Section 2 Digital Signal Processing

Chapter 1. Digital Signal Processing Research Program

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1.1 Introduction

The field of digital signal processing developed because the use of digital computers afforded more flexibility in implementing signal processing algorithms and systems. This field now encompasses a variety of both digital and analog technologies, spanning a wide range of applications, bandwidths, and realizations. The Digital Signal Processing Group at RLE carries out research on algorithms and their applications in signal processing. Currently, we are interested in applying these algorithms to signal enhancement, processing of 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, which occurs in a wide variety of physical environments, has potential application in problems involving signal design. We are also exploring potential applications 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 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 DSP group is also involved in research on broadband and secure communication systems. These activities include the development of spread spectrum and multiple-access communication systems exploiting sophisticated signal processing algorithms, used in, for example, personal wireless systems.

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 Oceanographic Signal Processing

Sponsor

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Professor Arthur B. Baggeroer

Our research programs in this area involve the application of signal and array processing to problems in ocean acoustics and geophysics. These problems require an understanding of both signal processing and wave propagation; moreover, most rely on 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 the opportunity to participate in oceanographic cruises.

1.2.1 Acoustic Thermometry of Ocean Climate

The Heard Island Feasibility Test demonstrated that coded acoustic signals can be transmitted over 10,000 km ranges. This test led to the Acoustic Thermometry of Ocean Climate (ATOC) program, which is funded by the Defense Advanced Projects Agency (DARPA). A network of acoustic sources and receivers will be deployed in the Pacific, Atlantic, and Indian Oceans to monitor them by measuring changes in acoustic travel times. These changes will be used to infer temperature changes.

1.2.2 Signal Processing for Large Aperture Arrays

There is extensive literature on the subject of adaptive processing for arrays with small numbers of sensors. However, when an array has a large number of sensors, some problems can result. The main problem is that the statistical characterization of the ambient field requires an amount of data which exceeds the short-term stationarity of the field. Introducing adaption by adding sensors can lead to lower quality performance because additional degrees of freedom are not well characterized. Our research develops adaptive array processing algorithms which incorporate the statistical characterization of the ambient field with a large number of sensors.

1.2.3 Matched Field Processing

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Matched field processing is used in underwater acoustics for processing data from large arrays when the inhomogeneity of the signal field is important. It is particularly applicable to vertical arrays used for low frequency, long-range propagation. Two major applications of matched field methods are to determine: (1) the location (range, depth and bearing) of an acoustic source; and (2) the sound speed of the acoustic channel when the location of the source and receiver (tomography) are known. Matched field processing can improve the resolution performance of an array by a factor of 100 by exploiting coherence among the multipath, or modes, in acoustic propagation. Some of our important research topics include (1) developing robust methods which are tolerant of environmental uncertainties and (2) determining "optimum array geometries" and adaptive processing which incorporates the mesoscale dynamics of the ocean.

1.3 Segmentation of Polarimetric Synthetic Aperture Radar Images Using Markov Random Field Techniques

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Project Staff

Saurav Dev Bhatta, Dr. Jacques Verly, Professor Alan V. Oppenheim

Synthetic Aperture Radar (SAR) images constitute one major class of speckled images. Speckle obscures boundries, so the goal of segmentation is to recover obscured regions from the speckled images. Derin and Kelly[3] and their collegues at the University of Massachusetts at Amherst have developed and tested a technique, which uses Markov Random Field models to segment speckled images. We are developing a method for segmenting Polarimetric SAR (POL-SAR) images, which is based on

3 H. Derin, P.A. Kelly, G. Vezina, and S.G. Labitt, "Modelling and Segmentation of Speckled Images Using Complex Data," *IEEE Trans. Geosci. Remote Sens.* (28 (1): 76-87 (1990).

the technique developed at University of Massachusetts.

The image model used here consists of a two-level hierarchical process that makes use of the fact that speckle exhibits spatial correlation. The first level, called the Region Process, is characterized by a Markov Random Field (MRF) that groups the pixels of the image into regions. The second level, the Speckle Process, is characterized by another random field (not necessarily MRF) peculiar to the region under consideration. The two types of processes are independent of each other and between themselves. Only the Speckle Process is observable. The precise details of the Speckle Process depend upon the kind of imagery under consideration: amplitude (complex) versus intensity (real) and single-look versus multilook.

The algorithm is based on a Maximum a Posteriori (MAP) formulation to estimate segmentation (which is the underlying Region Process), given the observation data (which is directly related to the Speckle Process). In particular, the Simulated Annealing (SA) procedure is used to obtain the MAP estimate of the regions of the image. This algorithm proceeds in a series of iterations, each consisting of an estimation of the necessary region and speckle parameters followed by segmentation based on these estimated parameters. The algorithm is adaptive, estimating necessary parameters automatically, when given the number of regions in the image.

