Section 2 Digital Signal Processing

Chapter 1. Digital Signal Processing Research Program

Academic and Research Staff

Professor Alan V. Oppenheim, Professor Arthur B. Baggeroer, Professor Anantha P. Chandrakasan, Professor Gregory W. Wornell, Dr. Steven H. Isabelle, Giovanni Aliberti

Visiting Scientists and Research Affiliates

Dr. Bernard Gold, Dr. S. Hamid Nawab,[1] Dr. James C. Preisig, Dr. Ehud Weinstein[2]

Graduate Students

Chalee Asavathiratham, Richard J. Barron, Soosan Beheshti, John R. Buck, Brian Chen, Trym H. Eggen, Christoforos N. Hadjicostis, Warren M. Lam, J. Nicholas Laneman, Li Lee, Haralabos C. Papadopoulos, Jeffrey T. Ludwig, James M. Ooi, Wendi B. Rabiner, Matthew J. Secor, Alan Seefeldt, Andrew C. Singer, Shawn M. Verbout, Kathleen E. Wage, Alex Che-Wei Wang

Technical and Support Staff

Darla J. Chupp, Janice M. Zaganjori

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. RLE's Digital Signal Processing Group carries out research on algorithms for signal processing and their applications. Current application areas of interest include signal enhancement and active noise cancellation; speech, audio and underwater acoustic signal processing; advanced beamforming for radar and sonar systems; and signal processing and coding for wireless and broadband multiuser communication networks.

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 ano 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 involves signal processing and coding for wireless multiuser systems and broadband communication networks. Specific interests include commercial and military mobile radio networks, wireless local area networks and personal communication systems, digital audio and television broadcast systems, and multimedia networks. Along with a number of other directions, we are currently exploring new code-division

[1] Associate Professor, Boston University, College of Engineering, Boston, Massachusetts.

[2] Department of Electrical Engineering, Systems Division, Faculty of Engineering, Tel-Aviv University, Israel; adjunct scientist, Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts.

multiple-access (CDMA) strategies, new techniques for exploiting antenna arrays in wireless systems, and new methods for modeling and management of traffic in high-speed packet-switched networks.

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 Model-Based Signal Enhancement

Sponsors

Lockheed Sanders, Inc.
Contract BZ4962
U.S. Army Research Laboratory
Grant QK-8819
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Richard J. Barron, Professor Alan V. Oppenheim

A common signal processing task is the estimation of an information-bearing signal from a distorted version of the waveform. Traditional solutions, such as the Wiener filter, assume a particular stochastic description for the signal source and channel distortion, and minimize an error criterion accordingly. In this project, we explore the use of signal features based on a model of the signal source as a method of signal estimation. A signal is often classified by the source from which it originates; speech or natural images are examples of certain classes of signal. Often there exist models of a particular signal source described by a few key parameters, or features, that capture much of the behavior of the signal. This feature information derived from the clean signal is often available to the processor that is enhancing the corrupted waveform. We have derived algorithms that exploit signal feature information to enhance signals from a variety of sources.

1.3 Multipass Receivers for Spread-Signature CDMA Systems

Sponsors

National Science Foundation
Grant MIP95-02885
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686
Grant N00014-95-1-0834
Grant N00014-96-1-0930

Project Staff

Soosan Beheshti, Professor Gregory W. Wornell

Spread-signature code-division multiple-access (CDMA) systems were recently introduced as an attractive alternative to conventional CDMA systems for use in time-varying multipath environments. Using long signatures in an overlapped manner for successive symbols, spread-signature CDMA can achieve a substantial temporal diversity benefit. Furthermore, the broadband nature of the signatures allows an additional spectral diversity benefit to be simultaneously realized.

In this research, computationally efficient multipass demodulation and decoding algorithms are developed for use in receivers with these systems. These algorithms efficiently suppress both intersymbol and interuser (multiple-access) interference to achieve a substantial diversity benefit and good near-far resistance characteristics. Moreover, it is shown that relatively few iterations are required for convergence to typical target bit-error rates. Several other aspects of the performance of the algorithms are also explored.

1.4 Single Mode Excitation in the Shallow Water Acoustic Channel Using Feedback Control

Sponsor

U.S. Navy - Office of Naval Research
Grant N00014-95-1-0362
Grant N00014-93-1-0686

Project Staff

John R. Buck, Professor Alan V. Oppenheim

The shallow water acoustic channel supports far-field propagation in a discrete set of modes. Ocean experiments have confirmed the modal nature of acoustic propagation, but no experiment has successfully excited only one of the suite of mid-frequency propagating modes propagating in a coastal environment. The ability to excite a single mode would be a powerful tool for investigating shallow water ocean processes. A feedback control algorithm incorporating elements of adaptive estimation, underwater acoustics, array processing and control theory to generate a high-fidelity single mode is presented. This approach also yields a cohesive framework for evaluating the feasibility of generating a single mode with given array geometries, noise characteristics and source power limitations. Simulations and laboratory waveguide experiments indicate the proposed algorithm holds promise for ocean experiments.

