Award is presented to Prof. Leon Sibul

for extensive contributions to the theory and practice of underwater acoustic signal processing and mentoring many students and colleagues in the field

Leon's life was anything but ordinary. Born in Estonia in 1932, he attended high school at a German refugee camp and emigrated to the United States when he was 17. He soon joined the U.S. Air Force, serving overseas during the Korean conflict as an electronics and radio technician.

After his military service, Leon earned a B.S. in Electrical Engineering in 1960 from George Washington University, and made his way to Bell Labs to work on electronic switching systems and the communication satellite TELSTAR. While at Bell Labs, he earned an M.S. in Electrical Engineering from New York University in 1963. Then, in 1964 Leon started working for Penn State's Applied Research Laboratory (ARL), receiving his Ph.D. from Penn State in 1968, and remaining there until 2002 when he retired as senior scientist and professor of acoustics. He continued working part-time for ARL, advising graduate students until he passed away in 2007. His impact at ARL cannot be overstated. Many of Leon's 70 masters and doctoral students were full-time ARL employees, pursuing advanced degrees on their own time. His leadership made this difficult undertaking possible, always encouraging and innovating with a high degree of integrity and humor. From his vast life experiences, Leon was able to generate an endless stream of stories relevant to the topic at hand. For all of these reasons, he not only had the respect of his colleagues, but also their love and friendship. Many of Leon's students would go on to successful careers in Undersea Signal Processing. His intellectual energy was contagious, and his brilliance was masked by his quiet humility.

As head of the Signal Processing Department at ARL, Leon made major contributions and advised students in detection and estimation theory, wideband active ranging, independent component analysis, and wavelet-based processing; all squarely aimed at application in the undersea environment. Leon always went where his insight led him, and not necessarily where all the other researchers were going. This allowed him to make many interesting discoveries that would have otherwise been missed. His highly collaborative work is captured in 2 patents, 40 journal papers, 6 book chapters, 100 conference proceedings, and an edited book on "Adaptive Signal Processing." Despite working at the Applied Research Lab, Leon somehow found a way to investigate some of the most theoretical topics and always managed to make them "applied," keeping his sponsors happy. He made many important contributions to the science of adaptive signal processing algorithms, focused specifically on deploying them for next-generation torpedo guidance and control.

For his many contributions, the underwater acoustic signal processing community is honored to present the 2021 Donald W. Tufts UASP Award to Professor Leon Sibul and his family.

Contributed by Mark Fowler, Lora Weiss, Michael Roan, John Tague, Ashwin Sarma and Geoffrey Edelson

Schedule at a glance

Wednesday October 13, 2021		Thursday October 14, 2021		Friday October 15, 2021	
		8:10–9:50	Session B Acoustic Loc. I	8:10–9:50	Session G Modes/Comms/ML
		9:50–10:10	Break	9:50–10:10	Break
		10:10–11:00	Session C Acoustic Loc. II	10:10–11:50	Session H Active/Passive
		11:00–11:50	Session D Subspace/BMF I		
		12:00–1:00	Lunch	12:00–1:00	Lunch
		1:10–3:15	Session E Subspace/BMF II		
		3:15–3:35	Break		
		3:35–5:15	Session F Spectra/Stats/Det		
5:00–6:00	Welcome Reception	5:15–6:45	Tilted Barn		
6:00–8:00	Dinner	7:00–8:00	Dinner		
8:00–9:00	Session A Plenary	8:00–?	SOB Session		

Sessions: Titles and presenters

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

Plenary lecture

A–1 *Passive Sonar for Coronary Artery Disease (CAD) Diagnosis*
Norman Owsley, Phonoflow Medical Corp. (LLC)

Session B: Thursday Morning, 8:10am–9:50am

Acoustic Propagation-based Localization I

B–1 *Three-dimensional Acoustic Source Localization and Tracking in Submarine Canyons*
Ying-Tsong Lin, Woods Hole Oceanograhic Institution

B–2 *Long Range Cross-Correlation and Source Localization in the Philippine Sea Using the Frequency-Difference Autoproduct*
David Geroski, University of Michigan Applied Physics Program

B–3 *Vector sensor signal processing*
Gopu Potty, University of Rhode Island

B–4 *A Computational Bayesian approach to inferences regarding the state of motion of a mobile scatter in a refractive ocean waveguide under a limited vertical aperture constraint*
Abner Barros, University of Massachusetts Dartmouth

Session C: Thursday Morning, 10:10am–11:00am

Acoustic Propagation-based Localization II

C–1 *Gaussian Process modeling at a virtual array for source localization*
Zoi-Heleni Michalopoulou, New Jersey Institute of Technology

C–2 *Polarization of waterborne particle velocity and normal modes*
Julien Bonnel, Woods Hole Oceanographic Institution

Session D: Thursday Morning, 11:00am–11:50am

Subspaces and Beamforming I

D–1 *Some Relationships Among Adaptive Beamforming Algorithms*
Henry Cox

D–2 *Likelihood Ratios for Passive Detection and Localization with Two Arrays of Sensors*
Louis Scharf, Colorado State University