1.4 Signal Synthesis from Marine Mammal Communications

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Project Staff

John R. Buck, Professor Alan V. Oppenheim, Dr. Josko Catipovic

Marine mammals such as whales and dolphins utilize a wide variety of sounds for underwater acoustic communication while engaging in a variety of complex social behaviors. It is appealing to speculate that millenia of evolution have brought these animals to the point where their signals are in some sense optimum for the underwater environment. By studying their signals, we hope to discover which features make the signals well-suited for the underwater acoustic environment and under which criteria, if any, the signals are optimal. The knowledge we gain by analyzing the biological signals should indicate an approach for synthesizing new optimal signals for underwater communications.

1.5 Analysis, Synthesis, and Applications of Synchronized Chaotic Systems

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Project Staff

Kevin M. Cuomo, Professor Alan V. Oppenheim

Chaotic systems provide a rich mechanism for signal design and generation for communications and a variety of signal processing applications. Because chaotic signals are typically broadband, noiselike, and difficult to predict, they can be utilized in various contexts for masking information-bearing waveforms and as modulating waveforms in spread spectrum systems. It is often desirable to utilize high-dimensional chaotic systems with specified characteristics. However, there is a practical difficulty because the analysis and synthesis of chaotic systems is not well understood due to their highly nonlinear nature. A significant challenge and opportunity exists for exploring and developing new methods for chaotic signal generation and detection.

One objective of this research is to develop theoretical and numerical techniques for synthesizing a class of high-dimensional chaotic systems. Another objective is to develop methods for embedding an information-bearing waveform in a chaotic carrier signal and then recovering the information at the receiver. We show that synchronized chaotic systems (SCSs) are well-suited to this problem because SCSs possess a self-synchronization property.[4] We also show that choatic signal masking allows analog signals to be privately transmitted and recovered at the intended receivers.[5] In Cuomo and Oppenheim,[6] we show that digital bit streams can be privately communicated by modulating a transmitter coefficient with the information waveform and detecting the modulation with an SCS at the receiver.

1.6 High-Resolution Direction Finding for Multidimensional Scenarios

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Recently, there has been considerable interest in applying high-resolution techniques for direction finding (DF) and time series analysis. Results from the research of Lee and Wengrovitz,[7] have improved our understanding of high-resolution direction finding techniques in the following areas:

- Beamformer design for beamspace approaches;
- Analytical expressions for the threshold SNR at which algorithms can resolve closely-spaced sources;
- Cramer-Rao lower bounds on the variances of unbiased estimators of direction;
- Covariance matrix eigenstructure for closely-spaced sources.

The results are applicable to far-field planar scenarios in which the location of each source is specified by a single angular parameter.

Many practical applications of DF techniques are multidimensional in nature, requiring estimation of a vector of parameters. For example, two angular parameters are necessary in 3-D far-field problems (azimuth, elevation). Three parameters may be required in some 3-D emitter localization problems (range, azimuth, elevation). Extension of 1-D approaches to multi-D is not always direct, as several high-resolution techniques, including MinNorm and minimum dimension Beamspace algorithms, fail to uniquely locate sources for multidimensional scenarios. This research explores the multidimensional direction finding problem and extends recent 1-D results to multi-D scenarios.

1.7 Signal Processing Applications of Chaotic Dynamical Systems

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Steven H. Isabelle, Professor Alan V. Oppenheim

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 charateristics of chaotic signals. In particular, because chaotic signals can generally be observed

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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 No. 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-92,* March 1992.

6 K.M. Cuomo and A.V. Oppenheim, *Synchronized Chaotic Circuits and Systems for Communications,* RLE TR No. 575. (Cambridge: MIT Research Laboratory of Electronics, 1992).

7 H.B. Lee and M.S. Wengrovitz, "Resolution Threshold of Beamspace MUSIC for Two Closely Spaced Emitters," *IEEE Trans. ASSP,* 38(9): 1545-1559 (1990); H.B. Lee and M.S.Wengrovitz, "Beamformer Preprocessing for Enhanced Resolution by the MUSIC Algorithm," *IEEE Trans. ASSP,* forthcoming; H.B. Lee, "The Cramer-Rao Bound on Frequency Estimates of Signals Closely Spaced in Frequency," *IEEE Trans. ASSP,* forthcoming.

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 this distortion are an important component of signal processing systems for chaotic signals. This research explores the effect 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 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.

1.8 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. Many phenomena exhibit scaling behavior, but a great number of them do this 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 these kinds of processes, we formulate algorithms for addressing a number of practical estimation problems involving these 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.

1.9 Acoustic Tomography through Matched Field Processing

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Ocean Acoustic Tomography is a technique in which sound is transmitted through the sea and, based on the received signal, inferences about properties of the ocean are made. Traditional ocean acoustic tomography has relied on travel time measurements of the signal to infer properties such as sound speed profiles and currents.

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

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

In this work, we explore the viability of exploiting more than travel time information embedded in the received signal to make tomographic inferences. Using efficient signal processing algorithms, we will initially focus on frontal location (a front is a region where an ocean sound speed profile suddenly changes). Data acquired during the Barents Sea Experiment of 1992 will be used in this investigation.

