

Section 2 Digital Signal Processing

Chapter 1 Digital Signal Processing Research Program

Chapter 2 Advanced Telecommunications and Signal Processing Program

Chapter 3 Combined Source and Channel Coding for High-Definition Television

Chapter 1. Digital Signal Processing Research Program

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1.1 Introduction

The field of digital signal processing grew out of the flexibility afforded by the use of digital computers in implementing signal processing algorithms and systems. It has since broadened into the use of a variety of both digital and analog technologies, spanning a broad range of applications, bandwidths, and realizations. The Digital Signal Processing Group carries out research on algorithms for signal processing and their applications. Current application areas of interest include signal enhancement, active noise cancellation, processing of speech, music and underwater acoustic signals, advanced beamforming for radar and sonar systems, and signal coding and transmission.

In some of our recent work, we have developed new methods for signal enhancement and noise cancellation with single or multisensor measurements. We have also been developing new methods for representing and analyzing fractal signals. This class of signals arises in a wide variety of physical environments and also has potential in problems involving signal design. We are also exploring potential uses of nonlinear dynamics and chaos theory for signal design and analysis.

In other research, we are investigating applications of signal and array processing to ocean and structural acoustics and geophysics. These problems require the combination of digital signal processing tools with a knowledge of wave propagation to develop systems for short time spectral analysis, wavenumber spectrum estimation, source localization, and matched field processing. We emphasize the use of real-world data from laboratory and field experiments such as the Heard Island Experiment for Acoustic Monitoring of Global Warming and several Arctic acoustic experiments conducted on the polar ice cap.

The group is also involved in research on new signal processing techniques for multiuser wireless communication systems. The intended applications range from mobile radio networks to indoor personal wireless systems to digital audio and television broadcast systems. Of special interest are broadband spread spectrum systems and code-division multiple access networks both for conventional and secure transmission.

Much of our work involves close collaboration with the Woods Hole Oceanographic Institution, MIT Lincoln Laboratory, and a number of high technology companies in the Boston Area.

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1.2 Single Mode Excitation in the Shallow Water Acoustic Channel

Sponsor

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Joint Graduate Program in Oceanographic
Engineering

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Under appropriate conditions, the ocean acts as an acoustic waveguide with sound propagation occurring almost entirely through a discrete set of normal modes. Shallow water is one such condition. The goal of this research is to use a vertical array to excite the underwater channel so that, as nearly as possible, only one of these discrete modes is propagating into the far field. Knowing that the propagating field consisted originally of only a single mode, tomographic techniques can characterize and locate ocean features by observing the distribution of modes received downrange.

The problem of exciting only a single mode is easily solved analytically for the time-invariant case with a uniform sound speed profile and bottom topography. However, it is much more challenging in a realistic ocean environment due to propagation losses and coupling between modes caused by time-varying inhomogeneities in the medium and range-dependent irregularities in the seafloor and water column. Our strategy is to use a reference array of receivers at the start of the far field to obtain information about the coupling in the channel, and then adaptively update the source array weights to obtain a single mode at this reference array.

Currently, we are focusing our attention on developing and investigating strategies for estimating the coupling between the sources and the reference array, and tracking the variations in this coupling over time due to a variety of ocean processes. We are using detailed ocean acoustic modeling programs such as SAFARI to simulate a realistic ocean environment for the work.

1.3 Self-Synchronization of Chaotic Systems: Analysis, Synthesis, and Applications

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Project Staff

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Chaotic systems provide a rich mechanism for signal design and generation, with potential applications to communications and signal processing. Because chaotic signals are typically broadband, noise-like, and difficult to predict, they can be used in various contexts for masking information-bearing waveforms and as modulating waveforms in spread spectrum systems. A particularly useful class of chaotic systems are those which possess a self-synchronization property. Specifically, two identical chaotic systems may synchronize when the second system is driven by the first.⁴ This property enables chaotic systems to be exploited for private communications.⁵ A potential difficulty, however, is that the analysis and synthesis of chaotic systems is not well understood because of the highly nonlinear nature of these systems.

