A Book of Abstracts for the

2025 Underwater Acoustic Signal Processing Workshop

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Welcome to the 2025 IEEE workshop on Underwater Acoustic Signal Processing.

This year's Special Session on Underwater Acoustic Communications is organized by James Preisig and Ryan McCarthy. 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 Dr. James C. Preisig.

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The 2025 Donald W. Tufts UASP Award is presented to Dr. James C. Preisig

for contributions to underwater acoustic communications and propagation modeling

James Preisig received a B.S. degree in Electrical Engineering from the U.S. Coast Guard Academy in 1980. After graduation, he served on active duty in the Coast Guard from 1980 - 1985. He received the S.M. E.E. degree from the Massachusetts Institute of Technology in 1988, followed by a Ph.D. degree in 1992 from the MIT/Woods Hole Oceanographic Institution Joint Program in Oceanography/Ocean Engineering. From 1992 - 1994, he was a Postdoctoral Fellow at WHOI. After spending 1994 - 1997 as a Visiting Professor at Northeastern University, Jim returned to WHOI as a member of scientific staff from 1997 - 2013. Jim retired from WHOI in 2013 as a Senior Scientist Emeritus to found JP Analytics, LLC (Falmouth, MA) which he still leads as President.

Jim's major scientific contributions focus on models for underwater acoustic propagation for the communications channel, as well as adaptive and array processing algorithms exploiting these models. Collaborating with Grant Deane, Jim first identified the critical role scattering and focusing from the sea surface plays in creating high amplitude, high Doppler arrivals that challenge adaptive equalizers. He subsequently derived important sensitivity and performance bounds for adaptive equalizers in collaboration with his student Milutin Pajovic. Jim also identified the Delay Doppler Spread Function (DDSF) as a parsimonious representation of many time varying underwater acoustic channels. The sparsity of the DDSF makes it a better representation of the underwater acoustic channel than the previously used time-varying impulse response. Collaborating with his students, Jim also designed channel estimators and equalizers that exploit the sparse DDSF structure. All of these contributions are critical in distinguishing the underwater acoustic channel from terrestrial wireless channels. They propelled a generation of researchers in underwater acoustic communications to deeper understanding of the underwater channel, and how best to design systems to exploit it.

Jim has long championed the critical role of real ocean data in studying the underwater acoustic channel. He served in a scientific leadership role for more than a dozen major underwater field experiments, including the SPACE08, MACE10, and KAM11 experiments. The data gathered in these experiments continue to serve the community, providing opportunities for scientists and students to test their algorithms against real ocean data.

Jim's contribution to the underwater acoustic signal processing community is multiplied by the many students and postdocs he mentored at WHOI. Many of his former students continue to work on underwater acoustics and signal processing in leadership roles in industry, academia and government. His technical involvement in the SBIR space is worth noting as he steadily works with larger companies in coupling small footprint processors used for novel acoustic communication and networking concepts with Navy sonars and transducers.

Jim is a Fellow of both the IEEE and the ASA. He previously won the ONR Ocean Acoustics Young Faculty Award in 1999.

With this as preamble, the underwater acoustic signal processing community is honored to present the 2025 Donald W. Tufts UASP Award to Dr. James Preisig.

Contributed by John Buck, John Goodemote, Geoff Edelson and Ashwin Sarma

Schedule at a glance

Wednesday October 15, 2025		Thursday October 16, 2025		Friday October 17, 2025	
		8:00–9:40	Session B Special Session	8:05–9:20	Session H Geoacoustic
		9:40–10:00	Break	9:20–9:40	Break
		10:00–10:45	Session C Propagation	9:40–11:45	Session I Active Sonar
		10:45–12:00	Session D ABF I		
		12:00–1:00	Lunch	12:00–1:00	Lunch
		1:00–2:40	Session E Localization I		
		2:40–3:00	Break		
		3:00–3:50	Session F Localization II		
		3:50–5:05	Session G ABF II		
5:00–6:00	Welcome Reception	5:05–6:00			
6:00–8:00	Dinner	6:00–8:00	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 *Presentation of Don Tufts UASP Award*
multiple speakers followed by Plenary Talk by James C. Preisig, JP Analytics

Session B: Thursday Morning, 8:00am–9:40am

Special Session: Underwater Acoustic Communications

B–1 *Hierarchical Bayesian model selection for space-time varying acoustic response estimation with application to low SNR acoustic communications*
Paul Gendron, UMASS-Dartmouth

B–2 *Revisiting HIFT: OFDM Communication Across a Nine Million Meter Acoustic Channel*
Zhengnan Li, University of Alabama

B–3 *Representing rapid fluctuations in the shallow water acoustic channel in signaling schemes - opportunities, challenges and pitfalls*
Ananya Sen Gupta, University of Iowa

B–4 *Experiments with Underwater Communications using Hermetically-Orthogonal Frequency Division Multiplexed Waveforms*
Harvey C. Woodsum, VertoComm Inc.

