

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 of signal design and analysis. Another research emphasis is on structuring

algorithms for approximate processing and successive refinement.

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.

A major application focus of the group is in the area of communication systems and networks. One current area of activity involves developing new signal processing techniques for multi-user 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.

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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.

1.2 Digital Audio Filter Design Using Frequency Transformations

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Audio filter design problems generally have a structure which is slightly different from that of conventional digital filter design problems. In audio filter specifications, frequency response is often specified on a logarithmic frequency scale. This is due to the fact that human auditory perception is more sensitive to low frequency resolution than to high frequency resolution. Similarly, the magnitude specification is usually specified in dB, which again stems from the way humans perceive differences in loudness. There is also a "just-noticeable" level of error. That is, at each frequency there is an allowable level of error within which the user cannot distinguish the output of the ideal from the approximated filter. One digital audio filter is equivalent to another if a listener cannot distinguish between the two different true frequency responses. In this thesis, by exploiting the special properties in audio filters and investigating the technique of frequency transformation, we seek to design a method which will yield filters with much shorter length than that resulting from conventional (for example, Parks-McClellan) design method. Specifically, we will study the all-pass frequency transformation, and its use to "warp" the frequency axis from a logarithmic scale to an approximately linear one. In taking the full benefit of "just-noticeable error," we will use McCallig's FIR filter design method, as opposed to the conventional one of Parks-McClellan. The benefit of the former is that it always guarantees that the output FIR filter is a filter of minimum length that meets the upper and lower "just-noticeable" error bound.

⁴ Bose Corporation, Framingham, Massachusetts.

1.3 Channel Equalization for Self-Synchronizing Chaotic Systems

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There is great potential for self-synchronizing chaotic systems in communications. Typical transmission channels introduce distortion that significantly degrades synchronization of transmitter and receiver. The types of distortions include time-varying attenuation due to fading, scattering, etc., and modification of the spectral characteristics due to channel filtering and multipath effects. It is our goal to correct these channel effects to restore synchronization. We propose the use of the properties of the transmitted chaotic signals and the synchronization property of the receiver to estimate and compensate for the channel distortions.

1.4 Techniques for Enhancing the Performance of Communication Systems Employing Spread-Response Precoding

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Soosan Beheshti, Professor Gregory W. Wornell

In wireless communication systems, various coding schemes are used to improve transmission reliability over a fading channel. A spread-response precoder, even in the absence of a coder, not only is effective in correcting errors, but also effectively transforms a Rayleigh fading channel into a simpler quasi-Gaussian one.

This research explores combining trellis coded modulation (TCM) with a spread-response precoding for transmission over a Rayleigh channel. We will assess the improvement yielded by the precoder in terms of system complexity, delay characteristics, and probability of error performance. We will also explore a multistage postcoder for use with spread-response precoding and assess the corresponding improved performance.

1.5 Active Noise Cancellation in Automobiles

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A fundamental goal of this research is to determine the feasibility of real-time active noise cancellation within the interior of an automobile. Active noise cancellation in a volume entails measuring the sound field at a finite number of microphone locations and using predictable characteristics of the acoustic noise to generate and transmit canceling signals from one or more speakers. The goal is to attenuate the noise at the sensors through destructive interference.

Using computer simulations on recorded road noise, we will assess the effectiveness of known algorithms, such as the estimate-maximize (EM), least-mean square (LMS), and optimal control approaches, in attenuating acoustic noise at a point within the car. An autoregressive-moving average (ARMA) model of the measured transfer function from car speaker to noise sensor represents the effect of car acoustics on the canceling signal.

