Award #1 Title
Acoustic Communications 2011 Experiment: Deployment Support and Post Experiment Data Handling and Analysis
Award #2 Title
Exploiting Structured Dependencies in the Design of Adaptive Algorithms for Underwater Communication
Award #3 Title
Coupled Research in Ocean Acoustics and Signal Processing for theNext Generation of Underwater Acoustic Communication Systems

James Preisig

Woods Hole Oceanographic Institution
Dept. of Applied Ocean Physics and Engineering
Ocean Acoustics and Signals Laboratory
Woods Hole, MA 02540

AND

JPAnalytics LLC
638 Brick Kiln Road
Falmouth, MA 02540

phone: (508) 566-0236 email: jpreisig@whoi.edu email: jpreisig@jpanalytics.com

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LONG-TERM GOALS

A high performance, versatile, and reliable underwater communications capability is of strategic importance to the U.S. Navy. Operational scenarios involving the use, monitoring, and coordination of multiple undersea assets, both manned and unmanned and mobile and fixed, are significantly enhanced by the ability to communicate quickly, reliably, in a wide range of environmental conditions, and with minimal constraints on the actions of the platforms involved. The long-term goal of the effort in underwater acoustic communications is to integrate research in the areas of physical oceanography, ocean acoustics, signal processing, and communications theory to

1. develop a underwater communications capability that can be deployed on and used by a wide range of platforms with minimal required external infrastructure and which achieves reliable and high performance under a wide range of environmental and operational conditions, and
2. develop an communications performance prediction capability that enables commanders to

reliably predict the performance of deployed communications systems as a function of environmental conditions and the location, velocity, and capabilities of the deployed assets.

OBJECTIVES

The objectives of the combined work supported by the three grants/contracts are to support the Long Term Goals by developing a fundamental understanding of the propagation physics at frequencies and in scenarios relevant to the acoustic communications challenge and how those physics depend on the physical oceanography and pushing the state of the art in our understanding of adaptive signal processing algorithms relevant to communications receivers. The objectives include

1. Work with Dr. William Hodgkiss, Scripps Institution of Oceanography, to conduct the KAM11 field experiment and gather a significant data set of spatially and temporarily coincident environmental and acoustic data with the acoustic data including receptions of ambient noise, general purpose channel probe signals, and specialized communications signals that are to be evaluated for specific applications. Perform appropriate quality control on the data and make it available to KAM11 participants.
2. Investigate and develop an understanding of the spatial and temporal scales and the statistics of acoustic propagation fluctuations at communications relevant frequencies including the MF (3 to 30 kHz) and UHF (300 kHz to 1 MHz) regimes and their impact on communications systems performance. This includes developing and validating models of the impact of sea surface and upper ocean boundary layer processes on the performance of underwater acoustic communications systems and develop an deployable VHF acoustic data transmission and acquisition system.
3. Develop signal models and processing algorithms that reduce to the extent possible the degrees of freedom (DOF) used to both represent the received signal and to adapt signal demodulation algorithms to accommodate the time variability of the acoustic channel.
4. Develop, analyze and predict the performance of adaptive decision feedback equalization algorithms that are suitable for use with receive arrays with large numbers of elements and which offer improved performance and reduced computational complexity when compared to standard full rank, hard decision directed adaptive equalization algorithms. Develop new methods for predicting the performance of adaptive equalization algorithms in the underwater acoustic environment and use them to determine optimal equalizer and array configurations in realistic environments.
5. Assist in the transition of developed methods into prototype products for Navy underwater acoustic communications system.

APPROACH

The approach taken to consists of a combination of analysis of field data, the development of new theory for adaptive signal processing algorithms, and the development, testing and analysis of new algorithms.

1. The conducting of the KAM11 field experiment was a straight forward exercise in experiment planning and execution with a large group of PIs. The approach was to gather PI requirements

signal transmissions and signal and environmental measurements, coordination with the ship and PMRF, working with technicians at WHOI and Scripps to plan deployment and recovery operations, and scheduling of the min-experiments that we incorporated into the overall KAM11 experiment. Post-cruise, the approach was to analyze the data in the sequence of transmissions, identify time periods when there were either transmission or reception problems, correct the problems when possible, review the environmental data to determine a representative set of time periods, and then quality check the data in those time periods and make it available to other KAM11 experiment participants. Now that we are past the two year post cruise time period, data from the KAM11 experiment is being made available to other researchers in the field on request.

