Section 2 Digital Signal Processing

Chapter 1. Digital Signal Processing Research Program

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1.1 Introduction

The Signal Processing Research Program is directed at the development of new algorithms and their applications in a variety of areas. In addition to specific projects being carried out on campus, there is close interaction with MIT Lincoln Laboratory and the Woods Hole Oceanographic Institution. The application areas that we are primarily involved with are speech, image, and underwater acoustic signal processing. In addition to algorithm development and applications, there are a number of projects directed at issues of algorithm implementation. Also affecting our research directions is the recognition that while, historically, signal processing has principally emphasized numerical techniques, it will increasingly exploit a combination of numerical and symbolic processing, a direction that we refer to as knowledge-based signal processing.

1.2 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 and structural acoustics and geophysics. They require an understanding of both signal processing and wave propagation; moreover, most employ 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. (So far, over a dozen students have been able to participate.)

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1.2.1 Arctic Acoustics

MIT and the Woods Hole Oceanographic Institution have participated in seven experiments on the polar ice cap in the high Arctic since 1977. These programs have used large arrays to conduct studies of acoustic propagation under the ice, elastic wave propagation in the ice, ambient noise, seismic reflection and refraction, and long-range reverberation. The signal processing involves short-time spectral analysis, directional, or frequency wavenumber, spectrum estimation, source localization, matched field processing, and geophysical inverse theory.

1.2.2 Heard Island Experiment for Acoustic Monitoring of Global Warming

Sponsor

Vertical Arrays for the Heard Island Experiment
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The Heard Island Experiment is a feasibility study for using acoustics to monitor global warming. The speed of sound in the ocean changes 4.6 m/s/degree C, so very small differences in temperature lead to significant changes in travel time when signals propagate over long ranges. A loud sound source transmitted signals from Heard Island in the southern Indian Ocean for nine days in early 1991. Receivers from nine countries aboard 14 ships were keyed to listen to the signals. We deployed two 1.4 km, 32-element vertical arrays off Bermuda and Monterey—17,000 km from the source—to listen to these signals. The objectives of this experiment are (1) to determine if the signals can be detected, and (2) to estimate the accuracy of the travel time of the modes.

1.2.3 Structural Acoustics

When sound impinges on a complicated object, it scatters into many different paths in the structure. These processes may involve reflection, mode conversion into elastic waves, and scattering. Analytic calculation of the structure response is usually intractable, so determining the energy partitioning among all the scattered signals requires both experiments and simulation using finite element and/or finite difference methods in combination with high-resolution array processing. This array processing must resolve the scattered signal components which are both transient and overlapped temporally and spatially; moreover, the signals are often measured in the nearfield so wavefront curvature and inhomogeneities become important. The research topic involves working with both experimental data and synthetics generated using finite differences to estimate both energy partitioning and scattered signals.

1.2.4 Signal Processing for Large Aperture Arrays

Adaptive processing for arrays with a small number of sensors has an extensive literature; however, when the number of sensors becomes large, there are a number of outstanding problems. The basic 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 can lead to a situation where more sensors lead to poorer performance because the additional degrees of freedom are not well characterized. The research concerns developing adaptive array processing algorithms where the statistical characterization of the ambient field with a large number of sensors is incorporated.

1.2.5 Matched Field Processing

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

1.3 Fault-Tolerant Algorithms and Architectures for Digital Signal Processing

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Dr. Bruce R. Musicus, Professor Alan V. Oppenheim, Paul E. Beckmann

In many digital signal processing applications, there is a high cost of failure, and continuous, error-free operation is needed. Traditionally, the problem of fault-tolerance has been solved through the use of modular redundancy. Several copies of the system operate in parallel, and their outputs are compared with voter circuitry. Modular redundancy is a very general technique and can be applied to any system. However, it does not take advantage of the details of a specific problem and thus requires substantial amounts of overhead (100% for single error detection, 200% for single error correction).

Recently, an alternative method of protecting signal processing operations called Algorithm-Based Fault-Tolerance (ABFT) has emerged.[3] It promises redundancy requirements and performance advantages similar to existing error-correcting codes. Operands are encoded using a high-level error-correcting code, and algorithms are modified to operate on the encoded data. The specific coding scheme chosen must commute with the operation being performed. ABFT thus combines the design of algorithms, architectures, and fault-tolerant systems, and results in more reliable, less costly systems.