1.6 Single Mode Excitation in the Shallow Water Acoustic Channel

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Project Staff

John R. Buck, Professor Alan V. Oppenheim, Dr. James C. Preisig

For shallow water acoustics, the ocean acts as a waveguide, with sound propagation in the far field occurring almost entirely through a discrete set of normal modes. The goal of this research is to use a vertical source array to excite the underwater channel so that only one of these discrete modes is propagating at a specified reference location. While open loop solutions are analytically tractable, they require extensive and accurate environmental data. Our approach is to use feedback control to estimate the channel such that the pressure at a vertical hydrophone array at the reference location consists of only a single mode. Knowing that the propagating field consisted originally of only a single mode at this reference location, observers further downrange can treat the feedback array as a virtual single mode source. This will allow experiments to determine the nature of propagation and scattering of modes through features of the water column such as internal wave fields.

The feedback control algorithm explicitly estimates the transfer function between each source array element and each hydrophone in the feedback array for the frequency of excitation. These estimated transfer functions are used to compute the source array weighting that should give the desired pressure profile of a single mode at the feedback array. By moving the control location from the source array to the feedback array, we have reduced the degrees of freedom we are trying to control from a large continuum to a small discrete set of modes. Recently, we formulated the transfer function estimator as a Kalman filtering state estimation problem. This formulation provides a means of incorporating information about the temporal and spatial statistics of the channel variations into the estimator.

The algorithm has been tested in detailed simulations of realistic ocean environments. The simulated ocean environments were based on measurements of conditions on the South Continental Shelf region around Martha's Vineyard. These simulations indicated the feedback control approach performs much better than previous attempts using open loop control in environments

with range-varying bathymetry when no a priori knowledge of the environment is available. In addition to these simulated results, a series of scale-model experiments in a wave tank have demonstrated the algorithm performs robustly under the limitations imposed by real propagation with physical sources and receivers. The control algorithm was able both to obtain the desired pressure field and react to transients in the environment. These results make us optimistic that the algorithm should work for full-scale ocean experiments.

1.7 Efficient Communication Using Chaotic Systems

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Project Staff

Brian Chen, Professor Gregory W. Wornell

This research explores the application of chaotic signals and systems for communication, with specific focus on discrete-time chaotic systems. Our main goals are to develop algorithms for estimating the state of chaotic systems under various forms of corruption and to investigate the potential role of these systems in communication applications such as modulation, coding, and deconvolution.

We have developed state estimation algorithms for chaotic sequences corresponding to so-called tent map dynamics in the presence of additive Gaussian noise, intersymbol interference, and multiple access interference and have evaluated their performance both analytically and empirically. In the additive white Gaussian noise case, the estimator is optimal under the maximum likelihood criterion, has a convenient recursive implementation, is asymptotically unbiased at high SNR, and has an error variance which decays exponentially with sequence length to a lower threshold which depends on the SNR. In the intersymbol interference case, we have developed three iterative algorithms and evaluated their performance empirically, considering the effects of sequence length, noise, and initialization. Finally, in the multiple access interference case, we have derived an optimal maximum likelihood estimator and the associated Cramer-Rao bound on the error variance. The estimator is asymptotically unbiased

at high SNR and has an error variance which depends on the particular tent map sequences.

We are currently investigating a modulation and coding system for transmission of analog sources in which the source data is encoded in the initial value of a tent map sequence. We are analyzing the performance of this so-called tent map coding system and are comparing it to the performance of other codes such as linear modulation codes and M-ary digital codes.

1.8 Signal Transmission and Recovery in Self-Synchronous Chaotic Systems

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Alan E. Freedman, Professor Alan V. Oppenheim

The goal of this project is to examine the extent to which it is possible to recover a transmitted signal, produced by a chaotic system and corrupted by an unknown filter present in the transmission channel. For self-synchronous chaotic systems, the uncorrupted transmission can drive an identical receiving system, and that drive signal can be exactly reproduced at the receiver. For small perturbations to the drive signal, the receiver will approximately synchronize to the transmitter. Measuring the energy in the error allows compensation for these perturbations, thus restoring the signal originally transmitted.

An algorithm will be developed in Matlab which will locate a compensating filter, ideally the inverse of the corrupting filter. Introducing this compensator at the receiver will cancel the corruption and restore the synchronization between the transmitter and receiver.