In this research, we are developing new methods for analyzing and synthesizing a class of high-dimensional dissipative chaotic systems which possess the self-synchronization property. For the class of chaotic systems that we consider, the synchronization is global and highly robust to perturbations in the drive signal. Synchronization error models are derived which provide a deeper understanding of the inherent robustness of these systems. We also develop several methods for embedding an information-bearing waveform in a chaotic carrier signal and for recovering the information at the synchronizing receiver. In addition, the practical aspects of synchronized chaotic

⁴ L.M. Pecora and T.L. Carroll, "Synchronization in Chaotic Systems," *Phys. Rev. Lett.* 64(8): 821-824 (1990); T.L. Carroll and L.M. Pecora, "Synchronizing Chaotic Circuits," *IEEE Trans. Circuits Syst.* 38(4): 453-456 (1991).

⁵ K.M. Cuomo, A.V. Oppenheim, and S.H. Isabelle, *Spread Spectrum Modulation and Signal Masking Using Synchronized Chaotic Systems*, RLE TR-570 (Cambridge: MIT Research Laboratory of Electronics, 1992); A.V. Oppenheim, G.W. Wornell, S.H. Isabelle, and K.M. Cuomo, "Signal Processing in the Context of Chaotic Signals," *Proc. IEEE ICASSP 4*: 117-120 (1992).

systems are demonstrated using a transmitter and receiver circuit with dynamics governed by the chaotic Lorenz system.⁶

1.4 Algebraic and Probabilistic Structure in Fault-Tolerant Computation

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Project Staff

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The traditional approach towards fault-tolerant computation has been modular redundancy. Although universal and simple, modular redundancy is inherently expensive and inefficient in its use of resources. Recently developed Algorithm Based Fault Tolerance (ABFT) techniques offer more efficient fault coverage, but their design is specific to each application. A particular class of ABFT techniques involves the design of arithmetic codes that protect elementary computations. For the case of computations that can be represented as operations in a group, the recent doctoral thesis by Beckmann⁷ has shown how to obtain a variety of useful results and *systematic* constructive procedures.

We are exploring the extension of this work to other algebraic structures. Our current studies indicate that much of Beckmann's framework can be generalized to the case of semigroups. In the other direction, we are examining refinements for the case of computations occurring in rings, modules, fields, or vector spaces. Beckmann's thesis shows how to take advantage of the underlying group structure in all these instances, but we expect that the additional structure that they possess can lead to stronger results and more efficient implementations of fault-tolerant schemes. Another objective of our research is to fold probabilistic models for failures and errors in to the design and analysis of arithmetic codes. This will allow us to better characterize and design fault-tolerant systems.

1.5 Signal Processing Applications of Chaotic Dynamical Systems

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Project Staff

Steven H. Isabelle, Professor Alan V. Oppenheim, Professor Gregory W. Wornell

Researchers in areas ranging from animal behavior and medicine to economics and geophysics have found evidence of chaotic behavior in an enormous number of empirically gathered time series. Indeed, the sheer volume of apparently random phenomena which appear to have a deterministic explanation underscores the need for signal processing techniques specifically tailored to the unique characteristics of chaotic signals. In particular, because chaotic signals can generally be observed only indirectly, e.g., through some propagation channel or nonideal laboratory instrumentation, a signal's chaotic structure may be partially obscured by additive noise and convolutional distortion. Consequently, algorithms for reducing these distortions are an important component of signal processing systems for chaotic signals. This research explores effects of convolutional distortion on chaotic signals along with techniques for reducing such distortions.

In general, the limiting trajectory of a chaotic system will be a highly structured set in the state space, while the scalar output will appear erratic and unstructured. It is this "hidden" structure that makes the signal interesting and allows for a simple description. One measure of structure which has been used to characterize a chaotic signal is the fractal dimension of its strange attractor. We are

⁶ K.M. Cuomo and A.V. Oppenheim, *Synchronized Chaotic Circuits and Systems for Communications*, RLE TR-575 (Cambridge: MIT Research Laboratory of Electronics, 1992); K.M. Cuomo and A.V. Oppenheim, "Chaotic Signals and Systems for Communications," *Proc. IEEE ICASSP 3*: 137-140 (1993); K.M. Cuomo and A.V. Oppenheim, "Circuit Implementation of Synchronized Chaos with Applications to Communications," *Phys. Rev. Lett.* 71: 65-68 (1993).