Session C: Thursday Morning, 10:00am–10:45am

Propagation Modeling for Beamforming

C–1 *Beamforming and Acoustic Propagation, Part 1*
Henry Cox, Lockheed Martin RMS

C–2 *Beamforming and Acoustic Propagation, Part 2*
Henry Cox, Lockheed Martin RMS

Session D: Thursday Morning, 10:45am–12:00pm

Adaptive Beamforming I

D–1 *Data-Driven Wideband Focusing for Adaptive Beamforming with Short Observation Time*
Michael Martinez and Jeffrey Krolik, Duke University

D–2 *Universal Diagonally Loaded Beamformer Under Steering Vector Mismatch*
Akshay Bondre and Christ Richmond, Duke University

D–3 *SINR Loss Estimation: Expectation Under Diagonal Loading*
Justin Htay, Duke University

Session E: Thursday Afternoon, 1:00pm–2:40pm

Ranging, Localization and Navigation I

E–1 *Matched-field source-range estimation with a physics-informed machine learning approach*
Yongsung Park, Woods Hole Oceanographic Institution

E–2 *Automatic Detection and Localization of an Unknown Number of Low-Frequency Impulsive Signals Using a Distributed Network of Unsynchronized Hydrophones*
Mark Goldwater, MIT-Woods Hole Oceanographic Institution

Session F: Thursday Afternoon, 3:00pm–3:50pm

Ranging, Localization and Navigation II

F–1 *Three-Dimensional Tracking of Individual Targets Using AUV-Mounted Split-Beam Echosounders*
Camille Wardlaw, MIT-Woods Hole Oceanographic Institution

F–2 *Soundscape-relative underwater localization*
Erin Fischell, Acbotics

Session G: Thursday Afternoon, 3:50pm–5:05pm

Adaptive Beamforming II

G–1 *Performance Bounds for the Null-Steered Performance Weighted Blended Beamformer*
Jeff Tucker, George Mason University

G–2 *Approaches to snapshot segmentation for adaptive beamforming in time-varying scenarios*
Manan Mittal, Stony Brook University

G–3 *Perturbation Analysis of Toeplitz Rectified Covariance Matrices*
Kathleen Wage, George Mason University

<u>Session H</u>: Friday Morning, 8:05am–9:20am

Geoacoustic Inversion

H–1 *Feature-based sediment characterization with generalized additive models and machine learning*
Zoi-Heleni Michalopoulou, New Jersey Institute of Technology

H–2 *Robust Geoacoustic Inversion via Transport-based Metrics*
Christina Frederick, New Jersey Institute of Technology

H–3 *Geoacoustic Inversion of Sub-Bottom Profiling Data Using an Autonomous Underwater Vehicle Equipped With a Sound Source and Towed Hydrophone Array*
Paige Pfenninger, MIT-Woods Hole Oceanographic Institution

<u>Session I</u>: Friday Morning, 9:40am–11:45pm

Active Sonar: Mono and Multi Statics

I–1 *A Blended Active Sonar Receiver that Adaptively Trades Detection Gain for Range Resolution*
Radienxe Bautista, Naval Undersea Warfare Center

I–2 *The Bayes Factor for broadband active sonar for an uncertain depth scattering body*
Paul Gendron, Naval Undersea Warfare Center

I–3 *Domain Adversarial Neural Networks for Active Sonar Target Classification using Continuous Variables*
Allie Battista, JHU-APL

I–4 *Layout of Omnidirectional Hydrophones for Minimum Uncertainty Volume Target Localization*
Alex Kachergis, UCONN

Abstract Listings

Presentation of Don Tufts UASP Award

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Don Tufts UASP Award Lecture

The Promise and Challenges of AI/ML Techniques in Underwater Acoustic Communications

While AI/ML techniques have penetrated deeply into many different application areas and the advantages are clear, such is not the case in underwater acoustic communications. While results are being reported, there is no clear picture that has emerged as to the appropriate and advantageous application of AI/ML techniques in this area. After reviewing challenges to and tasks of an underwater acoustic communications system throughout the communications stack, this talk will review selected AI/ML results to attempt to generate a picture of the state-of-the-art. Continuing challenges and some thoughts on approaches to addressing them will be discussed.

