Acoustic Communications 2011 Experiment: Deployment Support and Post Experiment Data Handling and Analysis

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Award Number: N00014-111-0426

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 work under this proposal are to support the Long Term Goals in the context of underwater **acoustic** communications systems. 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 not the data and make it available to KAM11 participants.

2. Develop and validate models of the impact of sea surface and upper ocean boundary layer processes on the performance of underwater acoustic communications systems.
3. Develop methods of detecting and synchronizing to known underwater acoustic communications signals at very low signal to noise ratios.
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.

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. The specifics are described below with paragraph numbers corresponding to the paragraph numbers in the preceding section (OBJECTIVES).

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.

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.

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.

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. Finally, the use of such equalizer in turbo-equalization algorithms posses great promise for future performance gains. Yet, in the underwater acoustic environment where the channel can change significantly over the duration of a communications packet. Thus, the equalizer must be adapted in each iteration of the turbo algorithm during which it has imperfect knowledge of the transmitted data symbols which are needed to guide the adaptation algorithm.

   The approach to addressing the ”large number of parameters” problem will be to develop beamspace processing methods and adaptive subarray processing methods for transforming the single high dimensional adaptation problem into one or more lower dimensional problems. Supporting results in the field of *Random Matrix Theory* will be 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 framework will serve as a starting point for this development.

WORK COMPLETED

This year the completed work has been in three areas. The first are concerns the use of Random Matrix theory to characterize the performance of adaptive processing algorithms which use sample correlation matrixes in the adaptation of algorithm parameters. We completed work includes characterizing the performance of and MVDR type filter utilizing a diagonally loaded sample correlation matrix and an initial assessment of the impact of the setup (numbers of filter taps) of adaptive decision feedback equalizers on their performance in both simulated and real world data. The second area in which work as been completed concerns the use of soft information in the adaptation of decision feedback equalizers. The work both characterizes performance and evaluates the optimization of equalizer filter length and type. Finally, work as been completed on the evaluation of KAM11 data to related the vertical wave number spectrum of the ambient noise field to the surface wind sped and surface conditions.

RESULTS

The work on random matrix theory has let to the interesting result that the optimal diagonal loading of a sample covariance matrix depends on the ”pointing” direction of the resulting array processing algorithm and the distance in angle space from the pointing direction to the true direction to a source or interferer. The adjustment in the diagonal loading factor smoothly transitions the adaptive MVDR algorithm from behaving like a spatial matched filter when the diagonal loading factor is high to a fully adaptive algorithm when the diagonal loading factor is low. The is result is presented in the Asilomar paper by Pajovic and myself. The RMT work has also led to preliminary results on optimal equalizer configuration. Somewhat surprisingly, it appears that the most effective feedfoward and feedback filters are very small in duration and that the equalizer gets its most effective processing gain when it has a large number of sensors but a small number in 2013. This work is presented in the Globecom paper by Pajovic and myself.

The work on the use of soft information in equalizer adaptation yielded significant results in several areas. It first demonstrated the performance improvements possible with the developed RELS technique when incorporated into a turbo equalizer structure. This work is presented in the Asilomar paper by Yellepedi and myself and the submitted paper to IEEE Transactions on Wireless Communications. It also derived several probabilistic models for the input to the equalizer decision device and determined that the performance loss of a simplified Gaussian model is small when compared to a more accurate Gaussian mixture model and that the computational complexity reduction with the simplified Gaussian model more than justifies its use. This work is presented in the ICASSP paper. Finally, the work theoretically evaluated a ”catastrophic failure” mode of adaptive decision feedback equalizers that has been reported in the literature but whose cause has never been studied. The work shows the conditions that leads to the failure mode of the equalizes and is reported in the submitted paper to IEEE Transactions on Wireless Communications.

Finally, the work on the vertical wavenumber spectrum has related optimal element spacing in the linear vertical array of a multichannel equalizer to the vertical wavenumber spectrum. It also showed a dependence of this spectrum on wind speed and identified some conditions under when there may be a ”noise notch” present in the spectrum. A summary of this work was presented in an invited talk to the Jasons in July 2013.

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 RELS based 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 this grant is closely related to the MURI funded project titled, ”Underwater Acoustic Propagation and Communications: A Coupled Research Program”, ONR Grant Number N00014-07-10738. Funds from both grants were used to pay for the KAM11 experiment and the post-experiment processing and analysis of the data as well as the development and analysis of channel equalization algorithms.

PUBLICATIONS

M. Pajovic, J. Preisig, “Performance Analysis of RLS based Channel Tracking using Random Matrix Methods”, submitted to *IEEE Trans. on Signal Processing*.

A. Yellepeddi, J. Preisig, “Soft Decision Directed Adaptation: Performance and Turbo Equalization”, submitted to *IEEE Trans. on Wireless Communication*.

G. Deane, J. Preisig, A. Lavery, “The Suspension of Large Bubbles near the Sea Surface by Turbulence and their Role in Absorbing Forward-Scattered Sound”, *IEEE Journal of Oceanic Engineering*, to appear.

M. Pajovic, J. Preisig, A. Baggeroer, “MSE of diagonally loaded Capon beamformer-based power spectrum estimator in snapshot deficient regime”, at *46th Asilomar Conf. on Signals, Systems and Computers*, Nov. 4 - 7, 2012, pp. 207 - 211.

A. Yellepeddi, J. Preisig, “Soft-adaptive turbo equalization: Using soft information in adaptation”, at *46th Asilomar Conf. on Signals, Systems and Computers*, Nov. 4 - 7, 2012, pp. 1541 - 1546.

A. Yellepeddi, J. Preisig, “Design of Decision Device for the Adaptation of Decision Directed Equalizers”, to appear in *Proc. IEEE Int. Conf. on Acoustics, Speech, and Signal Processing*. Vancouver, Canada, May 26 - 31, 2013.

M. Pajovic, J. Preisig, “Performance Analysis of Least Squares-based Multichannel Decision Feedback Equalization of Time-Varying Channels”, at *46th IEEE Global Communications Conference*, Dec. 9 - 13, 2013.