Applications of ABFT have thus far all been linear systems, and the data encoding and fault detection/correction techniques can be described using standard linear error-correcting codes. Extension of ABFT to other operations has been limited by the absence of suitable coding schemes.

In this research, we have two major goals. First, to develop a unifying framework capable of explaining coding schemes used in existing ABFT systems. Second, to apply this framework to extend ABFT to a wider range of systems, both linear and nonlinear. Substantial progress has already been made in both areas. By using a group theoretical approach, we have shown that the encoding scheme must define an algebraic homomorphism between the original operands and their encoded form. For the important class of systematic-separate codes, this homomorphism reveals the form which the redundant information must take, enabling all applicable coding schemes to be determined. Future work will focus on describing signal processing algorithms as operations in algebraic groups.

1.4 Imaging Ice-cracks Using Diffraction Tomography

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Project Staff

Professor Alan V. Oppenheim, Dr. S.D. Rajan, Saurav Dev Bhatta

This research deals with the development of an effective method for locating and observing the propagation of cracks in sea-water ice using diffraction tomography.

Ordinary tomography refers to the cross-sectional reconstruction of an object using transmission data. For geophysical applications, the data generally consist of travel time measurements of acoustic or electromagnetic waves. The objects are considered to be embedded in a homogeneous medium which contains the wave source and the receivers. As we deal with objects whose scale lengths approach the wavelength of the transmitted wave, the reconstruction from ordinary tomography becomes increasingly poor. Similar problems result when the coverage of the viewing angles is limited. Diffraction tomography, which uses the scattered (or diffracted) field to reconstruct the objects, performs much better in such cases. It can also yield a higher resolution.

[3] J.A. Abraham, "Fault-tolerance Techniques for Highly Parallel Signal Processing Architectures," *SPIE Highly Parallel Signal Proc. Arch.* 614: 49-65 (1986).

The main problems with applying such tomographic techniques in sea-water ice arise from the fact that the medium (ice) is inhomogeneous and elastic. The first step in the research consists of evaluating how good a reconstruction we can get using the fluid approximation. The second step involves modifying existing reconstruction algorithms, which assume a homogeneous inelastic embedding medium, to take into account the inhomogeneity and elasticity of the ice. We will focus primarily on ART (algebraic reconstruction techniques) type algorithms.

1.5 Implementation and Evaluation of a Dual-Sensor Time-Adaptive EM Algorithm for Signal Enhancement

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Professor Alan V. Oppenheim, Dr. Ehud Weinstein, John R. Buck

This research describes the implementation and evaluation of an adaptive time-domain algorithm for signal enhancement from multiple-sensor observations. The algorithm is first derived as a noncausal time-domain algorithm, then converted into a causal, recursive form. A more computationally efficient gradient-based parameter estimation step is also presented. The results of several experiments using synthetic data are shown. These experiments first illustrate that the algorithm works on data meeting all the assumptions made by the algorithm, then provide a basis for comparing the performance of the algorithm against the performance of a noncausal frequency-domain algorithm solving the same problem. Finally, an evaluation is made of the performance of the simpler gradient-based parameter estimation step.

This work was completed in August 1991.

1.6 Quantitative Comparisons of Dolphin Signature Whistles

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Professor Alan V. Oppenheim, John R. Buck

Marine mammal biologists have been interested in the identification and comparison of whistles from various dolphins for some time. The Bottle-nose Dolphin, *Tursiops truncatus*, is of particular interest because there is strong evidence that the whistles of this species are individually specific, i.e., the shape of the plot of the fundamental frequency against time of the whistle is unique to a given individual. This is generally referred to as the signature whistle hypothesis. There are varying explanations for why this occurs, but almost all researchers agree that the majority of the whistles made by a particular dolphin will exhibit the time-frequency contour characteristic of that individual and that most of the whistles not matching the individual's own signature whistle are imitations of the signature whistles of other members of the dolphin's social group.

Unfortunately, biologists have almost exclusively had to rely on qualitative judgments of similarity based on looking at printouts of spectrograms. This requires considerable investment of time until a particular researcher accumulates a sufficient mental library of contours to recognize and compare whistles within a large population. Because of the subjective nature of the judgments, the degree of similarity between two whistles can also be disputed by different researchers.