1.10 Binary Data Transmission and Detection with Chaotic Signals

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Project Staff

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Chaotic waveforms provide a rich family of signals whose behavior and properties are currently being explored in various signal possessing contexts. Classical signal processing corresponds to processing signals that are either deterministic and predictable or stochastic. Chaotic signals provide a qualitatively different workspace. Signals generated from chaotic systems are not predictable even though they are completely deterministic. These signals are very sensitive to infinitesimal changes in initial conditions, so that long-term predictability is impossible. Chaotic waveforms provide excellent candidates for use in a variety of contexts in signal processing due to their resemblance to random signals and their ease of implementation.

In this project, we are considering a very common problem in classical signal processing: the transmission of a binary data stream via waveform representation. Chaotic signals are introduced in coding the binary stream. Chaotic waveforms may prove very powerful in this context due to their ease of implementation, noiselike appearance, and broadband spectra. The signals used in such communication schemes have distinct dynamical behavior and characteristics. Given a sequence of noisy observations, we may readily set up a binary hypothesis test in order to detect the original binary stream.

A previously implemented preliminary heuristic detection algorithm shows that signal detection with sufficiently high SNR may be accomplished if sufficiently long-sequence representations are used. More sophisticated algorithms are explored which take into account properties of the distinct dynamical behavior of the chaotic signals that are used to code the binary stream.

1.11 Real-Time Active Noise Cancellation

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Project Staff

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In many industrial and consumer settings, undesirable acoustic noise often exists. 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.

Many predictive methods used in active noise cancellation are computationally intensive; to employ them in a practical manner, real-time hybrid techniques must be developed. Intelligent tradeoffs must be made between an algorithm's computational efficiency and its predictional effectiveness.

ANC techniques would be highly effective in a plane or helicopter cockpit. Such a setting has very high noise levels, and the noise is, to a certain degree, predictable. A set of headphones, utilizing a microphone and speaker in each ear cup, is being developed to cancel the cockpit noise. This is desirable because passive attenuation provided by headphones without ANC does not 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 to implement active noise cancellation techniques. A digital signal processor, which has high computational ability, was employed to execute the algorithms. A flexible set of hardware has been developed to evaluate different ANC algorithms in real time.

1.12 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 algorithms,[11] 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 that these algorithms 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.[12] The techniques exploit a distinguishing property of all chaotic systems—the simultaneous existence of stable and unstable manifolds.[13] 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.

1.13 Nonlinear Models for Signal Processing

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Project Staff

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This research involves the use of nonlinear system models in signal processing. Linear models and algorithms have been predominantly employed in this field due to the tractability of their analysis and 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, recently we have shown that nonlinear signal modeling techniques are both useful and practical for modeling a variety of signals for which linear techniques have proven inadequate. Furthermore, we have begun to find practical applications for nonlinear dynamic systems in chaos. Until recently, these applications were considered to be 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 of these solitary waves and other nonlinear phenomena in search of new paradigms and new directions in signal processing.

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10 A. Jazwinski, *Stochastic Processes and Filtering Theory* (New York: Academic Press, 1970); M.D. Richard, *State Estimation with Discrete-Time Chaotic Systems using the Extended Kalman Filter,* unpublished technical report, September 1991.

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

12 M.D. Richard, *State Estimation with Discrete-Time Chaotic Systems using the Extended Kalman Filter,* unpublished technical report, September 1991.

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

1.14 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 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 for synthetic music production or data compression has not been fully exploited. My research involves model-based compression of music from a single instrument, the trumpet.

1.15 Active Noise Cancellation

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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 a priori knowledge of the transfer function from the cancelling speaker to the point where noise cancellation is desired. Furthermore, almost all existing systems use two microphones, and, as a result, suffer from acoustic feedback between the cancelling speaker and input microphone.[14]

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

We have also developed a two-microphone version of the system mentioned above which does not suffer from a feedback problem and is more intelligent than the existing algorithms in using the outputs of the two microphones.[16]

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 desirable manner using only a finite number of sources.

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[14] L.J. Eriksson, M.C. Allie, and C.D. Bremigan, "Active Noise Control Using Adaptive Digital Signal Processing," *Proc. ICASSP,* New York, 1988, pp. 2594-2597.

[15] E. Weinstein, A. Oppenheim, and M. Feder, *Signal Enhancement Using Single and Multi-Sensor Measurements,* RLE TR No. 560 (Cambridge: MIT Research Laboratory of Electronics, 1990); A. Oppenheim, E. Weinstein, K. Zangi, M. Feder, and D. Gauger, "Single Sensor Active Noise Cancellation Based on the EM Algorithm," submitted to *IEEE Trans. Signal Proc.*

[16] K. Zangi, "A New Two Sensor Active Noise Cancellation Algorithm," *Proc. ICASSP,* Minneapolis, Minnesota, 1993.