Hierarchical Bayesian model selection for space-time varying acoustic response estimation with application to low SNR acoustic communications

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Subspace selection is a fundamental element of adaptive filtering for underwater acoustic communications. Invariably, whether by array tilt or dynamic motion of platforms, spatio-temporal subspaces must be selected wisely and based on both the observed acoustic field and the physics of the waveguide. In order to capture the space-time eigenpath structure accurately and ameliorate waveguide uncertainties, hierarchical mixture models are proposed and shown to be effective when coupled with a suitable spread spectrum signaling scheme. The result yields highly efficient statistical estimators of the acoustic response in angle-delay-Doppler spread environments and even at low SNR. We demonstrate this with M-ary orthogonal signaling and a mixture Gaussian model over beam angle, propagation delay and platform acceration associated Doppler processes. Each channel dimension in this space is coherent or incoherently reverberant. Coherent subspaces are not known apriori due to uncertainty in range, array tilt and motion dynamics, nevertheless they are a relatively small subspace and span the coherent time-varying eigen-structure of the response. The implied variable selection process is fulfilled in the hierarchy with indicator variables that specify the likely state of each channel dimension and a posterior density that is also mixture Gaussian and tractable in both point estimation and its higher moments. Iterative joint Bayesian posterior estimation of these coherent paths with the bit sequence supports communications at extremely low SNR.

[This work supported by the Office of Naval Research]

Revisiting HIFT: OFDM Communication Across a Nine Million Meter Acoustic Channel

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The Heard Island Feasibility Test (HIFT), conducted in 1991, evaluated the ability of man made acoustic signals to propagate across the world's oceans and served as a precursor to ocean acoustic tomography. The long-distance channel exhibited extreme propagation conditions, including non-minimum phase and multipath spreads approaching 20 seconds, at a time when effective methods for phase-coherent equalization was not yet developed. In this study, we used a BPSK modulated pseudorandom sequence transmitted during HIFT to extract the time varying impulse responses of the channel, which were subsequently employed to replay an orthogonal frequency division multiplexing (OFDM) signal for data transmission simulation. The HIFT signals operated at a symbol rate of 11.4 symbols/s with a center frequency of 57 Hz, and we analyzed a channel recorded at Ascension Island, corresponding to a 9 million meter path from Heard Island. Using OFDM, we achieved a data rate of 18.05 bps with an estimated mean squared error of -11 dB, demonstrating reliable communication across one of the longest and most challenging acoustic paths ever measured. These results highlight the unique challenges of ultra long range underwater propagation and the potential of advanced modulation schemes for basin scale acoustic communication experiments.

Representing rapid fluctuations in the shallow water acoustic channel in signaling schemes - opportunities, challenges and pitfalls

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Tracking the shallow water acoustic channel is a well-known signal processing challenge due to the rapid temporal fluctuations in the channel delay spread. Such rapid fluctuations typically occur due to multipath reflections from the moving ocean surface, fluid motion, surface wave focusing, acoustic scattering from the sea bottom, and other physical phenomena. While many channel estimation techniques have been proposed, the efficacy and accuracy of these computational methods are highly dependent on the time-varying channel representation itself. This talk will provide a brief review of the different mathematical representations of the time-varying shallow water channel and discuss the pros and cons of each representation. The talk will also cover the current state-of-art of opportunistic signaling schemes such as OFDM, OTFS, OSTM, etc. We will discuss potential overlap between prior art and new ideas and discuss challenges, opportunities and pitfalls in implementing some of these techniques in the shallow water acoustic paradigm.

Experiments with Underwater Communications using Hermetically-Orthogonal Frequency Division Multiplexed Waveforms

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Prior work presented in the application of Discrete Hermetic Transforms (DHTs) in Underwater Acoustics Signal Processing has focused on applications to beam-forming of ultra-small arrays. Current applications presented here study the application of Linear, Super-Resolution Transforms in the Time-Frequency Domain to the processing of Non-Orthogonal FDM signaling wherein sub-carriers with less than orthogonal spacing are modulated and demodulated using DHTs in order to increase spectral efficiency. Such applications have already been presented for the case of Radio-Frequency signals at UHF frequencies, the current work demonstrates acoustic signaling in an underwater environment at frequencies that could be useful for networking of autonomous undersea vehicles.

Beamforming and Acoustic Propagation, Part 1

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The physical concepts in underwater acoustic propagation are discussed emphasizing their relationship to beamforming for a vertical line array with which signal processing engineers are familiar. The presentation stresses fundamental concepts not computational methods and emphasizes geometric ideas. Both time domain and frequency domain approaches to beamforming for a vertical line array have their counterparts in ocean acoustic propagation. The relationships among beams, rays and modes are shown in Part 2.

In Part 1, the foundation is laid by reviewing familiar beamforming results, describing properties of the acoustic channel, and discussing propagation from the point of view of rays. Refraction in the water column, governed by Snells law and Fermats least time principle, and reflection at the ocean boundaries enable long range propagation. Almost everything follows from Snell's law, simple trigonometric relationships. and integration of slowly varying quantities over depth. The familiar ray-trace provides an intuitive, frequency independent description of propagation. It shows depth vs. range from a selected starting depth. The time-delay for beamforming a vertical line array and the transit time along the array have important parallels in acoustic propagation. The ray angle diagram (RAD), which shows depth vs. $\tan(\theta)$ contours for selected rays, is introduced.