This work uses techniques from speech processing to develop a quantitative measure of similarity between dolphin signature whistles. The algorithm first extracts the contour of the primary frequency from the whistle, then compares the contour against a library of known contours to find a best fit for the unknown whistle. Like human speech, dolphin whistles are subject to fluctuations in the registration and duration of the various segments. The algorithm for comparison uses the technique of dynamic time warping to obtain the best possible registration of contours, subject to continuity restrictions.

Preliminary results indicate that the algorithm performs as well or better than any other quantitative method of signature whistle comparison.

1.7 Applications of Synchronization in Chaotic Systems

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Project Staff

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This research is concerned with the analysis, synthesis, and application of synchronizing chaotic dynamical systems. A chaotic system is a nonlinear deterministic system whose behavior is erratic, irregular, and similiar in many respects to stochastic processes. Two well-known examples of systems which can behave chaotically are the Lorenz and the Rossler systems. The signals that evolve in these systems form a highly structured set in the system's state space; however, scalar observations often appear unstructured and exhibit a broad Fourier spectrum. These characteristics of a nonlinear system would seem to defy synchronization; however, a chaotic system which can be decomposed into a drive system and a stable response subsystem will synchronize by coupling the drive and response systems with a common drive signal.[4]

The ability to synchronize remote systems through the use of a common drive signal suggests several novel signal processing and communication applications. We are currently exploring the use of chaotic signals and systems for spread spectrum communication and signal masking purposes.[5] We plan to develop a theoretical and computational framework to analyze and synthesize synchronizing chaotic systems that have specific signal properties and to obtain performance bounds of these new systems.

1.8 Signal Processing Applications of Chaotic Dynamical Systems

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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 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 the 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 examining the effect of convolution on fractal dimension and using these results to develop deconvolution algorithms. The major challenge here is developing optimal computa-

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

tionally efficient techniques which are uniformly applicable to a broad class of chaotic signals.

1.9 High-Resolution Direction Finding for Multidimensional Scenarios

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Project Staff

Professor Alan V. Oppenheim, Jacek Jachner

Recently, there has been considerable interest in High-Resolution Techniques for direction finding and time series analysis. Recent results[6] 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, and
- 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 (e.g., azimuth, elevation). Three parameters may be required in some 3-D emitter localization problems (e.g., 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.

1.10 Wavelet-Based Representation and Algorithms for Generalized Fractal Signals

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Professor Gregory W. Wornell, Warren M. Lam

Numerous signals and phenomena in nature can be modeled as generalized fractals. Indeed, it can be readily shown that generalized fractals constitute a much richer class of signal models that includes the familiar ARMA signals.

Despite the wide extent to which generalized fractals can be used in modeling, no satisfactory algorithms have been developed for the analysis of such signals. In the classical ARMA framework, the power spectral density (psd) of a signal is typically modeled as a sum of one-pole terms. Consequently, the psd is approximated as having sections that decay as integral powers of frequency. The psd of a generalized fractal process, on the other hand, consists of sections that decay as fractional powers of frequency. Standard methods for analyzing ARMA processes, therefore, are inadequate for the solving of problems involving generalized fractal processes.

Recently, the use of the discrete wavelet transform has made practical the analysis of 1/f or fractal processes.[7] We have been successful in applying similar methods to the analysis of a broader class of fractal processes whose psd is white at low frequencies and fractional at high frequencies. Parameter estimation problems involving such first order fractal processes have been formulated and solved. By decomposing a generalized fractal

[6] H.B. Lee and M.S. Wengrovitz, "Resolution Threshold of Beamspace MUSIC for Two Closely Spaced Emitters," *IEEE Trans. ASSP*, 38: 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; H.B. Lee, "Eigenvalues and Eigenvectors of Covariance Matrices for Signals Closely Spaced in Frequency," submitted to *IEEE Trans. ASSP*.

[7] G.W. Wornell, "A Karhunen-Loeve-like Expansion for 1/f Processes via Wavelets," *IEEE Trans. Info. Theory* IT-36: 859-861 (1990).

signal into a sum of one-pole fractal terms and applying the discrete wavelet transform, we investigate the possibility of an efficient algorithm for analyzing generalized fractals.

1.11 Signal Processing for Ocean Acoustic Tomography

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Professor Alan V. Oppenheim, Dr. John L. Spiesberger, James M. Njeru

An average sound speed profile over a 1000 km section of the northeast Pacific Ocean is obtained using Ocean Acoustic Tomography from data acquired during the 1987 SVLA experiment on a long 900 m 120-hydrophone vertical acoustic array.