Session E: Thursday Afternoon, 1:10pm–3:15pm

Subspaces and Beamforming II

E–1 *A GLRT for Channel Matrix Based Data Model Adopted in Cognitive Radar/Sonar*
Christ Richmond, Arizona State University

E–2 *DOA Estimation using Optimal Subspace Estimates of Shift Invariant Sparse Arrays*
Kaushallya Adhikari, University of Rhode Island

E–3 *Mitigating Ocean Random Medium Effects in Array Processing by Using a Cross-Correlator Beamformer Instrumentation*
Ivars Kirsteins, Naval Undersea Warfare Center

E–4 *The Median Dominant Mode Rejection Beamformer Improves Robustness Against Array Element Perturbations*
David Campos Anchieta, University of Massachusetts Dartmouth

E–5 *Covariance Matrix Tapered MVDR Beamformer That Is Universal Over Notch Width*
Savas Erdim, University of Massachusetts Dartmouth

Session F: Thursday Afternoon, 3:35pm–5:10pm

Spectra/Statistics/Detection

F–1 *Performance Weighted Blended Power Spectral Density Estimation*
Jeff Tucker, George Mason University

F–2 *The K distribution and the pulse train model of noise*
Leon Cohen, City University of New York

F–3 *Universal Transient Detection on Sources With Unknown Statistics Via Universal Source Coding*
Andrew Finelli, Univ of Connecticut

F–4 *Modeling sonar performance using J-divergence*
Douglas Abraham, CausaSci LLC

Session G: Friday Morning, 8:10am–9:50am

Modes/Comms/Classification

G–1 *Modal Array Signal Processing: A Brief Introduction*
Kevin Bongiovanni, Raytheon

G–2 *Packet Specification Proposal for Underwater Acoustic Communications*
David Li, The MITRE Corporation

G–3 *Representation analysis of Circular Synthetic Aperture Sonar Data*
J. Daniel Park, Applied Research Laboratory The Pennsylvania State University

G–4 *Blind Equalization and Automatic Modulation Classification of Underwater Acoustic Signals*
Caitlyn Marcoux, The MITRE Corporation

Session H: Friday Morning, 10:10am–11:50pm

Active/Passive Processing

H–1 *Adaptive Signal Processing for HDC Sonar*
Travis Cuprak, L3Harris Adaptive Methods

H–2 *Joint Coherent Processing of a Randomly Drifting, Sparse Sonobuoy Array*
Brandon Hombs, Eigen LLC

H–3 *Wavefront Adaptive Sensing for Passive Underwater Acoustic Interference Suppression*
Anil Ganti, Duke University

H–4 *UUV Passive Acoustic Sensor Processing and Multi-source Tracking*
Ashwin Sarma, BAE Systems/URI

Abstract Listings

Passive Sonar for Coronary Artery Disease (CAD) Diagnosis

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So-called 'smart' stethoscopes have claimed superior CAD diagnosis performance relative to both 'dumb' stethoscope auscultation and blood flow perfusion stress testing. Both procedures continue to under perform clinical expectations relative to the gold-standard angiogram that, however, is both costly and invasive. Modern passive sonar has much to offer for the noninvasive detection, location and classification of low-level noise mechanisms, primarily coronary artery occlusion-induced turbulent blood flow that, all too frequently, is misdiagnosed by the perfusion stress test. First, this presentation reviews the engineering essentials necessary to invoke modern sensor array processing and high gain post-processing for low-cost, pre-symptomatic CAD screening and intervention follow-up. Next, examples of the analysis of human passive sonar data focusing on the anterior coronary arteries are presented and compared to independent, best-practice diagnoses.

Three-dimensional Acoustic Source Localization and Tracking in Submarine Canyons

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An underwater sound propagation experiment was conducted at the Alvin Canyon in the New England Shelf-break area in June 2021. Two vertical hydrophone arrays and one three-dimensional (3D) hydrophone array were deployed around the canyon forming a triangular listening network. All of the arrays were time synchronized via chip-scale atomic clocks and GPS signals. The objectives of this experiment include investigating 3D sound propagation effects caused by the complicated canyon bathymetry and source localization in this type of complicated undersea environments. A sound source emitting low frequency 115 Hz and mid frequency 1.25 kHz chirp signals was towed by a ship along the canyon axis, and the signals recorded on the three hydrophone arrays were utilized to test a number of localization methods, including triangulation with arrival time differences, matched field processing and back-propagation. A sound propagation model employing the parabolic-equation (PE) method is also established to assist the data analysis, especially to study 3D sound propagation effects. This listening network configuration also enables a test of utilizing a single 3D hydrophone array for localizing sound sources in a submarine canyon environment. Preliminary results of this field work study will be presented, along with a comparison to another canyon experiment conducted at the southern edge of the East China Sea.

[Work supported by the Office of Naval Research. The authors would like to acknowledge all of the participants in the New England Shelf Break Acoustics (NESBA) experiment for their collaboration. Thanks also go to the captain and crew members of the R/V Neil Armstrong.]