⁷ P.E. Beckmann, *Fault-Tolerant Computation using Algebraic Homomorphisms*, RLE TR-580 (Cambridge: MIT Research Laboratory of Electronics, 1993).

examining the effect of convolution on fractal dimension and using these results to develop deconvolution algorithms. The major challenge here is developing optimal computationally efficient techniques which are uniformly applicable to a broad class of chaotic signals.

1.6 Wavelet-Based Representation and Algorithms for Generalized Fractal Signals

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Project Staff

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While the $1/f$ family of fractal processes have become increasingly appealing for modeling statistically scale-invariant phenomena, we study a generalization of this signal model to account for more general scaling behavior found in a wide variety of natural phenomena. While many phenomena exhibit scaling behavior, a great number of them do so only over a finite range of scales. For example, while seafloor morphology is observed to be self-similar at fine scales, such scaling behavior is absent in long length scales due to the lack of correlation among points far apart.⁸ On the other hand, many phenomena exhibit scaling behavior which varies over scales. For instance, such varying scaling behavior is encountered in the study of diluted gels and colloidal aggregates in the field of materials science.⁹

In this work, we focus on the development of a class of generalized fractal processes for capturing such nonuniform scaling behavior. Exploiting the role of the wavelet transformation as a whitening filter for such processes, we formulate algorithms

for addressing a number of practical estimation problems involving such signals. Adopting a maximum-likelihood criterion and invoking an estimate-maximize algorithm, we derive consistent, computationally-efficient spectral parameter estimators which are useful for the classification of generalized fractals. We also formulate a Bayesian minimum mean-squares error signal estimation scheme which is directly applicable in a variety of signal separation and signal recovery scenarios.

1.7 Approximate Signal Processing

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Jeffrey T. Ludwig, Professor S. Hamid Nawab

In any given problem-solving domain, an algorithm, A , may be considered an approximation to another algorithm, B , provided A is more computationally efficient than B by virtue of A being designed to yield a lower quality answer than B . Such approximate algorithms are said to carry out *approximate processing* with respect to the problem-solving domain under consideration. The need for such algorithms arises in applications with real-time constraints.

Within the context of real-time systems, a signal processing task may have to be performed within a time interval whose duration may or may not be determined prior to run time. In the case of a predetermined time allocation, an approximate processing algorithm may be used to obtain the highest quality answer possible within the allotted time. In cases where the time allocation is not predetermined, it is desirable to use an approximate processing algorithm which produces an answer of improving quality as a function of time. This allows the algorithm to be terminated whenever desired and the quality of the resulting answer is directly proportional to the actual execution time. These types of approximate processing algorithms are said to carry out *incremental refinement* of their answers.

In previous research it has been demonstrated that approximate processing algorithms can be effective

⁸ J.A. Goff and T.H. Jordan, "Stochastic Modeling of Seafloor Morphology: Inversion of Sea Beam Data for Second-Order Statistics," *J. of Geophys. Res.* 93(B11): 13589-13608 (1988).

⁹ *Fractals in the Natural Sciences*, eds. M. Fleischmann, D.J. Tildesley, and R.C. Ball (Princeton, New Jersey: Princeton University Press, 1989), pp. 35-53.

in the real-time analysis of music, speech, and synthetic signals. For example, approximations to the short-time Fourier transform (STFT) have been shown to retain prominent time-frequency characteristics of such signals while improving computational efficiency by an order of magnitude over FFT-based calculations of the exact STFT.

Our aim is to develop a formal structure for using approximate processing concepts in designing novel signal processing algorithms in areas such as time-frequency analysis, cepstral analysis, multi-pass analysis, adaptive filtering, and adaptive beamforming.

1.8 CDMA for Digital Storage

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James M. Ooi, Professor Gregory W. Wornell

Code division multiple access (CDMA) is an active area of communications research. It is a method by which multiple users superpose their transmissions on top of one another throughout an entire symbol period and throughout a larger frequency bandwidth than required. Although the technique was once primarily associated with military spread spectrum systems, designers today are finding that it has properties which are desirable outside of military use. These properties include automatic interference reduction when other users are not transmitting, ease of adding new users, graceful degradation as channel capacity is exceeded, and robustness against intentional jamming and multipath interference.