Beamforming and Acoustic Propagation, Part 2

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Important characteristics and parameters of both beamforming and acoustic propagation depend on frequency. These include wavelength, beamwidth, number of orthogonal beams or modes, low frequency cutoff and interference patterns. The normal mode solution of the Helmholtz equation provides the frequency dependence for acoustic propagation. In order to satisfy the boundary conditions of the acoustic channel, there can be only a finite number of modes at any frequency. The N-th mode is described by a cyclic amplitude vs. depth pattern that has N positive or negative peak values. The Wenzel, Kramers, Brillouin (WKB) approximation visualizes a propagating normal mode as a coherent pair of upward and downward propagating locally plane waves that interfere to give the modal amplitude vs. depth pattern. Each pair is identified by a sound speed that corresponds to a ray parameter. The ray angle diagram (RAD) presents contours of depth vs. $\tan(\theta)$ for different values of the ray parameter. Thus, each mode is associated with a contour on the RAD. The WKB phase integral, the number of modes, modal cut off frequencies and the adiabatic invariant are proportional to "areas" on the RAD that are easily visualized. The correspondence between the propagation and beamforming parameters is identified.

Data-Driven Wideband Focusing for Adaptive Beamforming with Short Observation Time

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This work addresses the need to exploit signal bandwidth to estimate adaptive beamformer weights in highly dynamic environments. Traditional adaptive beamformers are limited by the observation time required to make stable estimates of narrowband covariance matrices across the signal bandwidth. Reducing observation time by simply averaging narrowband covariance matrices across frequency often results in interference "rank inflation" with a commensurate degradation in array gain. To avoid rank inflation, various model-based focusing methods, e.g. steered covariance [1], rotational signal subspace [2], and waveguide invariant [3], have been proposed to align directional components across frequency before averaging covariance matrices. These methods have had some success but are especially challenged by endfire interference (e.g. caused by tow-ship noise), particularly for long arrays. This paper introduces data-driven wideband focusing methods for passive sonar that optimize parameterized unitary matrices to align signal subspaces across the frequency band without relying on wave propagation models which are subject to mismatch in complex multipath environments. Inspired by machine learning methods, the proposed approach minimizes the log-determinant of the wideband covariance, a measure indicative of matrix rank, ensuring the coherence of wideband data and preserving SNR. We discuss two approaches: a fully-adaptive method with parameters scaling directly with the number of frequency bins, and a partially-adaptive method that shares parameters across frequencies to improve robustness to noise [4]. These focusing methods are paired with two new adaptive beamforming methods: 1) mode-selective minimum variance (MS-MV), which is a variant of the well-known dominant mode rejection (DMR) MV beamformer [5], and 2) and an adaptive beamformer classifier (ABC) approach which employs second-order blind source separation (BSS) rather than eigen-decomposition of focused covariance matrices [6]. Simulations are conducted in a shallow-water waveguide scenario to compare the performance of these methods for near endfire interference suppression with both short and long horizontal arrays. Results from the SWellEx-96 S59 event validate both approaches, showing improvement in tonal target detection and localization in the presence of strong dynamic wideband interference.

References:

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[2] H. Hung and M. Kaveh, Focussing matrices for coherent signal-subspace processing, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. 36, no. 8, pp. 1272-1281, 2002

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Universal Diagonally Loaded Beamformer Under Steering Vector Mismatch

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Universal adaptive beamforming (UABF) is applied to diagonally loaded ABF under the non-ideal conditions of steering vector mismatch and signal contamination of data. UABF is leveraged to optimize over the diagonal loading level which we choose as the regularization parameter. Assuming a statistical characterization for the steering vector mismatch, we use the average output signal-to-interference plus noise ratio (SINR) loss that accounts for signal mismatch as the loss function metric used by the UABF to determine the best coefficients for linear combination in the formation of the UABF weight vector. The SINR loss metric requires an accurate estimate of the interference-only covariance matrix in practice. Our initial goal here, however, is to focus on the ability of the UABF to choose good coefficients for linear combination that provide improved performance with respect to the choice of diagonal loading level. Thus, we consider a more idealized version of the universal beamformer, where it is assumed that interference-only training samples are available to compute this loss function metric. Numerical results showing the usefulness of this simplified beamformer will be presented that provide insight and motivation to consider more practical versions of the universal beamformer where signal-free training data may not be as available. We also discuss varying certain UABF framework-specific parameters and the resulting behavior of the UABF weight vector. Lastly, we discuss a potential approach to estimate the loss function metric consistently in order to applythe UABF in a practical scenario where signal-free training samples are not available.