Long Range Cross-Correlation and Source Localization in the Philippine Sea Using the Frequency-Difference Autoproduct

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Matched autoproduct processors, including Frequency Difference Source Localization (FDSL) [Geroski & Dowling 2019] and Phase-Only Matched Autoproduct Processing (POMAP) [Geroski & Dowling 2021], have been applied to successfully localize moored acoustic sources transmitting at moderate (hundreds of Hz) frequencies over long (hundreds of km) ranges during the PhilSea10 [Worcester et al. 2013] experiment to a water-column spanning vertical line array. These methods work by cross correlating a measured cross spectral density matrix (CSDM) of the frequency-difference autoproduct of a measured pressure field with an equivalent (replica) autoproduct calculated with a user s knowledge of the environment at single-digit Hz difference frequencies. Because these methods allow the processing to be performed at such low difference frequencies, these autoproduct processors predict source location much more robustly than traditional source localization methods like Matched Field Processing (MFP) [Bucker 1976]. This study presents both cross correlation and source localization results obtained using FDSL and POMAP for all six moored sources measured during PhilSea10. Each of these six sources is placed at approximately one kilometer in depth, and the sources are arrayed at various latitudes and longitudes between 129 and 450 km away from the receiving array. The sources each transmit a linear frequency modulated chirp for approximately 135 seconds, with a center frequency between 172.5 and 275 Hz, and a bandwidth of either 60 or 100 Hz. These signals are then pulse compressed to a length of 1 to 3 seconds and zero-padded out to a length of 16 seconds for the purpose of interrogating the frequency-difference autoproduct at low difference frequencies. An acoustic sound speed profile measured near to the location of the vertical line array is used to compute the replica autoproducts. The ocean environment is assumed to be constant over the spatiotemporal extent of the experiment. Despite this assumption, both FDSL and POMAP are found to successfully localize all of the sources during the experiment, and POMAP is found to robustly and consistently localize the three closest sources that are within 250 km of the receiving array even with very little snapshot averaging.

[Sponsored by ONR]

Vector sensor signal processing

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Ocean Bottom Recorders (OBXs) are commercially available 3-axis seismic sensors with a co-located hydrophone. These sensor packages measure acoustic particle velocity along three mutually perpendicular directions in addition to pressure. In addition to these acoustic/seismic sensors OBX also provides its orientation via pitch, roll and heading sensors. Numerous algorithms are available in literature to estimate direction of arrival of acoustic/seismic waves using vector sensor data. Most of these algorithms were developed for a plane wave propagating in an infinite homogeneous medium. The OBX is located at the boundary between the water column and the seabed, that is at an interface of high impedance contrast. The OBX hence records the combination of different types of waves including incident, reflected, and refracted acoustic waves along with converted waves. For a given source excitation, several acoustic/seismic paths may be possible different directions and arrival times. The accuracy of DOA estimate depends also on the correct knowledge of the location and orientation of the OBX. Application of existing model-based DOA estimation algorithms to OBX data will be investigated using data collected on an array of OBXs deployed near Coastal Virginia Offshore Wind (CVOW) farm and in Narragansett Bay. The reliability of the in-built orientation sensors will also be assessed.

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A Computational Bayesian approach to inferences regarding the state of motion of a mobile scatter in a refractive ocean waveguide under a limited vertical aperture constraint

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A computational Bayesian approach to inference regarding the range, depth and speed of a submerged large mobile object is considered. Closely spaced coupled multipath arrivals in Doppler and vertical angle in a refractive environment are modeled in the presence of uncertainty in the ambient acoustic noise power. Vertical angles and Doppler frequencies of the arrival returns are jointly inferred. The scattering body allows energy transfer between eigenpaths adding to the complexity of the Doppler-angle spread scattered returns. From the posterior density of the arrival structure the posterior density of the object's range, depth, and speed is accomplished through acoustic ray interpolation. Case studies including that of the classic Munk sound speed profile are presented to lend credence to the approach.

[Work funded by the Office of Naval Research]

Gaussian Process modeling at a virtual array for source localization

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Matched-field processing for source localization entails the maximization of a correlation measure between received data at deployed hydrophones and replica fields calculated via an acoustic model for a set of values for source range and depth. Those values that maximize the correlation form the estimates. Matched-field performance is degraded by noise. This work alleviates the noise effect by using data to predict the field at a dense number of virtual receiving phones using Gaussian Processes. The predicted field, computed after the choice of a suitable kernel and associated hyperparameters, is correlated with replicas calculated at the same phone depths, leading to a Gaussian Process matched-field processor. It is demonstrated with Monte Carlo simulations for a range of Signal-to-Noise Ratios as well as with real data that the new Gaussian Process- based estimator has a superior performance to that of conventional Bartlett matched-field processing. Comparison of ambiguity surfaces computed with the two approaches illustrates that the Gaussian Process matched-field method reduces sidelobes. This improves source localization accuracy and reduces uncertainty in the estimation process. After the methods are compared in the case of a perfectly known propagation medium, environmental mismatch is considered. Matched-field inversion is then conducted using the two processors. The Gaussian Process based approach is still found to be superior to Bartlett processing in both source localization and environmental inversion.