In particular, we pulse compress the received signal with a phase-only matched filter. The signal, centered at 80 Hz, is phase-modulated by a maximal length sequence. A fast m-sequence cross-correlation algorithm based on the Hadamard transform is used. In addition, wideband Doppler correction and coherent averaging of repetitions of the signal are performed.

The tomographic inversion is initialized from a range averaged climatological profile. Multipaths are identified from ray theory. The identified arrivals are inverted for a range-independent sound speed profile change estimate. Estimates of source and array position error are also obtained. For the limited data set used, the sound speed change estimate is found to be insignificant and a significant instrument position estimate is obtained.

This work, which is being done under the MIT-Woods Hole Oceanographic Institution Joint Program, is co-supervised by Dr. John L. Spiesberger of the Woods Hole Oceanographic Institution, Woods Hole, Massachusetts.

1.12 Chaotic Signaling in Binary Data Transmission and Detection

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Chaotic waveforms provide a rich family of signals whose behavior and properties are currently explored in various signal processing 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. They are very sensitive to infinitesimal changes in initial conditions so that longterm 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, noise-like appearance, and broadband spectra. The signals used in such communication schemes have their own 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 preliminary heuristic detection algorithm that has already been implemented 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.

1.13 Adaptive Matched Field Processing in an Uncertain Propagation Environment

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Professor Alan V. Oppenheim, James C. Preisig

Adaptive array processing algorithms have achieved widespread use because they are very effective at rejecting unwanted signals (i.e., controlling sidelobe levels) and in general have very good resolution (i.e., have narrow mainlobes). However, many adaptive high-resolution array processing algorithms suffer a significant degradation in performance in the presence of environmental mismatch. This sensitivity to environmental mismatch is of particular concern in problems such as long-range acoustic array processing in the ocean where the array processor's knowledge of the propagation characteristics of the ocean is imperfect. An Adaptive Minmax Matched Field Processor has been developed which combines adaptive matched field processing and minmax approximation techniques to achieve the effective interference rejection characteristic of adaptive processors while limiting the sensitivity of the processor to environmental mismatch.

The derivation of the algorithm is carried out within the framework of minmax signal processing. The optimal array weights are those which minimize the maximum conditional mean squared estimation error at the output of a linear weight-and-sum beamformer. The error is conditioned on the propagation characteristics of the environment and the maximum is evaluated over the range of environmental conditions in which the processor is expected to operate. The theorems developed using this framework characterize the solutions to the minmax array weight problem and relate the optimal minmax array weights to the solution to a particular type of Wiener filtering problem. This relationship makes possible the development of an efficient algorithm for calculating the optimal minmax array weights and the associated estimate of the signal power emitted by a source at the array focal point. An important feature of this algorithm is that it is guaranteed to converge to an exact solution for the array weights and estimated signal power in a finite number of iterations.

The Adaptive Minmax Matched Field Processor can also be interpreted as a two-stage Minimum Variance Distortionless Response (MVDR) Matched Field Processor. The first stage of this processor generates an estimate of the replica vector of the signal emitted by a source at the array focal point, and the second stage is a traditional MVDR Matched Field Processor implemented using the estimate of the signal replica vector.

Computer simulations using several environmental models and types of environmental uncertainty have shown that the resolution and interference rejection capability of the Adaptive Minmax Matched Field Processor is close to that of a traditional MVDR Matched Field Processor which has perfect knowledge of the characteristics of the propagation environment and far exceeds that of the Bartlett Matched Field Processor. In addition, the simulations show that the Adaptive Minmax Matched Field Processor is able to maintain its accuracy, resolution, and interference rejection capability when its knowledge of the environment is only approximate and, therefore, it is much less sensitive to environmental mismatch than is the traditional MVDR Matched Field Processor.

1.14 State and Parameter Estimation with Chaotic Systems

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Project Staff

Professor Alan V. Oppenheim, Michael D. Richard

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 longterm 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,[8] a recursive state estimator for nonlinear systems, and several related algorithms,[9] 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 which are loosely motivated by maximum likelihood state estimation.[10] The techniques exploit a distinguishing property of all chaotic systems—the simultaneous existence of stable and unstable manifolds.[11] 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.