The goal of this research is to extend the CDMA concept to computer storage devices in order to reap some of the above advantages. The analogous behavior on which the work will focus is graceful degradation as capacity is exceeded, tolerance to faulty memory, ease of adding new users, and an automatic improvement in fidelity as the storage requirements decrease. It is also hoped that this may also lead to a more general theory of distributed coding for the storage channel.

1.9 Estimation and Detection of a Class of Chaotic Signals

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Chaos arises in a broad range of physical and man-made systems. Naturally, there is a need for efficient and robust algorithms for estimation and detection of chaotic signals in the presence of noise. We have obtained Maximum Likelihood algorithms for signal filtering, smoothing, and prediction for a class of chaotic signals, which although nonlinear, possess very convenient recursive implementations that are efficient both in terms of computation and storage. Generalized estimation algorithms are suggested for a much broader class of chaotic signals.

The pseudorandom, broadband characteristics, and ease of generation of chaotic signals make them appealing candidates for signaling waveforms in secure and low probability of intercept communication systems. We explore the viability of spread spectrum communication schemes using chaotic signals, and consider generalized antipodal coding schemes which possess attractive properties for secure communications. Optimal discrimination strategies for a class of these chaotic signaling schemes and robust, computationally efficient demodulators have been obtained. Monte Carlo simulations reveal that chaotic signaling is potentially useful for the particular communication applications mentioned above.

1.10 Real Time Active Noise Cancellation

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Brian M. Perreault, Professor Alan V. Oppenheim

In many industrial and consumer settings, there often exists undesired acoustic noise. In many cases, the noise is obtrusive enough to warrant an

effort to reduce its effect. In situations where the noise cannot be reduced using physical, passive methods, active noise cancellation (ANC) techniques are a suitable alternative. Many types of noise have certain predictable characteristics; active noise cancellation exploits these characteristics, and uses destructive interference from a secondary acoustic source to eradicate the noise.

One algorithm for active noise cancellation results from approaching the problem from a control perspective. If the noise is measured at a single sensor, the essence of the technique is to use a control algorithm to give feedback through an actuator, constraining the measurement at the sensor to zero.

In many applications, such as airplane cockpits, high levels of narrowband noise are present. This type of algorithm may be useful for the cancellation of these narrow bands of noise. This algorithm is being investigated to determine its effectiveness in such an environment. A set of headphones, utilizing a microphone and speaker in each ear cup, is being developed to cancel the cockpit noise. Such a set of headphones is very desirable for a pilot, especially considering that passive attenuation of the headphones is not enough to fully compensate for the noise.

Digital predictive techniques are being used in conjunction with analog to digital (A/D) and digital to analog (D/A) converters in order to implement the active noise cancellation techniques. A digital signal processor is employed to execute the algorithms, due to its high computational ability. A flexible set of hardware has been developed in order to evaluate different ANC algorithms in real time.

1.11 State and Parameter Estimation with Chaotic Systems

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Chaotic systems are nonlinear dynamical systems characterized by extreme sensitivity to initial conditions. A signal generated by a chaotic system may appear random, despite its having been generated by a low-order, deterministic dynamical system. Both random and chaotic signals lack long-term predictability; but, in contrast to truly random signals, chaotic signals exhibit short-term predictability. Evidence of chaotic behavior has been reported in many diverse disciplines including physics, biology, engineering, and economics.

We are exploring techniques for state and parameter estimation with chaotic systems. We have implemented the extended Kalman filter,¹⁰ a recursive state estimator for nonlinear systems, and several related algorithms¹¹ and have evaluated their effectiveness as state estimators for chaotic systems. Preliminary results have shown these algorithms to perform reasonably well. But, these results have also shown these algorithms to suffer potentially unacceptable deficiencies when applied to chaotic systems.

More recently, we have developed and begun testing several related novel, state-estimation techniques loosely motivated by maximum likelihood state estimation. The techniques exploit a distinguishing property of all chaotic systems—the simultaneous existence of stable and unstable manifolds.¹² The combination of these techniques with an estimate-maximize (EM) algorithm is also being considered. Finally, we plan to ascertain the value of various state-estimation techniques in improving the short-term predictability of chaotic signals.