[This work is supported by ONR.]

Polarization of waterborne particle velocity and normal modes

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This article proposes a physics-based signal processing framework to describe the polarization of waterborne particle velocity in ocean acoustics. We introduce the Stokes parameters, four real-valued parameters widely used to describe polarization properties in wave physics, notably optics. We then show how Stokes parameters can be estimated from the bivariate particle velocity signal using spectral properties. The concept of polarization spectrogram, which enables visualization of the Stokes parameters in the time-frequency domain, is also introduced. Finally, we detail polarization properties of normal modes, as measured using a single particle velocity sensor in the water column. The whole framework is illustrated on simulated data, as well as on experimental marine data collected during the 2017 Seabed Characterization Experiment.

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Some Relationships Among Adaptive Beamforming Algorithms

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Over the last several decades, a variety of algorithms have been presented in a vast literature on this general subject of adaptive beamforming. The goal of this paper is to present an overview, that stresses the close relationships among several algorithms. Common issues in applications include: mismatch between the assumed signal characteristics and the sensor outputs, problems of non-stationarity where only limited sample support is available, and signal suppression where inclusion of the signal in the sample covariance matrix can lead to signal suppression. Typically, algorithms involve additional assumptions leading to modifications of the sample covariance matrix or its inverse. There are many variations on this theme directed at overcome the issues of mismatch, limited sample support and signal suppression. A distinction is drawn between beamformers that produce an output time series that can be used for post-beamformer coherent processing and those array processors that do not. The Capon estimator and the related minimum variance distortion response MVDR beamformer provide a useful starting point for understanding many of the variants of adaptive array processing. Variations discussed include DMR, PCI, diagonal loading, white noise gain constraint, and sub-space approaches including MUSIC which is shown to be equivalent to the Capon estimator under the model assumptions of strong signals in white noise.

Likelihood Ratios for Passive Detection and Localization with Two Arrays of Sensors

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In this paper we establish a general first-order statistical framework for passive detection and localization with two arrays of sensors. We demonstrate that the resulting generalized- likelihood-ratio (GLR) detectors for the two-channel problem can be decomposed into two terms. The first term is a convex combination of the two GLR detectors that arise from considering each channel separately. This result is then modified by a fusion or cross-validation term, which expresses the level of confidence that the two single-channel detectors have detected a common source. Of particular note are the constant false-alarm rate (CFAR) detectors that allow for the scale-invariant detection in two sensor-arrays with different noise powers. The results are applied to the problem of passively detecting sources of acoustic or electromagnetic radiation.

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A GLRT for Channel Matrix Based Data Model Adopted in Cognitive Radar/Sonar

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The classical problem of radar/sonar target adaptive detection relies on both a primary data set (consisting of possibly target returns plus noise), and a secondary or training data set (consisting of noise only data samples). Kelly [1] derived a generalized likelihood-ratio test (GLRT) statistic for this problem when based on a radar data model that characterizes clutter via the data covariance matrix. Cognitive radar/sonar, however, has adopted a data model that characterizes both target and clutter via use of channel matrices because it simplifies the desired waveform optimization [2]. The present work derives a GLRT statistic for the classical adaptive detection problem but when based on the cognitive radar/sonar data model that uses channel matrices to characterize all waveform dependent components in the presence of additive Gaussian noise. The resulting GLRT statistic clearly illustrates the value of waveform diversity for channel estimation, and it has a similar form to the average likelihood-ratio test (ALRT) statistic derived in our previous work [3].

[1] E. J. Kelly, "An adaptive detection algorithm," IEEE Trans. Aerosp. Electron. Syst., vol. AES-22, pp. 115-127, 1986.

[2] J. R. Guerci, Cognitive Radar: The Knowledge-Aided Fully Adaptive Approach, Artech House, Inc., Norwood, MA, 2010.

[3] T. Ali and C. D. Richmond, "Optimal Target Detection for Random Channel Matrix- Based Cognitive Radar / Sonar," IEEE Radar Conference, Atlanta, GA, May 10-14, 2021.

DOA Estimation using Optimal Subspace Estimates of Shift Invariant Sparse Arrays

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Subspace-based signal processing algorithms use eigenvalue decomposition of the sample covariance matrix to find bases for the signal-plus-noise subspace and noise subspace. These subspaces are perturbed from the noise-free subspaces, especially in low SNR and snapshot deficient scenarios. An optimal subspace estimation (OSE) algorithm for datasets received by fully populated linear and planar arrays has been recently derived using the first order matrix perturbation theory [Vaccaro, Asilomar 2019]. The OSE achieves the Cramer-Rao bound on subspace accuracy. This work formulates a class of shift invariant sparse arrays (SISAs) that are designed to exploit the features of OSE. For DOA estimation with a SISA, we (a) obtain the optimal signal-plus-noise subspace basis using the OSE, (b) interpolate the missing elements in the optimal basis by imposing the shift invariant condition, and (c) apply ESPRIT using the interpolated basis. Comparison of the MSE estimates for a SISA and a coprime array with an equal number of sensors and aperture shows that SISAs DOA estimates are more accurate. The advantage of a SISA over a coprime array increases with an increase in the SNR and an increase in the power difference between sources.