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SINR Loss Estimation: Expectation Under Diagonal Loading

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Adaptive beamformers (ABF) use the covariance of the data received by a sensor array and the signal array response vector to compute optimal weights that maximize the output signal-to-interference-plus-noise ratio (SINR). In practical scenarios, where the covariance must be estimated and the array response vector is not perfectly known, system designers typically introduce a regularization term to the covariance matrix in a process known as diagonal loading. This technique yields robust beamformers that lie on a continuum between the unregularized optimal ABF solution and the conventional, data-independent beamformer (CBF). SINR loss measures the reduction in SINR relative to optimal ABF experienced by another competing filter. SINR loss was recently used as a design criterion for an alternative robust ABF approach that constrains the SINR loss of the optimal beamformer while minimizing white noise gain, as opposed to the typical approach that constrains white noise gain while minimizing output power. We analyze the asymptotic behavior of estimates of this SINR loss term using diagonally loaded estimates of the sample covariance matrix (SCM). Our results show rapid convergence of the SCM-based estimate of the SINR loss to its asymptote in the number of snapshots and sensors. By characterizing the behavior of estimates of the critical loss term, this analysis facilitates further development of the SINR loss constrained beamformer.

Matched-field source-range estimation with a physics-informed machine learning approach

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A physics-informed machine learning approach is developed to estimate source-receiver range in ocean acoustic environments. The framework performs matched-field range estimation without requiring explicit seabed acoustic information. The method leverages sparse pressure measurements and a known sound speed profile to train a physics-informed neural network (PINN) that predicts the acoustic field at the receiver location as a function of candidate source-receiver ranges (replica fields). By enforcing the Helmholtz equation as a constraint during training, the PINN maintains physical consistency while learning the acoustic propagation characteristics from data. Source localization is achieved by comparing measured pressure data with PINN-predicted replica fields using the Bartlett match function. Unlike traditional matched-field processing, which requires complete environmental information, including unknown bottom properties, our method implicitly captures environmental effects through data-driven learning. This physics-informed-data-driven approach enables accurate range estimation and holds promise for source localization and geoacoustic inversion in scenarios with only partial environmental information.

Automatic Detection and Localization of an Unknown Number of Low-Frequency Impulsive Signals Using a Distributed Network of Unsynchronized Hydrophones

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Most methods for sound source localization in the ocean, such as matched field processing or single-hydrophone modal filtering and inversion, require either expensive arrays of synchronized sensors or optimization over a high-dimensional source and environment parameter space for each signal of interest. Here, we consider 2D (latitude-longitude) source localization using a distributed array of unsynchronized hydrophones. The method is restricted to low-frequency sound sources and coastal waters. In our approach, a temporal convolutional network detects low-frequency dispersive signals of interest and independently ranges them on each sensor from which they are identified. Then, using the notion of group-k consistency, individual range measurements are automatically assigned to a unique source and outlier measurements, caused by unanticipated environments and/or source configurations, are rejected. The performance of the proposed method is thoroughly evaluated on a network of 17 bottom-mounted hydrophones (TOSSITs) in both simulation and on experimental data collected on the New England Mud Patch during the 2022 Seabed Characterization Experiment (SBCEX22). In simulation, our method is applied to source-receiver ranges with noise levels representative of both propagation- based range measurements as well as measurements from a synchronized sensor system with clock drift to demonstrate the flexibility of our approach. The SBCEX22 experimental data uses ground-truthed rupture induced underwater sound sources (RIUSSs). In both cases, our method demonstrates comparable performance to when data association and outlier measurements are known and is consistently able to identify the number of sources present.

[work supported by ONR and NDSEG]

Three-Dimensional Tracking of Individual Targets Using AUV-Mounted Split-Beam Echosounders

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Quantifying the behavioral responses of individual organisms in the deep scattering layer (DSL) requires precise three-dimensional localization and tracking in dynamic environments. Broadband split-beam echosounders mounted on autonomous platforms can provide fine-scale target positions and motion estimates, enabling assessment of individual responses to anthropogenic sound sources. This work applies broadband split-beam processing, 3D projection, and multi-target tracking to data collected from an autonomous underwater vehicle (AUV) operating at 300m depth within 500m slant range of a single 210c.u. airgun. Each detection was compensated for beampattern and spherical spreading, projected into global 3D coordinates using split-beam angles and AUV navigation data, and clustered into tracks based on spatial proximity. Extended Kalman filter tracking smoothed trajectories and provided per-ping velocity estimates.

Analysis revealed that a subset of individuals exhibited a consistent post-pulse dive response, characterized by rapid downward displacement and increased vertical velocity, while no significant changes occurred in the DSLs vertical structure or aggregation. These results demonstrate an integrated processing chain for high-accuracy 3D target tracking from a moving platform and its application to quantifying fine-scale behavioral change in mesopelagic organisms.