A successful algorithm for compensating for channel corruption would allow small perturbations such as speech signals to be masked beneath the chaotic signal with excellent opportunity for recovery at a receiving station; a receiver would need only achieve approximate synchronicity between the signal arriving and the reproduced output, and then subtract the two to recover the message being transmitted beneath the chaotic masking signal.

1.9 Algorithmic Structures for Signal Processing in Relation to the Rapid Prototyping of Application Specific Signal Processors

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Design and prototyping of signal processors must inherently take into account the structure of signal processing algorithms. Traditionally the signal processing community has relied heavily on a foundation of linear system theory and the associated tools of transform analysis. While this approach has formed the basis for many useful algorithms in a wide variety of applications over the last several decades, it has become increasingly clear that there are large classes of signal processing applications for which this basis is fundamentally inadequate. Many emerging classes of algorithms stem from new models and approaches, and current design tools and implementation structures are not necessarily well matched to these algorithms. The ARPA/Tri-Service Rapid Prototyping of Application Specific Signal Processors (RASSP) program is focused on dramatically improving the *process* by which digital signal processors are designed. Therefore, it is important that the RASSP tools and process be exercised and challenged by the demands of these new algorithms.

This research has two major goals. One is to identify and develop appropriate classes of emerging as well as more classical signal processing algorithms that are important but that the current RASSP tools and process cannot accommodate well. Classes of such algorithms may include, for example, interactive algorithms, algorithms oriented around partial differential equations, and certain types of multidimensional and multirate algorithms. These algorithms could have an impact on applications such as radar, sonar, geophysical processing, computed tomography, and biomedical ultrasound. A second

goal of this research is to provide specific input for a next generation of RASSP tools and process. One phase of this follows from having identified weaknesses in the currently available tools. Future work will consider how the tools need to be expanded and modified in order to make these new algorithms readily accessible to the designer of future signal processing systems.

1.10 Algebraic and Probabilistic Structure in Fault-Tolerant Computation

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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 Beckmann⁵ has shown how to obtain a variety of useful results and systematic constructive procedures.

In our research so far, we have been able to generalize this work to the case of computations occurring in semigroups and semirings⁶ and to outline a procedure that reflects such algebraically-based ABFT design into hardware. Currently, we are exploring extensions of our approach to sequences of "computations" associated with the evolution of dynamic systems in particular algebraic settings, such as linear systems over groups or rings, or finite automata and discrete-event systems. We also intend in future work to fold probabilistic models for failures and errors into the design and analysis of ABFT systems.

⁵ P.E. Beckmann, *Fault-Tolerant Computation Using Algebraic Homomorphisms*, RLE TR-580 (Cambridge: MIT Research Laboratory of Electronics, 1993).

⁶ C.N. Hadjicostis, *Fault-Tolerant Computation in Semigroups and Semirings*, RLE TR-594 (Cambridge: MIT Research Laboratory of Electronics, 1995).

1.11 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 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 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.

1.12 Multiscale Signal Processing with Fractal Renewal Processes

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Point processes with fractal characteristics have a potentially important role to play in the modeling of numerous natural and man-made phenomena, ranging from the distribution of stars and planets in the universe to the arrival pattern of transmission errors in communication channels. However, in contrast to fractal waveforms, which have been explored in considerable depth,⁷ the development of efficient algorithms for synthesizing, analyzing, and processing fractal point processes has generally proven difficult, largely due to the lack of an adequate mathematical framework. In this work, we introduce a novel multiscale representation for fractal point processes and apply it to a number of practical signal processing problems involving such point processes.