Mitigating Ocean Random Medium Effects in Array Processing by Using a Cross-Correlator Beamformer Instrumentation

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Ocean random medium effects such as internal waves can cause distortions to signal wave fronts or equivalently, spatial coherence losses, that are highly detrimental to detecting and localizing signals with acoustic arrays. Deleterious effects to array processing include reduction of achievable array gains and large localization errors and biases. In principle, one could attempt incorporating models of the spatial coherence losses into optimum estimators or detectors to improve performance. However, the stochastic characteristics of the coherence losses are not well understood nor known a priori.

Motivated by the lucky imaging techniques used in astronomy [1] we have found evidence in real array data that even in environments with apparently low spatial coherence, when the data is looked at a much finer time scales, there can be brief moments when the wave front has little distortion that can be exploited to improve localization [2]. However, the challenge has been developing reliable metrics for ranking the quality of the data frames in order to detect and exploit the lucky moments since the signal-to-noise ratio is often low. In this paper, we propose a simple frame quality metric based on a cross-correlator instrumentation of the delay-sum beamformer [3] evaluated by arithmetically averaging the higher-order spatial correlation lags along the time delay trajectory corresponding to the beamformer focal location and then utilizing only the highest ranked frames in the localization. We connect this metric to beamformer focus sharpness by showing that the mainlobe curvature increases monotonically with the number of lags used and demonstrate the approach on simulated and real data.

[1] https://www.ast.cam.ac.uk/research/lucky

[2] I.P. Kirsteins, "A characterization of the instantaneous wave front distortion characteristics in SW06 and ASIAEX 2001 data and strategies for exploiting lucky scintillations," JASA 143, 1976 (2018).

[3] E. Ruigrok et al., "Cross-correlation beamforming," Journal of Seismology, 21, 495-508, 2017.

The Median Dominant Mode Rejection Beamformer Improves Robustness Against Array Element Perturbations

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Abraham's and Owsley's dominant mode rejection (DMR) beamformer [Abraham Owsley, Oceans, 1990] modifies Capon's minimum variance distortionless response (MVDR) [Capon, Proc. IEEE, 1969] to operate with low-rank sample covariance matrices (SCM). DMR estimates the ensemble covariance matrix (ECM) from a low-rank SCM by replacing the eigenvalues of the noise subspace with an estimated noise power based on the sample mean of those same eigenvalues. This estimated noise power is negatively biased when the dominant subspace dimension is overestimated. Practical implementations of the DMR often do overestimate the dominant subspace dimension to minimize the power of interferers in the output of the beamformer [Messerschmitt & Gramann, IEEE JOE, 1997]. The proposed median DMR estimates the noise power from the median of the nonzero eigenvalues of the SCM. The median estimator of the noise power is based on numerical evaluations of Marchenko-Pastur distribution of the eigenvalues of Wishart matrices. The median estimator is more robust to overestimating the dominant subspace dimension by exhibiting a lower bias squared and, consequently, a lower mean squared error than the mean estimator when estimating the background noise power. Simulations show that the median DMR improves the white noise gain (WNG) when compared to the standard DMR in snapshot deficient scenarios with overestimated interferer subspace dimension. The improvement in WNG resulted in a 1.1 dB increase in output signal to interferer and noise ratio (SINR). Higher WNG also implies increased robustness to array element perturbations, such as position errors or phase response of the electronic components of the array [Gilbert & Morgan, BSTJ, 1955]. This work compares the median DMR to standard DMR in simulations with perturbed array element phase responses in a scenario with two interferers and background white noise. The median DMR preserved deeper notches than standard DMR in this scenario, increasing the output SINR by roughly 1.3 dB.

[Work supported by ONR 321US]

Covariance Matrix Tapered MVDR Beamformer That Is Universal Over Notch Width

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Adaptive beamformers suppress interferers and reduce background noise by adjusting the complex array weights in response to the received array data. Practical adaptive beamformers like the minimum variance distortionless response (MVDR) beamformer [Capon, 1969] balance these two competing requirements while maintaining unity gain for the desired look direction. The MVDR beamformer places sharp notches in the location of the interferers to minimize the interferer output power. For stationary sources, the MVDR is an optimal beamformer, but the performance of the MVDR degrades in the presence of moving interferers. Interferers moving at different bearing rates reside inside beamformer resolution cells for different durations [Baggeroer & Cox, 1999], challenging MVDR s ability to place accurate notches in the interferer direction. Consequently, the moving interferer is generally no longer within the single sharp notch location. A more efficient approach to deal with the interferer motion could be applying a flatter and broader notch near the interferer location. One common approach to widen the notch in the beampattern is the covariance matrix taper (CMT) MVDR [Guerci, 1999]. Applying the CMT broadens the beampattern notches, mitigating the issues from moving interferers. However, the CMT increases the notch width by a fixed amount, and the best notch width depends on each interferer's unknown bearing rate, which may change over time. A single fixed CMT notch width cannot suppress all moving interferers perfectly. Therefore, the need for different notch widths for different bearing rates leads us to the possibility of designing a universal algorithm for this parameter. A universal algorithm is an online algorithm that processes the data sequentially. A common structure for the universal beamformer is blending competing solutions in a set of beamformers so that the universal beamformer's regret approaches zero asymptotically [Buck & Singer, 2018]. The regret is the difference between the universal algorithm solution and the solution of the best beamformer in the competing set. The regret for the universal beamformer can be proven to vanish asymptotically when we choose the blend weights correctly. Both software simulations and real data collected with a microphone array demonstrate that the universal algorithm matches or exceeds the performance of the best fixed notch width CMT beamformer in the presence of moving interferers.