[Work supported by the National Defense Science and Engineering Graduate (NDSEG) Fellowship and the Office of Naval Research.]

Soundscape-relative underwater localization

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Underwater vehicles and sensing systems without GPS access require reliable non-drifting navigation information. While acoustic navigation infrastructure (e.g. gateway modems) cannot exist everywhere for navigation aiding, reliable and identifiable sources of anthropogenic noise in known locations already exist throughout the worlds oceans: shipping, pile driving, and seismic surveys. These sources of opportunity may be utilized for underwater navigation in real time or post processing by fusing relative acoustic localization with a priori understanding of the local soundscape. Addition of bathymetry information (e.g. from a depth sounder) provides a reasonably bounded navigation solution in many regions of the globe for AUV and drifting system navigation. Simulation and field postprocess examples are presented of how soundscape-relative underwater navigation might be utilized for drifting or mobile underwater sensing systems using array or single-hydrophone acoustic packages. Bearing and range are estimated along with associated errors; that information is filtered with available navigation streams to estimate relative position. AIS or other GPS reference on the soundscape location along with water depth is then used to get a maximum likelihood estimate of underwater position. The results show the potential for supplementation of existing navigation methods on systems carrying acoustic payloads by leveraging soundscape cues, as well as the limitations of the technique.

[Work funded in part by NSF]

Performance Bounds for the Null-Steered Performance Weighted Blended Beamformer

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The output signal-to-interference-plus-noise ratio (SINR) of an array processor determines its ability to detect and localize signals. SINR is a function of a processor's white noise gain (WNG), which quantifies signal-to-noise ratio improvement when the noise is white, and its interference leakage (IL), which measures the residual interference passed through the processor. A key metric for evaluating beamformers is SINR loss, defined as the ratio between the processor's output SINR and that of a clairvoyant Capon beamformer. The optimal Capon weights minimize the beamformer's output power subject to a unity gain constraint for the desired signal [Proc. IEEE, 1969]. Abraham and Owsley's Dominant Mode Rejection (DMR) beamformer [IEEE Oceans, 1990] uses the eigendecomposition of the sample covariance matrix to implement a Capon weight vector that attenuates the loudest interferers without sacrificing too much WNG. Hulbert and Wage [IEEE OJSP, 2022] bounded the SINR loss of the DMR beamformer using predictions of WNG and IL derived from random matrix theory. Tucker and Wage's Null-Steered Performance-Weighted Blended (NS-PWB) beamformer [IEEE SAM, 2024] also minimizes output power subject to a unity gain constraint. Instead of using the optimal Capon weights, it blends a set of fixed-taper beamformers based on their relative output powers. While NS-PWB may not achieve the same SINR loss as DMR in the steady state, it often converges with fewer snapshots and therefore offers improved performance in snapshot-deficient scenarios. This talk explores the worst-case SINR loss of NS-PWB and compares its performance to DMR.

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Approaches to snapshot segmentation for adaptive beamforming in time-varying scenarios

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An adaptive beamformer trades white-noise gain for interferer suppression by updating its sample covariance matrix. In time-varying environments, these updates are typically based on fixed sliding windows or forgetting factors, which require an a priori choice of stationarity regions. Such schemes degrade when the scene changes abruptly (e.g., interferers enter or exit), and naive averaging over all snapshots can underperform an omniscient, context-aware beamformer by expending degrees of freedom on interferers that are inactive while neglecting newly active ones - an effect that is pronounced with intermittent interferers.

We show that the optimal hindsight partition of snapshots can be computed from data via dynamic programming, an application of Bellman's optimality principle. In particular, the Capon beamformer is a special case of this formulation. Building on this, we propose universal methods over a class of time-partitioned beamformers and space-partitioned beamformers (in directional cosine) for snapshot segmentation. Tools from universal data compression and prediction for piecewise-stationary sources enable implicit implementation and performance-weighted mixing over an exponentially large model class with only polynomial complexity, using structures such as the linear transition diagram and a context tree. The resulting procedure adapts without preset window lengths or forgetting factors and competitively approaches the error of the best model in the considered class with vanishing regret.