¹⁰ A. Jazwinski, *Stochastic Processes and Filtering Theory* (New York: Academic Press, 1970); M.D. Richard, *State Estimation with Discrete-Time Chaotic Systems*, RLE TR-571 (Cambridge: MIT Research Laboratory of Electronics, 1992).

¹¹ A. Willsky, Course Notes for 6.433 "Recursive Estimation," Unpublished, 1989.

¹² J. Eckmann and D. Ruelle, "Ergodic Theory of Chaos and Strange Attractors," *Rev. of Mod. Phys.* 57(3) Part 1: 617-656 (1985).

1.12 Model Based Analysis of Music

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Stephen F. Scherock, Dr. Bernard Gold

Many digital audio signals can be broadly classified as speech or music. Speech signals have been studied extensively since the 1950s, for the purposes of automatic speech recognition, production, and data compression. Work has advanced toward each of these goals in part because of the development of models for speech production and recognition. While physically based models exist for music production, applying these models to synthetic music production or data compression has not been fully exploited. This research involves model-based compression of music from a single instrument, the trumpet.

1.13 Nonlinear Models for Signal Processing

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This research involves the use of nonlinear system models in signal processing. Predominantly, linear models and algorithms have been employed in this field due to the tractability of their analysis and the richness of the class of signals for which these models are well-suited. However, by trading complexity for tractability, we can further broaden the boundaries of signal processing. For example, we have recently shown that nonlinear signal modeling techniques are both useful and practical for modeling a variety of signals for which linear techniques have proven inadequate. Further, we have begun to find practical applications for nonlinear dynamic systems in chaos, which until recently were thought of as more paradoxical than practical. Another interesting phenomenon whose curious behavior

has crossed many disciplines of science, is the theory of solitons. It is our goal to exploit the behavior these solitary waves and other nonlinear phenomena in search of new paradigms and new directions in signal processing.

1.14 Environmental Robustness in Automatic Speech Recognition

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Shawn M. Verbout, Professor Alan V. Oppenheim

As the technology in automatic speech recognition (ASR) becomes increasingly advanced, greater consideration is given to speech as a viable means of interaction between humans and machines. However, as more ASR systems become incorporated into real-world applications, greater attention must be devoted to making these systems robust with respect to changes in their operating environments. The need for environmental robustness is clear when we consider the difficulties presently encountered in attempting to make ASR systems yield consistently high performance in practical settings. In particular, even the most sophisticated ASR systems today are extremely sensitive to variations in recording conditions, and must be carefully retrained for each new environment in which they are deployed.

In the context of ASR, environmental variability is a broad notion encompassing the various changes that can occur in any factor affecting acoustic measurements. For example, environmental changes can be due to physiological differences between different speakers, variations in loudness for a single speaker, differences in the physical relationship between the speaker and the microphone, changes in the ambient noise level, differences in recording equipment, and differences in room acoustics. In addition to these factors, which may be termed "physical," there are yet other environmental factors that may be viewed as "situational." For example, consider the operation of an ASR system in the relatively benign setting of a typical business office. In this environment, a speech signal can be corrupted during a recording session by unanticipated acoustic events such as sudden activation of a manifold system, intermittent keyboard clicking, background conversations, slamming doors and file cabinet drawers, and reverberation due to acoustic reflections from walls, floors, and various objects in the room.

The long-term objective of our research in this area is to determine the signal processing technology required to make the performance of an ASR system relatively insensitive to these kinds of changes in the recording environment, while keeping the performance at a consistently high level. One viable method for making an ASR system more robust is to replace the single sensor in a typical recording environment with multiple sensors located in various parts of the room. This multisensor approach affords some redundancy in the measurement of the primary speech signal, and simultaneously provides a diverse set of reference measurements for noise events throughout the room. The resulting multiple signal measurements may then be processed to extract the primary speech signal from the interfering noise sources.¹³ Important intermediate goals in our investigation of the multisensor robustness approach will include quantifying the performance of an existing ASR system with and without preprocessing of the sensor measurements using algorithms such as those mentioned above.