Our study of fractal point processes is focused primarily on an important subclass called fractal renewal processes which possess a sense of stationarity as well as self-similarity. Recently, we have developed a multiscale representation whereby a fractal renewal process is viewed as the random mixture of a multiscale family of constituent Poisson processes.⁸ This decomposition facilitates the study of fractal renewal processes, as it allows exploitation of existing efficient algorithms for Poisson processes. Indeed, based on the multiscale framework, we have developed an efficient algorithm for synthesizing fractal renewal pro-

⁷ G.W. Wornell, "Wavelet-based Representations for the $1/f$ Family of Fractal Processes," *Proc. IEEE* 81: 1428-1450 (1993).

⁸ W.M. Lam and G.W. Wornell, "Multiscale Synthesis and Analysis of Fractal Renewal Processes," *Proceedings of the Sixth IEEE Digital Signal Processing Workshop*, October 1994, pp. 67-70.

cesses. In addition, we have successfully applied this framework to several practical detection and estimation problems including estimation of the fractal dimension of a point process and the recovery of a fractal renewal process from corrupted measurements. Application of this framework to other practical problems is currently being investigated.

1.13 Low-Complexity Receivers for Code Division Multiple Access Systems

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Code Division Multiple Access (CDMA) systems, in which different users are assigned distinct signature waveforms for modulating their messages, have many attractive characteristics for wireless communication applications. For example, CDMA systems are well-suited to scenarios where there is significant multipath fading or jamming, or where the users transmissions are intermittent in nature.

Because the transmissions of users in CDMA systems are typically only quasi-orthogonal, traditional CDMA receivers based on single-user detection strategies suffer from a near-far problem, whereby inter-user interference can severely limit performance. CDMA receivers based on multi-user detection strategies—in which all users are decoded jointly at a receiver—overcomes this problem, but typically at great cost in terms of computational complexity, particularly when large numbers of users are involved.

This research is aimed at developing and analyzing alternative receiver structures for CDMA systems that require substantially less computational complexity than traditional implementations of multi-user detection, but with minimal performance degradation. Several such algorithms are currently under investigation, including a variety based on iterative EM-type maximum likelihood estimation.

1.14 Approximate Signal Processing

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Jeffrey T. Ludwig, Dr. 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 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.

We have successfully applied approximate processing concepts to the area of low-power signal processing. Techniques for reducing power consumption have become important due to the growing demand for portable multimedia devices. We have developed an approach to the design of low-power frequency-selective digital filters based on the concepts of adaptive filtering and approximate processing. The technique uses a feedback mechanism in conjunction with well-known implementation structures for FIR and IIR digital filters. Our algorithm is designed to reduce the total switched capacitance by dynamically varying the filter order based on signal statistics. A factor of 10 reduction in power consumption over fixed-order filters has been demonstrated for the filtering of speech signals.

In previous research it has been demonstrated that approximate processing algorithms can be effective 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 extend the development of formal structures for using approximate processing concepts in designing novel signal processing algorithms to areas such as time-frequency analysis, cepstral analysis, multipass analysis, adaptive beamforming, and image coding.

1.15 Tools for Decentralized Decision and Control with Application to Multiple Access Communications

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Decentralized decision-making is the problem of how more than one agent should make decisions when each agent has access to different information about the state of the system. Engineers have struggled with this problem with little progress for at least 25 years. In 1985, Tsitsiklis and Athans showed that the problem of finding optimal control laws for these problems is inherently difficult, explaining in part why so little progress had been made. Our goal is to develop and analyze easily computable bounds and computationally efficient protocols for this difficult problem. These tools can be applied to any problem that is recognized as one of decentralized decision-making, of which there are many. We will also explore the relationship between information structures and optimal performance. Finally, we plan to demonstrate the utility of our bounds and protocols on well-known difficult problems such as optimally controlling a multiple access communications channel.