Perturbation Analysis of Toeplitz Rectified Covariance Matrices

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Assuming planewave signals, the ensemble covariance matrix for a uniform line array is a Toeplitz matrix, while the sample covariance (SCM) is not. Toeplitz rectification imposes the desired structure by averaging the sub-diagonals of the SCM with the goal of improving performance when snapshots are limited. Quijano and Zurk [JASA, 2017] applied rectification to implement subspace beamforming for the Shallow Water Array Performance (SWAP) Experiment. Their beamformer uses the eigenvectors of the Toeplitz Rectified SCM (TR-SCM) to project data into the signal subspace. For SWAP, the TR-SCM beamformer achieves higher signal to interference and noise ratio (SINR) for a calibration tone than the SCM subspace beamformer. We analyzed the eigenstructure of the TR-SCM and showed that rectification degrades performance for quiet signals in the presence of loud interference [Chavali Wage, IEEE SSP Workshop, 2025]. The degradation is linked to cross terms (CTs) in the covariance calculation. A CT is the multiplication of two independent signals. Assuming zero mean signals, CTs disappear with snapshot averaging while straight-through terms in the covariance persist. CTs are often ignored by assuming long integration times or focusing on quiet environments. Neither assumption is reasonable for large sonar arrays. This talk presents a perturbation theory analysis of the TR-SCM that predicts the behavior of its eigenvalues and eigenvectors. The analysis motivates a new cross term aware (CTA) subspace estimator. Applying the CTA approach to the SWAP data set reduces artifacts and further improves SINR for the quiet signal.

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Feature-based sediment characterization with generalized additive models and machine learning

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Understanding seabed properties within an oceanic waveguide is essential for interpreting acoustic wave propagation and for enabling tasks such as sound-emitting source detection and localization. As a result, seabed characterization remains a central focus in ocean acoustics research. In this study, three distinct methods are applied to the problem of seabed identification, following a preprocessing stage that derives relevant features from time-series data obtained after the transmission of a broadband acoustic signal. The first method is based on unsupervised learning exploring whether the extracted features are indeed informative for seabed classification. After this is established, non-linear regression, namely the method of generalized additive models, is implemented, leveraging the extracted features for estimating geoacoustic parameters associated with the sediments. The third approach relies on the design of a decision tree classifier to perform sediment type identification. All techniques are evaluated using synthetic data representing five sediment classes and demonstrate successful recovery of information related to seabed characteristics.

Robust Geoacoustic Inversion via Transport-based Metrics

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We investigate the use of transport-based metrics for characterizing acoustic responses from ocean environments with varying sediment profiles. Unlike classical distances such as the least-squares norm (L^2), which are sensitive to amplitude noise and temporal shifts, the Wasserstein metric (W_2) from optimal transport theory can capture geometric features of wave signals induced by sound speed and other seabed properties. Since W_2 compares probability distributions, it requires preprocessing of geoacoustic signals to enforce nonnegativity and mass normalization. In contrast, the recently introduced HV metric is a transport-based distance specifically designed for signed waveforms. It measures the minimum action required to deform one signed signal into another via a PDE-constrained optimization, jointly penalizing spatial and amplitude mismatches while preserving waveform structure.

Using a synthetic dataset of acoustic responses generated via normal mode theory and Fourier synthesis over a grid of sediment thicknesses and sound speeds, we evaluate the ability of HV and L^2 metrics to recover environmental parameters from signals corrupted by distortion and additive noise. While the L^2 metric is highly sensitive to small phase shifts, the HV metric demonstrates robust performance across varying levels of signal corruption. Our results demonstrate the advantages of transport-based misfits over classical norm-based approaches in geoacoustic inversion, revealing that preserving waveform structure leads to more accurate and physically meaningful recovery of environmental parameters.

Geoacoustic Inversion of Sub-Bottom Profiling Data Using an Autonomous Underwater Vehicle Equipped With a Sound Source and Towed Hydrophone Array

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Numerical and experimental studies were conducted to investigate bottom geoacoustic inversion using arrival-time measurements of wide-angle seabed reflections from an autonomous underwater vehicle equipped with a sound source and towed hydrophone array. To address variability in returns from inhomogeneous bottom interfaces, multi-task Gaussian process regression was used to estimate the mean and variance of direct, bottom, and sub-bottom arrivals, which is then input into a Bayesian inversion scheme to inform the data covariance and ultimately update the prior distributions of geoacoustic parameters. Experimental data were collected during the Seabed Characterization Experiment at the New England Mud Patch in 2022. This method provides range-dependent geoacoustic parameter estimates in the experiment area with a resolution on the order of 10 meters.

Numerical results indicate that, for low-variance timing data, arrival times can be used to accurately estimate seabed properties, while increased variance reduces accuracy and strengthens the coupling between layer thickness and sound speed. Experimental data show substantial variability in sub-bottom arrivals, likely due to inhomogeneities and roughness at the mudsand interface. Results highlight the need for additional prior information to resolve ambiguities and uniquely constrain seabed properties.