2. Objective 2 is being pursued in collaboration with Dr. Grant Deane, Scripps Institution of Oceanography and Dr. Andone Lavery, Woods Hole Oceanography. The approach is to develop analytical and numerical models to characterize the channel impulse response from a transmitter to a receiver resulting from surface scattered arrivals. This characterization includes intensity fluctuations due to surface shape and the path length fluctuations due to surface motion. Also included are the effects of near surface bubbles on the intensity and time-variability of the surface scattered arrivals. This characterization is carried out with the analysis of derived expressions, numerical evaluation of expressions, and numerical simulations. The derived expressions and numerical models are validated by comparison between predicted impulse responses and those measured during the KAM11 and SPACE08 field experiments. Environmental measurements taken during those two experiment is used as the inputs to the numerical simulations used to generate the impulse response predictions. In addition, a new field deployable VHF data acquisition and transmission system is being developed for the 375 kHz to 725 kHz band to support the work in that region and initial tests in the near shore region and in a wave tank are being conducted. This work was supported by Award 1 and is now supported by Award 3.

3. The approach taken to achieving low-snr detection of and synchronization to communications signals it to hypothesize that a specific location (time) in a received signal is the starting point for a known detection and synchronization portion of a received signal and train a specially constructed adaptive equalizer to demodulate the signal known signal. If the signal is actually present and it is a bpsk modulated signal, we would expect that the equalizer would achieve a bit error rate of something less than 0.5 (random guessing for a bpsk signal) indicating the detection of and synchronization to a communication signal. If there is no communications signal present, then the equalizer would be expected to achieve a bit error rate of approximately 0.5. indicating that there has been no detection. This work was pursued with the support from award 1.

4. Adaptive, multi-channel, phase-coherent equalizers reliably achieve higher performance in terms of achievable data rates in underwater acoustic channels than other forms of signal detection and demodulation algorithms. However, in their customary form they have a large number of parameter that need to be adjusted. This large number of parameters results in both a high computational complexity of the equalizer adaptation algorithm and the need to utilize a long averaging interval in order to achieve stable operation. This long averaging interval decreases the rate of unmodeled acoustic channel fluctuations that can be successfully tracked and compensated for by the equalizer. This poses a significant challenge in the underwater acoustic environment.

   The approach to addressing the ”large number of parameters” problem is multifaceted. It includes developing adaptive subarray processing methods for transforming the single high dimensional adaptation problem into one or more lower dimensional problems as well as iterative methods for equalizing and decoding received signals. Supporting results in the field of *Random Matrix Theory* are developed to assist in the guidance of optimal methods and the prediction and analysis of their performance. The approach to developing an adaptive equalizer capable of operating within the iterative framework of a turbo-equalizer will be to use soft symbol information in the equalizer adaptation process place more weight on symbol decisions which are believed to be reliable and to discount the decisions which are not. The Expectation-Maximize (E-M) framework will serve as a starting point for this development. This work will be pursued primarily with the support from Awards 1 and 3. Finally, the covariance matrix of the filter regressor vectors (the frequency domain input signals to both the feedforward and feedback filters) posses a predictably sparse structure that will be exploited to improve performance and/or reduce computational complexity with respect to an adaptive equalizer that does not exploit any particular structure of the matrix. This later approach will be pursued primarily with the support of Awards 2 and 3

WORK COMPLETED

This year the completed work and new work started has been extensive. A new framework for using Random Matrix theory in the analysis of time-varying adaptive equalizers has been developed. We completed work including characterizing the performance of multichannel linear and decision feedback equalizers as a function of filter parameters (the number of taps in each of the filters), array configuration (spacing between elements in a linear array), and the acoustic propagation conditions. This work has been extended under the support of contract #N00014-14C-0230 to create a model of the sample covariance matrix of the input signal that both explicitly models the time-variability of the acoustic channel and is amenable to analysis using the tools of asymptotic random matrix theory and the analysis of the model is on-going.