1.16 Analysis and Applications of Systems Exhibiting Stochastic Resonance

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Stochastic resonance is a phenomenon encountered in certain bistable nonlinear systems (i.e., *systems with two stable points*) when driven by a periodic signal in noise. Specifically, these systems often display an enhancement of output signal-to-noise ratio (SNR) in a given input SNR regime. For this reason, such systems are appealing candidates for use in a variety of engineering contexts. In terms of signal analysis, such systems constitute potentially useful models for natural phenomena such as the regularity of appearance of earth's ice ages⁹ as well as for Detection Mechanisms in Certain Species, such as predator sensing by crayfish.¹⁰ In terms of signal synthesis, the induced SNR enhancement renders them attractive in a number of applications in signal communication and processing, such as robust communication and interference suppression. In order to exploit the phenomenon of stochastic resonance in such applications, there is a need for tools to analyze these systems in the presence of various forms and degrees of distortion.

One of the main directions of the proposed thesis is towards the development of novel techniques for analysis of dynamical systems exhibiting stochastic resonance, and considering their viability in various signal processing and communication contexts. In addition, the thesis explores the phenomenon of stochastic resonance in the context of general signal processing problems which includes signal detection, classification, and enhancement.

1.17 Distributed Digital Signal Processing Architectures for Approximate Processors

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⁹ R. Benzi, A. Sutera, and A. Vulpiani, "The Mechanism of Stochastic Resonance," *J. Phys.* A14: L453-L457 (1981).

¹⁰ J.K. Douglass, L. Wilkens, E. Pantazelou, and F. Moss, "Noise Enhancement of Information Transfer in Crayfish Mechanoreceptors by Stochastic Resonance," *Nature* 365: 337-340 (1993).

This research investigates the area of distributed digital signal processing (DDSP), studying the interactions among, and collective performance of, multiple approximate digital signal processors or processes. The approach we take is to study networks of digital signal processing (DSP) modules whose parameters and functionality can be varied to adapt to changes in specifications and constraints. The specifications that we focus on are ones that provide metrics or tolerances for various features of input and output quality (such as time- and frequency-resolution, quantization, probability of error), thus allowing the individual DSP modules to carry out what has come to be known as approximate processing. The flexibility allowed by approximate processing can be critical to accommodating constraints in a DDSP system. The constraints of interest involve such resources as cost, time, power, memory, and inter-processor communication. Our research includes modeling, the development of resource allocation and scheduling schemes, and simulation/testing (using University of California at Berkeley's Ptolemy system as a platform).

1.18 Sinusoidal Analysis-Synthesis

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Alan Seefeldt, Professor Alan V. Oppenheim

The use of a Sinusoidal Analysis-Synthesis (SAS) technique in the enhancement of speech corrupted by additive white noise is being examined. The SAS technique involves analyzing speech with the short time Fourier transform. Peaks in the magnitude spectrum of each frame are selected and matched from frame-to-frame in order to form sinusoidal tracks. The speech is then synthesized by interpolation through these tracks.

Currently, clean speech is being used as an aid in selecting sinusoidal tracks from noise-corrupted speech. Methods for smoothing these optimally-selected noisy tracks are being investigated. If satisfactory noise reduction can be achieved through this smoothing, methods for selecting tracks from

the noisy speech without any a priori information will be examined.

1.19 Signal Processing and Communication with Solitons

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Traditional signal processing algorithms rely heavily on models that are inherently linear. This is predominantly due to the tractability of their analyses and the applicability of such techniques to the rich class of signals that can be represented with Fourier methods. Nonlinear systems that support soliton solutions enjoy many of the properties of linear systems, making them attractive from an engineering standpoint. Although nonlinear, these systems are solvable through a technique known as inverse scattering, which can be viewed as an analog of the Fourier transform. Soliton systems also possess a class of eigenfunctions, known as solitons, which satisfy a nonlinear form of superposition. As these systems evolve in time, their soliton solutions display interesting signal dynamics as multiple solitons interact. In this thesis, we explore some of the signal processing implications of the dynamics of soliton signals. We present new analog circuits that support soliton signals and can be used as natural modulators and demodulators for a variety of soliton communication applications. These circuit models play an important role in investigating the effects of noise on soliton behavior. Finally, the signal dynamics also provide a mechanism for simultaneously decreasing signal energy while potentially boosting detection and estimation performance.