1.15 Active Noise Cancellation

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Project Staff

Kambiz C. Zangi, Professor Alan V. Oppenheim

Unwanted acoustic noise is a by-product of many industrial processes and systems. With active noise cancellation (ANC), one introduces secondary noise sources to generate an acoustic field that interferes destructively with the unwanted noise, thereby eliminating it. Examples of such unwanted noise include machinery noise, aircraft cabin noise, and fan noise.

Traditional active noise cancellation systems assume that the statistical characteristic of the primary noise is known *a priori*. Furthermore, almost all of the existing systems use two microphones, and as result suffer from an *acoustic feedback* between the canceling speaker and the input microphone.¹⁴

We have developed an adaptive active noise cancellation system which only uses one microphone, and therefore has no feedback problem. This system uses the estimate maximize (EM) algorithm to simultaneously estimate the noise statistics and the transfer function between the canceling speaker and the microphone. An optimal canceling signal is then generated based on these estimates.¹⁵

We have also developed a two microphone version of the above system which does not suffer from the feedback problem and is more intelligent in using the outputs of the microphones.

We are currently studying the problem of noise cancellation in a volume. A topic of fundamental interest is to find analytically simple ways to describe the sound field over a volume from measurements made at a finite set of points in that volume. Similarly, we would like to find ways to alter the sound field in a desired manner using only a finite number of sources.

1.16 Oceanographic Signal Processing

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Project Staff

Professor Arthur B. Baggeroer

Our research programs involve the application of signal and array processing to problems in ocean acoustics and geophysics. They require an understanding of both signal processing and wave propa-

¹³ K.M. Cuomo, A.V. Oppenheim, and S.H. Isabelle, *Spread Spectrum Modulation and Signal Masking Using Synchronized Chaotic Systems*, RLE TR-570 (Cambridge: MIT Research Laboratory of Electronics, 1992); E. Weinstein, M. Feder, and A.V. Oppenheim, *Multi-Channel Signal Separation Based on Decorrelation*, RLE TR-573 (Cambridge: MIT Research Laboratory of Electronics, 1992).

¹⁴ L.J. Eriksson, M.C. Allie, and C.D. Bremigan, "Active Noise Control Using Adaptive Digital Signal Processing," *Proc. ICASSP 4*: 2594-2597 (1988).

¹⁵ E. Weinstein, A. Oppenheim, and M. Feder, *Signal Enhancement Using Single and Multi-Sensor Measurements*, RLE TR-560 (Cambridge: MIT Research Laboratory of Electronics, 1990); K. Zangi, "A New Two Sensor Active Noise Cancellation Algorithm," *Proc. ICASSP 1*: (1992).

gation; moreover, most use data from laboratory or field experiments, so an appreciation of real world issues such as noise, sensor calibrations and modeling errors is needed. Several of the topics provide opportunities to participate in oceanographic cruises.

1.16.1 Acoustic Thermometry of Ocean Climate

The Heard Island Feasibility Test demonstrated that coded acoustic signal can be transmitted over 10,000 km ranges. This has led to the Acoustic Thermometry of Ocean Climate (ATOC) program by ARPA. A network of acoustic sources and receivers is being deployed in the Pacific ocean this spring to monitor the ocean by measuring changes in acoustic travel times. These changes will be used to infer temperature changes.

1.16.2 Directional Spectrum Estimation: a Maximum Likelihood Method

Project Staff

John P. Mann, Professor Arthur B. Baggeroer

Antenna arrays can be used to estimate the direction of arrival of signals incident upon the array. Capon's minimum variance method is commonly used to estimate the frequency spectrum arriving from a desired direction. However, this method has some limitations. A finite set of samples from the array is used to form a sample covariance, which must then be inverted to calculate the minimum variance estimate of the spectrum. In order for the sample covariance to provide reasonable results, it has been shown that the number of samples necessary is approximately twice the number of sensors in the array.¹⁶ In cases where the signal environment changes fast enough, this requirement becomes a problem. Estimates need to be updated faster than is possible given this constraint.

In this research, we will investigate methods of directional spectrum estimation when the number of samples available is limited. In addition to some commonly used algorithms, a maximum likelihood method for estimating frequency wavenumber spectra has been derived and will be evaluated and

refined.¹⁷ The performance of the different methods will be compared through simulations on a series of different array geometries and signal environments. The effect of errors in the gain and position of sensor elements will also be investigated.