1.15 Causal Filters with Negative Group Delay

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Professor Alan V. Oppenheim, Stephen F. Scherock (Independent Research)

Traditional prediction schemes often assume an input signal with a fixed (ARMA) model. The performance of such schemes degrades as the input signal deviates from the model. In particular, a fixed linear predictor performs poorly when the characteristics of the signal vary with time (as would the output of a frequency-hopping transmitter). This research investigates the design of filters which have a negative group delay characteristic over a desired frequency range. Since negative group delay corresponds to positive time advance, one might initially think that all filters of this type were noncausal, therefore unrealizable in real time. In fact, causal filters with negative group delay can be realized. Such filters could provide a new framework for predicting a signal that is only known to lie within a certain band.

Two filter design algorithms have been implemented, one for FIR filters and one for IIR filters. The FIR filter design algorithm is based on minimizing prediction error over a band and reduces to a Weiner predictor in extreme cases. The IIR design algorithm uses the complex cepstrum to specify group delay, then attempts to find a rational Z transform matching the desired cepstrum. The quality of the filters has been characterized with two new error measures.

Further work may include the development of new design algorithms and theoretical bounds on performance.

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8 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.

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

10 M.D. Richard, "State Estimation with Discrete-Time Chaotic Systems Using the Extended Kalman Filter," unpublished technical report, September 1991.

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

1.16 Codebook Prediction: A Nonlinear Signal Modeling Paradigm

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A standard assumption made in signal processing is that an observed signal is the product of a linear, time-invariant system. Signals of this class have a rich history for which the mathematics is tractable, and many techniques have been explored. Based on this assumption, many linear methods of signal prediction and smoothing are used in applications such as speech and image coding and forecasting. However, many signals of interest arise from physical processes that are inherently nonlinear. Consequently, nonlinear dynamical system models may be much better suited to these phenomena.

Recently, the subject of nonlinear dynamics in general, and chaotic dynamics in particular, has attracted increased attention in the research literature. A number of new paradigms for signal modeling have emerged. While initial attention had focused on the study of the richness of behavior and properties of such systems, now there is considerable interest in the problems of modeling data based on this class of systems and addressing problems of signal processing for them. We are presently considering a number of these problems, based on the preliminary work of Farmer, Casdagli, Abarbanel, and others which address problems of prediction and smoothing based on a state space framework.

1.17 Synthesis, Analysis, and Processing of Fractal Signals

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Project Staff

Professor Gregory W. Wornell

Fractal geometry arises in a truly extraordinary range of natural and man-made phenomena. The 1/f family of fractal random processes, in particular, are appealing candidates for data modeling in a wide variety of signal processing scenarios involving such phenomena. In contrast to the well-studied family of ARMA processes, 1/f processes are typically characterized by persistent long-term correlation structure. However, the mathematical intractability of such processes has largely precluded their use in signal processing. We introduce and develop a powerful Karhunen-Loeve-like representation for 1/f processes in terms of orthonormal wavelet bases that considerably simplifies their analysis. Wavelet-based representations yield highly convenient synthesis and whitening filters for 1/f processes and allow a number of fundamental detection and estimation problems involving 1/f processes to be readily solved. In particular, we obtain robust and computationally efficient algorithms for parameter and signal estimation with 1/f signals in noisy backgrounds, coherent detection in 1/f backgrounds, and optimal discrimination between 1/f signals. Results from a variety of simulations are presented to demonstrate the viability of the algorithms.

In contrast to the statistically self-similar 1/f processes, homogeneous signals are governed by deterministic self-similarity. Orthonormal wavelet bases play an equally important role in the representation of these signals, and, in fact, are used to construct orthonormal "self-similar" bases. The spectral and fractal characteristics of homogeneous signals make them appealing candidates for use in a number of applications. As one potential example, we consider the use of homogeneous signal sets in a communications-based context. In particular, we develop a strategy for embedding information into a homogeneous waveform on all time-scales. The result is a unique multirate modulation strategy that is well-suited for use with noisy channels of simultaneously unknown duration and bandwidth. Computationally efficient modulators and demodulators are developed for the scheme, and the results of a preliminary performance evaluation are presented. Although not yet a fully developed protocol, "fractal modulation" represents a novel and compelling paradigm for communication.

1.18 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 that the statistical characteristic of the primary noise is known a priori. Furthermore, almost all of the existing systems use two microphones and, as a result, suffer from an acoustic feedback between the cancelling speaker and the input microphone.[12]

We have developed an adaptive active noise cancellation system which 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 cancelling speaker and the microphone. An optimal cancelling signal is then generated based on these estimates.[13]

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 finding 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.

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