1.20 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. Although it is true that many modern speech recognizers can perform well in certain adverse environments when trained extensively under actual environmental conditions, even the most sophisticated ASR systems today are extremely sensitive to variations in these conditions, and must be carefully retrained for each new setting in which they are deployed.

In a given setting, a number of environmental factors can interfere with obtaining quality measurements of spoken language. In an office environment, for example, a spoken command intended for the recognizer can be corrupted by many different types of additive noise. Examples of noise sources in the workplace include background conversations, keyboard clicking, mechanical air circulation, and in general any movement of people, chairs, desk drawers, books, and paper. A noise source may produce sudden and transient interference, as in the case of a slamming door or a ringing telephone, or it may produce steady and persistent interference, as in the case of a humming computer fan or a buzzing overhead light.

The long-term objective of research in this area is to determine the signal processing technology required to maintain consistently good ASR performance in spite of possible changes in the recording environment. To improve the performance of an existing recognizer in the presence of additive noise, we have adopted a two-stage processing system in which: (1) the noisy input waveform is filtered to enhance the underlying speech, and (2) the enhanced waveform is analyzed by the recognizer in an attempt to decode the utterance.

In particular, our research has been directed at developing the first stage of this system for the case in which the physical parameters of the speaker are known in advance, but the parameters of the noise are unknown.

Both speech and noise are modeled as autoregressive processes over short time intervals; however, to account for the fact that the autoregressive parameters of these processes may change over time, each process is modeled globally using a Hidden Markov Model (HMM). The unknown HMM noise parameters are then computed iteratively such that the likelihood function over the entire utterance is eventually maximized. This iterative estimation is accomplished using the EM (Expectation-Maximization) algorithm, which is now a widely used mathematical tool for computing maximum likelihood estimates. As a by-product of this parameter estimation procedure, an optimum estimate of the underlying speech signal itself is generated, and hence may be subsequently used as input to the speech recognizer.

Important intermediate goals in this research will include quantifying the performance of an existing ASR system when the sensor measurements are preprocessed using the approach described above, and comparing this performance to that obtained using more standard adaptive filtering techniques.

1.21 Oceanographic Signal Processing

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Project Staff

Professor Arthur B. Baggeroer

Our research programs involve application of signal and array processing to problems in ocean acoustics and geophysics. These applications require an understanding of both signal processing and wave propagation; 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.

1.22 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 DARPA. A network of acoustic sources and receivers is being deployed in the Pacific this spring to monitor the ocean by measuring changes in acoustic travel times. These changes will be used to infer temperature changes.

1.22.1 Estimation of Normal Mode Amplitudes for Underwater Acoustics

Sponsor

University of California/Scripps Institute of Oceanography
Contract 1003-73-59

Project Staff

Kathleen E. Wage, Professor Arthur B. Baggeroer

The goal of this research is to develop array processing techniques for detection and estimation of broadband signals propagating in the low-order acoustic modes of the deep ocean. Normal modes are the eigenfunctions of the ocean waveguide and are useful to acousticians and oceanographers because the modal expansion coefficients contain valuable information about acoustic sources and the propagation medium. The objective of ocean acoustic tomography is to exploit the environmental dependence of signals, such as modes, in order to infer sound speed and temperature structure over large ranges. One goal of the ongoing Acoustic Thermometry of Ocean Climate (ATOC) project is to use modal phase and group velocity measurements in tomographic inversions. The purpose of our research is to develop methods of processing signals for experiments such as ATOC.

This work addresses two related signal processing problems. The first is the estimation of a time series of modal excitation coefficients based on measurements from a vertical hydrophone array. Since the modes form an orthonormal basis, similar to planewaves, computation of the mode coefficients is often referred to as modal beamforming.