1.16.3 Single Mode Excitation in the Shallow Water Acoustic Channel

Project Staff

Denis J. Peregrym, Professor Arthur B. Baggeroer

Acoustic signals can be described as a sum of normal modes or eigenfunctions of a waveguide. Under certain conditions in shallow water, a small set of discrete modes carries almost all of the acoustic energy. Our goal is to design a source which excites only a single mode in the waveguide at a reference receiver some distance down the guide from the source. The signal at this receiver is used to adaptively update the source weights and converge to a single mode. Our current research involves investigating the field shape in the waveguide and deciding on an optimum position for the reference array in the near field/far field boundary. In addition, we are investigating how the mode shapes or weights at the source array required to generate a single mode can be used to characterize the channel.

1.16.4 Estimation of Normal Mode Amplitudes for Underwater Acoustics

Project Staff

Kathleen E. Wage, Professor Arthur B. Baggeroer

In underwater acoustics, signals are often efficiently represented as weighted sums of the normal modes or eigenfunctions associated with the waveguide. The weights are usually estimated from data obtained with a vertical array of hydrophones. The purpose of this research is to investigate methods for estimating the modal amplitudes (i.e., weights). In addition to studying modal amplitude estimation techniques previously developed, this work also applies several bearing estimation algorithms to the modal amplitude problem. A least squares formulation is the most common in the literature, and

¹⁶ I.S. Reed, J.D. Mallett, and L.E. Brennan, "Rapid Convergence Rate in Adaptive Arrays," *IEEE Trans. Aerosp. Electron. Syst.* AES-10: 853-863 (1974).

¹⁷ A.B. Baggeroer, "A True Maximum Likelihood Method for Estimating Frequency Wavenumber Spectra," *Proc. Inst. Acoust.* 13(2): 217-224 (1991).

several variations of this generalized approach are under consideration. The three-bearing estimation algorithms which will be implemented were developed in the context of direction-of-arrival location for diversely-polarized antennas.¹⁸ This thesis considers the effects of noise, modeling assumptions, mode coherence, and multiple modal sources on the behavior of each estimator.

The motivation for this work is twofold. First, modal decomposition is a natural way to approach solutions to the acoustic wave equation, and a knowledge of the modal structure is often helpful in exploring the effects of a particular channel on a transmitted signal. Specifically, mode attenuation is

important in many acoustic and oceanographic processes, e.g., Kuperman and Ingenito have explored the attenuation of modes due to scattering at rough boundaries.¹⁹ A second motivating factor is that the modal amplitudes are useful in matched mode processing and in matched field tomography. Recent papers indicate that source range and depth information may be obtained by performing matched field beamforming using the modal coefficients.²⁰ Shang has also used adiabatic normal mode theory to develop a method for acoustic tomography.²¹ Obviously, the ability to estimate the mode coefficients accurately and efficiently is essential to the success of these types of algorithms.

¹⁸ E.R. Ferrara and T.M. Parks, "Direction Finding with an Array of Antennas Having Diverse Polarizations," *IEEE Trans. Antennas Propagat.* AP-31(2): 231-236 (1983).

¹⁹ W.A. Kuperman and F. Ingenito, "Attenuation of the Coherent Component of Sound Propagating in Shallow Water with Rough Boundaries," *J. Acoust. Soc. Am.* 61(5): 1178-1187 (1977).

²⁰ S.M. Jesus, "Normal-mode Matching Localization in Shallow Water: Environmental and System Effects," *J. Acoust. Soc. Am.* 90(4): 2034-2041 (1991); E.C. Shang, "An Efficient High-resolution Method of Source Localization Processing in Mode Space," *J. Acoust. Soc. Am.* 86(5): 1960-1967 (1989); T.C. Yang, "A Method of Range and Depth Estimation by Modal Decomposition," *J. Acoust. Soc. Am.* 82(2): 1736-1745 (1987).

²¹ E.C. Shang, "Ocean Acoustic Tomography Based on Adiabatic Mode Theory," *J. Acoust. Soc. Am.* 85(4): 1531-1537 (1989).