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Performance Weighted Blended Power Spectral Density Estimation

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Estimation of a power spectral density (PSD) from a sequence of measurements is a common task in signal processing. The conventional Welch-Bartlett solution to PSD estimation is to split the data into blocks, compute the modified periodogram for each block, and average the results [Bartlett, Biometrica, 1950; Welch IEEE Trans. Audio Electroacous., 1970]. The modified periodogram is the magnitude-squared Fourier transform of the windowed data block. It is useful to visualize the periodogram as the output of a filter bank where each filter is a frequency-shifted version of a narrowband filter defined by the data window. The Fourier transform of the window determines the bandwidth and sidelobe properties of the filter. Since the Welch-Bartlett method is based on a single window (filter), it provides the same resolution and interference rejection at every frequency regardless of the input data. While Welch-Bartlett PSD estimators have the advantage of simplicity, they also require significant design tradeoffs, e.g., sacrificing resolution at all frequencies to ensure adequate rejection of a few loud tones. The spectral estimator proposed by Capon [Proc. IEEE, 1969] solves this problem by using a bank of adaptive filters. Each filter is designed to minimize the power output while maintaining unity gain at its center frequency. Thus the adaptive filter passes the frequency of interest with no distortion and attenuates other signals and noise. The Capon PSD estimator can achieve higher resolution and excellent interference rejection, but this comes at the cost of increased sample data requirements and greater computational complexity.

Buck and Singer proposed a performance weighted blended (PWB) filter that computes its weights as a mixture of weights from an ensemble of filters [IEEE SAM, 2018]. The PWB algorithm determines the mixture coefficients based on the past filter performance, as measured by the accumulated output power. When filters are constrained to have unity gain at the center frequency, the only difference in their outputs is due to noise and interference from other frequencies. Buck and Singer showed that the PWB algorithm produces a filter that asymptotically approaches the best performance of any of the filters in its ensemble.

This talk proposes a PSD estimator based on a bank of PWB filters. Each filter in the bank is a weighted combination of fixed-window filters. The filter ensemble used in the PWB approach includes filters with high resolution (rectangular window) and filters with low sidelobes (Chebychev windows). A final normalization step is required to convert the spectral estimate to a PSD. The talk compares PSD estimates from the PWB algorithm with estimates generated using the Capon filter and fixed window estimators. The PWB estimator achieves the high resolution of the rectangular window in parts of the spectrum dominated by loud sources while maintaining the improved sidelobe performance of other windows in quieter parts of the spectrum. Therefore it is able to outperform any of the fixed window estimators in the filter ensemble. Further, the PWB estimator requires much less data than the Capon estimator. The PWB estimator is an exciting new option for autonomous applications where choosing a good fixed window ahead of time is hard and in data deficient environments.

[Work is supported by ONR]

The K distribution and the pulse train model of noise

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The K distribution has found application to many fields of science and statistics. In underwater acoustics, the K distribution is considered the best fit for sea clutter data. Many plausibility arguments and derivations have been given for its applicability and effectiveness. We review these arguments and ask whether it can be derived from a fundamental model, namely by a pulse train model of noise. We give partial answers using theoretical arguments and simulations.

Universal Transient Detection on Sources With Unknown Statistics Via Universal Source Coding

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Quickest detection problems are fairly common to see in surveillance applications as framing problems in this field as a change in an observation sequence's statistics is often apt. Performance in these quickest detection problems are characterized by the average delay to detection ($\overline{T}_D$) and the average time between false alarms ($\overline{T}_{\mathrm{FA}}$). Canonical solutions that optimize the trade-off between minimizing $\overline{T}_D$ and maximizing $\overline{T}_{\mathrm{FA}}$ include Page's cumulative sum (CUSUM) test and the Wald sequential probability ratio test (SPRT), both of which require knowledge of observation statistics before and after transition to perform the tests. More recently, it has been shown that these tests can be modified using a universal source coding algorithm to remove the need for knowledge of the post-change observation statistics. In this work, we consider the scenario where an appropriate statistical description of our observations does not exist, neither before nor after the transient we are trying to detect. In this vein, we explore the use of a universal source coding algorithm, specifically, the Database Lempel-Ziv, or LZ77, procedure, to detect this transient in the observation data. The LZ77 algorithm has been shown to encode a source asymptotically and optimally without a priori knowledge of the source's statistics. LZ77 is also known to have phrase lengths that are asymptotically distributed as a Gaussian, which allows us to form a quickest detection problem around statistics of the LZ77 coded output and determine a change in the statistics from this processed stream of observations. This work specifies procedures to perform source agnostic transient detection using Locally Optimal (LO) and Generalized Likelihood Ratio (GLR) based tests, such as a variation of the Mean Agnostic Sequential Test (MAST). The work also seeks to quantify the effectiveness of this technique with respect to $\overline{T}_D$ and $\overline{T}_{\mathrm{FA}}$, and finally show an application of this novel algorithm to sound-based observation data.