We focus on robust adaptive methods of modal beamforming for broadband sources. An intriguing aspect of the waveform estimation problem is that it can be approached from the perspective of adaptive beamforming or of linear inverse theory. Both viewpoints provide valuable insights and indicate that this research is relevant to a broader class of signal processing problems.

The second aspect of this research concerns the detection of pulse arrivals in the mode waveforms and the estimation of mode travel times based on these arrivals. Although numerous researchers have proposed using modal group delay perturbations for tomographic inversions, very few have examined the signal processing required to determine the arrival times. In particular, tomographers have generally taken a naive approach to receiver design and have ignored possible effects of random phenomena such as internal waves. An objective of this work is to design optimal receivers for the modes which account for random fading and coupling as well as the intrinsic dispersion in the waveguide. Similar issues arise in the context of communication over fading dispersive channels. We hope to extend the concept of a diversity receiver for a single channel to the multidimensional mode estimation problem.

1.22.2 Detection and Estimation in Fractional Brownian Motion

Sponsor

U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Alex Che-Wei Wang, Professor Gregory W. Wornell, Professor Arthur B. Baggeroer

The $1/f$ family of fractal processes models a wide range of statistically scale-invariant phenomena. One model for this class of processes is the fractional Brownian motion framework developed by Mandelbrot and Van Ness¹¹ However, many basic finite-interval detection and estimation problems in this framework have been mathematically intractable. This research explores using techniques of state variable random processes to investigate solutions to these problems.

¹¹ B.B. Mandelbrot and H.W. Van Ness. "Fractional Brownian Motions, Fractional Noises and Applications," *SIAM Rev.* 10.: 422-436 (1968).

1.23 Publications

1.23.1 Journal Articles

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- Ludwig, J.T., S.H. Nawab, and A. Chandrakasan. "Low Power Filtering Using Approximate Processing for DSP Applications." *Proceedings of the Custom Integrated Circuit Conference*, May 1995.
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- Singer, A.C. "A New Circuit for Communication Using Solitons." IEEE Workshop on Nonlinear Signal and Image Processing, Halkidiki, Greece, June 1995.
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- Winograd, J.M., and S.H. Nawab. "A C++ Software Environment for the Development of Embedded Signal Processing." Conference *Proceedings ICASSP 95*, May 1995, vol. 4, pp. 2715-2718.
- Winograd, J., J. Ludwig, S.H. Nawab, A. Chandrakasan, and A.V. Oppenheim. "Approximate Processing and Incremental Refinement Concepts." Second ARPA RASSP Conference, Washington, D.C., July 1995.
- Winograd, J.M., and S.H. Nawab. "Mixed-radix Approach to Incremental DFT Refinement." SPIE '95 Conference, San Diego, California, July 1995.
- Wornell, G.W. "Efficient Multiuser Communication in the Presence of Fading." *Proceedings of the IEEE International Symposium on Information Theory*, Whistler, Canada, September 1995.

Wornell, G.W. "A New Class of CDMA Systems for Fading Channels." IEEE Workshop on Information Theory, Multiple-Access, and Queuing, St. Louis, Missouri, April 1995.

1.23.3 Technical Reports

Hadjicostis, C.N. *Fault-Tolerant Computation in Semigroups and Semirings*. RLE Technical Report No. 594. Cambridge: MIT Research Laboratory of Electronics, 1995.

Isabelle, S.H. *A Signal Processing Framework for the Analysis and Application of Chaotic Systems*. RLE Technical Report No. 593. Cambridge: MIT Research Laboratory of Electronics, 1995.

1.23.4 Book

Wornell, G.W. *Signal Processing with Fractals: A Wavelet-Based Approach*. Englewood Cliffs, New Jersey: Prentice-Hall, 1995.