The second area in which work as been completed concerns the exploitation of the predictable sparse structure of the input signal covariance matrix in a frequency domain equalizer. Work completed includes the development of a graph structured least squares algorithm and an iterative E-M based method for adapting equalizer coefficients. The analysis of the frequency domain signal covariance matrix structure has also been completed and it being prepared for publication.

Finally, the first phase of the development of the VHF acoustic transmission and data acquisition system has been completed. In addition, two VHF acoustic wave tank experiments and one surf zone VHF ambient noise experiment using legacy equipment have been conducted in collaboration with and with the support of Dr. Grant Deane at the Scripps Institution of Oceanography. The analysis of data from these experiments is on-going.

RESULTS

The work on random matrix theory has let to the interesting result that the optimal configuration of both linear and decision feedback multi-channel equalizers in a time-varying ocean environment utilized filter lengths that are much shorter than the delay spread of the acoustic channel. This I addition, the new framework illustrates that simple and realistic assumptions on the decorrelation of certain sample sums in the channel's Delay-Doppler spread function leads to a model of a time-invariant correlation matrix in the least squares normal equations. The implications of this result is still being analyzed.

The work on the exploitation of sparse structured covariance matrices in equalizer adaptation yielded significant results in several areas. The analysis of field data from the SPACE08 and KAM11 experiments has shown the that graph structured least squares algorithm both reduces computational complexity and improves algorithm performance when used for adaptive equalization. Furthermore, the E-M based algorithm provides a framework that generalizes the more restrictive graph structured approach.

It was first demonstrated the that the sparse structure exhibited in the covariance matrix of the frequency domain inputs to the feedforward and feedback filters is also present in the inverse of this covariance matrix. This implies that the sparse structure influences the solution to the least squares adaptation problem. Second, Recursive Least Squares (RLS) algorithms which exploit this structure were derived and analyzed. The first stage of this work was presented at the ICASSP 2014 conference by Atulya Yellepeddi and myself.

IMPACT/APPLICATIONS

All of the above results are directed towards improving the performance of underwater acoustic communications systems. In particular, these results will allow future systems to operate reliably at lower signal to noise ratios and achieve a higher level of reliability and data transmission rate in challenging environments than is possible with current systems. The RMT work and work on surface scattering will facilitate a theory based approach to optimal spacing and partitioning of arrays and the two stage processing of subarrays of sensors and the combining of their outputs. The work on sparse structure covariance matrices in adaptive equalization will improve the performance of adaptive equalizers in challenging environments such as those with low SNRs or rapid environmental fluctuations.

RELATED PROJECTS

The work under these grants and contracts is closely related to the ONR OA SRA project titled ”Optimal scheduling for underwater communications in multiple-user scenarios”, ONR Grant Number N00014-13-10142.

PUBLICATIONS

A. Yellepeddi, J. Preisig, “Representing the Structure of Underwater Acoustic Communication Data Using Probabilistic Graphical Models”, at *168th Meeting of the Acoustical Society of America*. Indianapolis, IN, Oct. 27 - 31, 2014

B. Tomasi, J. Preisig, ”Energy heuristic scheduling with partial queue and channel state information”, at 2014 ACM Workshop on Underwater Wireless Networks (WUWNet’14), Rome, Italy, Nov. 12 - 14, 2014.

J. Preisig, “Underwater Acoustic Communications: Enabling the Next Generation at the Intersection of Ocean Acoustics and Signal Processing”, at the ONR Ocean Acoustics Program Peer Review, April 7-8, 2015.

B. Tomasi, J. Preisig, "Energy efficient transmission strategies for delay constrained traffic with limited feedback", *IEEE Transactions on Wireless Communications*, Vol. 14, No. 3, March 2015, pp. 1369-1379.

B. Tomasi, D. Munaretto, J. Preisig, M. Zorzi, "Redundancy allocation in time-varying channels with long propagation delays", *Elsevier Journal on Ad-hoc Networks*, In Press.

A. Yellepeddi, J. Preisig, "Adaptive Equalization in a Turbo Loop", to appear in *IEEE Trans. on Wireless Communication*.

M. Pajovic, J. Preisig, "Performance Analytics and Optimal Design of Multichannel Equalizers for Underwater Acoustic Communications", to appear in *IEEE Journal of Oceanic Engineering*.