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A Blended Active Sonar Receiver that Adaptively Trades Detection Gain for Range Resolution

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Active sonar systems interrogate their surroundings by transmitting pulses and listening for echoes. A classic approach to range estimation is pulse compression, where a high bandwidth pulse is transmitted and echoes are processed with matched filters. Matched filters optimize output SNR while range resolution is limited by the pulse bandwidth. Extensive research has focused on waveform design for sonar range resolution, often assuming matched filter receivers. However, relatively little research has explored alternative processing for active sonar receivers. Sharma and Buck (2011) proposed the variable resolution and detection receiver (VRDR), which smoothly adjusts range resolution between the matched filter and the inverse filter with one parameter, trading off detection gain for range resolution. In practice, the VRDR requires prior knowledge of the target and noise background to choose the best resolution and detection gain tradeoff. A new receiver is proposed to blend different VRDR receivers with different detection and resolution tradeoffs to adapt to the environment. The outputs of each VRDR receiver are weighted based on performance, using a mixture of experts' approach inspired by universal linear prediction [Singer & Feder, 1999] to implement the blending and minimize the regret of using a fixed individual VRDR filter.

The Bayes Factor for broadband active sonar for an uncertain depth scattering body

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The Bayes Factor (BF) is proposed for high frequency broadband active sonar with short vertical arrays operating in ocean waveguides. Relevant environmental information regarding the refractive media, rough surface and volume reverberation are incorporated under each of the composite hypotheses of null and alternative. Acoustic scattering off of a mobile object of interest under depth uncertainty characterizes the compound alternative hypothesis. Proper marginalization of the BF is juxtaposed to the more classical generalized likelihood ratio test (GLRT). The BF yields and is fully characterized by a set of time varying quadratic forms at each range of interrogation and it is shown to sensibly null non-target subspaces and optimally, in the minimum average risk sense, downweight reverberation subspaces. In this way inference regarding the presence of the mobile body of interest are determined against a composite null hypothesis of reverberation and ambient acoustic noise. We demonstrate a discriminating information expansion of the BF whose first term is associated with the depth invariant modes (DIM) of the waveguide that capture a significant portion of relevant energy associated with the scattering body's depth uncertainty. From such an expansion the performance of the BF is shown to be better than any specular arrival detector even if that specular detector had exact information regarding the depth of the target. In this way we show that the BF detector by sensible multipath combining outperforms a specular detector with exact information regarding the target depth. We present such lower bounds on performance the BF inferential approach with a few representative waveguides. Performance in conventional terms of probability of detection as a function of false alarm rate are presented.

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Domain Adversarial Neural Networks for Active Sonar Target Classification using Continuous Variables

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High clutter levels in active sonar data due to scattering of acoustic waves from man-made, biological, and geophysical oceanic features present a persistent challenge for target classification. Researchers have proposed deep neural networks (DNN) as a tool to help mitigate the clutter problem; however, the performance of the DNN varies due to oceanic parameters such as bottom type and depth. We propose the adoption of a Domain-Adversarial Neural Network (DANN) to learn a feature representation invariant to the oceanic environment. The DANN architecture includes a feature extractor and label predictor for learning the target classification. A domain classifier is then connected to the feature extractor via a gradient reversal layer. During training, the domain classifier competes against the label predictor, encouraging the feature extractor to learn features that perform well for the label predictor but poorly for the domain classifier, thus learning invariance across domains. While DANNs have traditionally been applied to problems with discrete domain labels and images, we investigate its application to a set of continuous labels for underwater acoustic classification. We utilize the Expanded Malta Plateau Clutter and Target Database (i.e., Clutter07) to leverage the environmental measurements collected in conjunction with the acoustic data. We train a baseline model composed solely of a feature extractor and label predictor in order to compare its performance to that of our custom DANN.

Layout of Omnidirectional Hydrophones for Minimum Uncertainty Volume Target Localization

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This work deals with a transmitter (TX) buoy and n omnidirectional receivers (RX) buoys on the ocean surface in a polygon pattern. A target (TGT) is located underwater somewhere in the neighborhood. Each RX measures (with additive noise) the TDOA between the direct signal from the TX and the reflected signal TX-TGT-RX. From these noisy TDOAs we use the Maximum Likelihood (ML) algorithm to estimate the 3D location of the target. We evaluate the Cramer- Rao Lower Bound (CRLB) and show statistical efficiency of the estimator using hypothesis testing. An explicit solution is presented for the optimal TX-RX layout for minimizing the uncertainty volume of a target at an arbitrary depth.

Wednesday October 15, 2025		**Thursday October 16, 2025**		**Friday October 17, 2025**	
		8:00–9:40	Session B Special Session	8:05–9:20	Session H Geoacoustic
		9:40–10:00	Break	9:20–9:40	Break
		10:00–10:45	Session C Propagation	9:40–11:45	Session I Active Sonar
		10:45–12:00	Session D ABF I		
		12:00–1:00	Lunch	12:00–1:00	Lunch
		1:00–2:40	Session E Localization I		
		2:40–3:00	Break		
		3:00–3:50	Session F Localization II		
		3:50–5:05	Session G ABF II		
5:00–6:00	Welcome Reception	5:05–6:00			
6:00–8:00	Dinner	6:00–8:00	Dinner		
8:00–9:00	Session A Plenary	8:00–?	SOB Session		