Modeling sonar performance using J-divergence

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This presentation introduces the use of J-divergence as a performance measure for detection in a sonar system. The inherent inaccuracies between system-level performance and the cell-level (PD,PF) detector operating point used in traditional analysis open the door to using approximate performance measures such as J-divergence. The properties of J-divergence making it an appealing choice are covered: summing to accrue J-divergence across multiple independent measurements (e.g., from multiple source signals, waveforms, or arrays), a data-processing inequality dictating that processing cannot improve J-divergence, and an asymptotic relationship to the traditional (PD,PF) operating point. Simple forward models of J-divergence are presented for matched filters and energy detectors when applied to the standard signal models in Gaussian noise. A "design" J-divergence, which is chosen by the desired qualitative performance level, is used in simple equations to obtain the "design" SNR require to achieve it. This provides a direct replacement for the detection threshold (DT) term in the sonar equation that is easier to apply and evaluate. Example applications of J-divergence are presented illustrating its utility in the modeling and adaptation of current systems as well as the design and analysis of new ones.

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Modal Array Signal Processing: A Brief Introduction

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The scalar acoustic pressure sound field obeys the Helmholtz equation. It can be shown that under certain restrictions, pressure or particle velocity measurements on a closed surface can be used to uniquely infer the entire acoustic field. This observation leads to a different beamforming methodology for volumetric arrays where instead of combining filtered outputs of individual sensors, the measurements can be used to spatially decompose the array response in terms of an orthonormal basis defined by the array surface resulting in "eigenbeams". Modal beamforming is then performed by linearly processing over the eigenbeams to realize desired beampatterns. Since the number of modes can be significantly less than the number of sensors there is a computational advantage over classical methods. Additionally, modal processing offers a layer of abstraction (the "eigenbeams space") to accommodate a wide variety of array geometries which simplifies downstream processing tasks. However, sensitivities to shape deformation, such as an array in typical oceanic currents may undergo, are not amenable to classical error tolerance analysis and novel approaches to "array-healing" must be explored. To clarify these concepts an example for an hypothetical moored cylindrical array of sensors in deep ocean channel is shown.

Packet Specification Proposal for Underwater Acoustic Communications

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There is currently no digital acoustic communication standard adopted universally on the market, which introduces interoperability obstacles for users and developers. Underwater devices such as remote sensors, unmanned underwater vehicles, and manned platforms need to exchange information, but if their modems receive and transmit different packet structures, these devices cannot communicate. This work describes a packet structure appropriate for Underwater Acoustic Communication (UAC) with the objectives of minimizing overhead while permitting a system design that is based on a layered communication stack. This packet structure is applicable to a diverse set of UAC applications, and facilitates interoperability by permitting independent development at different layers. This specification does not constrain the modulation type, control mechanism, or media access approach. We describe a proposed compact packet structure compatible with a layered communication stack, comparing it to two alternatives in the open literature: the WHOI Micromodem and JANUS.

Representation analysis of Circular Synthetic Aperture Sonar Data

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In underwater remote sensing applications, synthetic aperture sonar (SAS) processing is used to generate imagery for environmental characterization or object detection. Images are primarily useful for representing the initial geometric response of acoustic scattering, but there are additional information embedded in the raw data that are not well-represented in images. For example, responses such as internal multiple scattering or elastic scattering are delayed in time relative to the geometric response, and the delayed arrival of these responses cause them to appear defocused in imagery, which in turn becomes less robust against external noise and interference. Complementary representations may improve the focus of these scattering responses, which can more reliably provide additional information about the environment or the object, and aid data interpretability. While images provide shape information in the spatial domain, resonant response of a cylindrical shell appear as traces of resonant peaks in the spatial wavenumber domain. Using an in-air circular synthetic aperture sonar data collection framework, we collected circular SAS data in a controlled and repeatable fashion with significantly less resources than underwater experiments. With an emphasis on learning physics-based features that will be reliably extractable across diverse environments, we employed a set of signal processing steps to separate geometric and elastic responses, and data augmentation techniques such as randomized image occlusion to train convolutional neural networks (CNN) for binary classification tasks with various extensions of spatial domain and spatial wavenumber domain representations. In this work, three classes of cylindrical objects, that are dimensionally identical but structurally distinct, are used to investigate how the choice of preprocessing and representation affect the ability of CNNs to capture different types of responses of the overall acoustic data and how they are utilized for binary classification tasks. This work is supported by the U.S. Office of Naval Research.