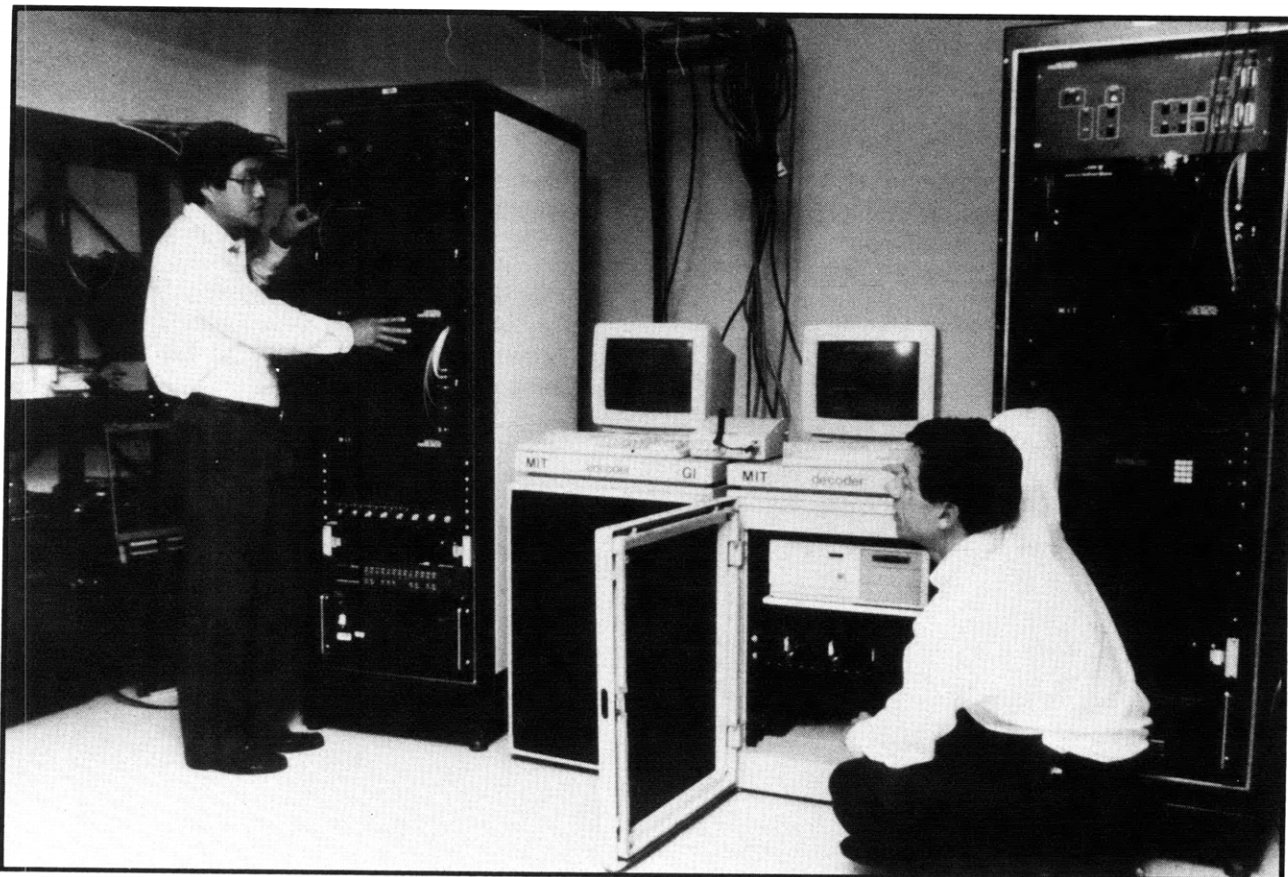
1.23.5 Theses

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Isabelle, S.H. *A Signal Processing Framework for the Analysis and Application of Chaotic Systems*. Ph.D. diss. Dept. of Electr. Eng. and Comput. Sci., MIT, 1995.



Professor Jae S. Lim (left) and graduate student Julien Nicolas working with the channel compatible digiCipher system at the Advanced Television Test Center in Alexandria, Virginia.