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Blind Equalization and Automatic Modulation Classification of Underwater Acoustic Signals

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Automatic characterization of underwater acoustic signals enables better use of the acoustic spectrum through activities such as interference avoidance and marine mammal protections. Determining the modulation of a received waveform can permit sonar and communications within the same bandwidth with minimal collisions, and it can identify systems operating outside their permitted regime. The characterization system determines the signal modulation in the presence of an unknown, time- varying channel impulse response. The work presented here demonstrates the use of blind equalization along with Convolutional Neural Networks (CNNs) for automatic classification of underwater signals. The current research focuses on classification of constant modulus signals since these signals are widely used for acoustic communications. The proposed approach provides an approximate 24 percent improvement in modulation classification compared to approaches without equalization and requires less training data than previous approaches. The constant modulus modulations examined here are binary phase shift keying (BPSK), quadrature phase shift keying (QPSK), minimum shift keying (MSK), frequency shift keying (FSK) and 8 phase shift keying (8PSK) signals. Simplified channels were simulated in MATLAB and the data were passed through the channels to analyze the approaches. Future work will include improving classification on more realistic channel models simulated via the Sonar Simulation Toolkit, verifying the results on underwater acoustic data collected from open water experiments, and expanding the types of signals analyzed.

Adaptive Signal Processing for HDC Sonar

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High Duty Cycle (HDC) or Continuous Active Sonar (CAS) systems offer high update rates compared to pulsed active sonar (PAS) alternatives. These systems also present unique challenges for receive processing. Due to the continuous transmission, any receive processing block may contain echoes, reverberation, and interference from all ranges in the receive cycle. The use of FM sweeps can be used to provide frequency isolation of these phenomenon, but there is a tradeoff between maximum processing range and system bandwidth. The application of adaptive algorithms such as adaptive beamforming can also be used to help mitigate and contain interference. However, obtaining necessary sample support is difficult in the non-stationary HDC environment. In this project, we are investigating techniques that take advantage of the characteristics of linear frequency modulated (LFM) waveforms to provide improved sample support for adaptive signal processing. We will describe the techniques currently being investigated and illustrate the performance and processing tradeoffs with these approaches.

[ONR321 (Dr. Keith Davidson)]

Joint Coherent Processing of a Randomly Drifting, Sparse Sonobuoy Array

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Wavefront Adaptive Sensing for Passive Underwater Acoustic Interference Suppression

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Detecting weak sources in underwater acoustic environments cluttered with surface shipping, biological sources, and other acoustic interferers remains a serious passive sonar challenge. Traditional detection methods consist of plane-wave beamforming and temporal spectral analysis which discriminate sources based on their spatial directionality and/or temporal frequency content, completely ignoring the effects of complex multipath propagation. Previous attempts to incorporate complex multipath effects in the signal processing chain have had limited success due to their reliance on detailed physical modeling of the underwater acoustic channel [1]. In this paper, blind source separation methods are used to adaptively estimate the complex spatial wavefronts of strong interference [2]. By exploiting their different source spectra and/or frequency-selective multipath fading characteristics, the approach constructs "signal-free" covariance matrices without the need to impose directionality constraints [3]. This facilitates separation of strong distant multipath interference from weak direct-path sources at shorter ranges. Simulation examples are presented comparing the signal-to-noise ratio gain achieved by wavefront adaptive sensing versus conventional and eigenvector-based adaptive beamformers.

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UUV Passive Acoustic Sensor Processing and Multi-source Tracking

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The objective of the presentation is to describe a framework useful for aiding the detection, classification, localization and tracking of distant targets using underwater acoustics when constrained by sensor size, capability, or platform noise. The core of this framework is a processing string that can estimate arrival angles of signal arrival wavefronts for a large class of signals that may possess tonal, broadband periodic, or noise-like characteristics with varying levels of temporal stability. As part of this, the processor continually returns multiple ranked signal hypotheses that can be exploited by subsequent processing to further separate any superimposed signals and provide greater system gain. These results are ultimately combined with measurements from other sensors in a dual-state multi-hypothesis tracker framework. An exemplary implementation employing an acoustic vector sensor onboard a UUV will be reviewed along with initial results from an at-sea data collect against vessels of interest in a high shipping density environment.

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Wednesday October 13, 2021		Thursday October 14, 2021		Friday October 15, 2021	
		8:10–9:50	Session B Acoustic Loc. I	8:10–9:50	Session G Modes/Comms/ML
		9:50–10:10	Break	9:50–10:10	Break
		10:10–11:00	Session C Acoustic Loc. II	10:10–11:50	Session H Active/Passive
		11:00–11:50	Session D Subspace/BMF I		
		12:00–1:00	Lunch	12:00–1:00	Lunch
		1:10–3:15	Session E Subspace/BMF II		
		3:15–3:35	Break		
		3:35–5:15	Session F Spectra/Stats/Det		
5:00–6:00	Welcome Reception	5:15–6:45	Tilted Barn		
6:00–8:00	Dinner	7:00–8:00	Dinner		
8:00–9:00	Session A Plenary	8:00–?	SOB Session		