1.5 Coding and Modulation of Analog Data for Transmission over Broadcast and Fading Channels

Sponsors

National Defense Science and Engineering Fellowship
U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686
Grant N00014-95-1-0834
Grant N00014-96-1-0930

Project Staff

Brian Chen, Professor Gregory W. Wornell

In many communication applications, the information to be transmitted over the channel of interest is inherently analog (i.e., continuous-valued) in nature. A traditional digital approach for transmitting such data involves appropriately quantizing the source data and encoding the quantized data using a suitably designed channel code so that the quantized data can be recovered with arbitrarily low probability of error. Shannon's source-channel separation theorem is frequently invoked to argue that performance need not be sacrificed using such an approach. However, for many important classes of channels that arise in practice, Shannon's theorem does not apply and in fact, performance is necessarily sacrificed using this digital approach. Such is the case, for example, when the signal-to-noise ratio (SNR) is unknown at the transmitter, or equivalently, in broadcast scenarios where there are multiple receivers with different SNRs, as well as in low-delay systems operating in the presence of time-selective fading due to multipath propagation.

In these kinds of settings, which arise in, for example, a variety of wireless communication systems, separate source and channel coding is inherently suboptimum. Such digital approaches are inadequate because their performance depends crucially on being able to choose the proper number of quantization levels, which in turn depends on there being a specific target SNR. Motivated by these observations, in this research we explore efficient analog coding strategies for scenarios precisely of this type.

The properties of chaotic dynamical systems make them useful for channel coding for a variety of practical communication applications. We have developed novel analog error-correcting codes that are potentially useful in such applications, examples of which are communication over broadcast channels and low-delay communication in time-varying fading environments. These *systematic* analog codes are generated from iterations of a nonlinear state space system governed by chaotic dynamics, with the analog message embedded in the initial state. Within this class are practical codes having computationally very efficient recursive receiver structures and important performance advantages over conventional codes. We are in the process of developing a method for generalizing and optimizing such codes. Indeed, we envision a general framework for developing a broader error-correction coding theory that encompasses both the theory of modern digital codes and classical analog modulation techniques.

1.6 Underwater Acoustic Communication

Sponsor

U.S. Navy - Office of Naval Research
Grant N00014-95-1-0362
Grant N00014-93-1-0686

Project Staff

Trym H. Eggen, Professor Arthur B. Baggeroer

Underwater acoustic coherent communication is possible by conventional communication systems only when the underwater communication channel is especially simple. One problem is the frequency dispersion of the channel, and by using a specific linear time variant model analysis of conventional receivers as well as development of a new receiver has been carried out. The receivers have been tested on real data from the ocean, and emphasis has been on communication channels where the Doppler spread is more severe than the delay spread which is only a subset of all the existing underwater communication channels. Some physical conditions for these channels to exist have been derived, and examples from the real ocean has been found. Many conventional receivers, as used in other communication areas as cellular radio and indoor wireless, are not well suited for this type of channel.

1.7 Algebraic and Probabilistic Structure in Fault-Tolerant Computation

Sponsors

Lockheed Sanders, Inc.
Contract BZ4962
U.S. Navy - Office of Naval Research
Grant N00014-93-0686

Project Staff

Christoforos N. Hadjicostis, Professor George C. Verghese

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 Beckmann[3] 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 semirings[4] 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 semirings, or finite automata and discrete-event systems. Along these lines, we have obtained an illuminating characterization of all possible redundant linear time-invariant (LTI) state-space embeddings of a given LTI state-space model. We also intend in future work to fold probabilistic models for failures and errors into the design and analysis of ABFT systems.

1.8 Multiscale Signal Processing with Fractal Renewal Processes

Sponsors

U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686
Grant N00014-95-1-0834
Grant N00014-96-1-0930

Project Staff

Warren M. Lam, Professor Gregory W. Wornell

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 occurrence of transmission errors in communication channels and traffic over a number of packet-switched networks. However, in contrast to fractal waveforms, which have been explored in considerable depth,[5] 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.[6] Exploiting existing efficient algorithms for Poisson processes, this framework has appeared promising for the study of fractal renewal processes. Indeed, based on the multi-

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

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

5 G.W. Wornell, "Wavelet-based Representations for the Family of Fractal Processes," *Proc. IEEE* 81(10): 1428-1450 (1993).

6 W.M. Lam, and G.W. Wornell, "Multiscale Synthesis and Analysis of Fractal Renewal Processes," *Proceedings of the Sixth IEEE DSP Workshop*, Yosemite, California, October 1, 1994.

scale framework, we have developed an efficient algorithm for synthesizing fractal renewal processes. In addition, we have successfully applied this framework to several practical signal processing problems including estimation of the fractal dimension of a point process and recovery of a fractal renewal process from corrupted measurements. Application of this framework to other practical problems is currently being investigated.

1.9 Estimation and Equalization of Wireless Fading Channels

Sponsors

National Science Foundation
  Graduate Research Fellowship
  Grant MIP 95-02885
U.S. Navy - Office of Naval Research
  Grant N00014-93-1-0686
  Grant N00014-95-1-0834
  Grant N00014-96-1-0930

Project Staff

J. Nicholas Laneman, Professor Gregory W. Wornell

A central issue in the wireless communications setting is the problem of signal fading. Due to multiple propagation paths, many copies of the transmitted signal arrive at the receiver antenna, each with a given attenuation level and phase shift. When the receiver antenna is set in motion, as is usually the case in such applications as cellular telephony, the received power level fluctuates since the multipath components add constructively or destructively. Because of this fading characteristic, wireless channels exhibit dramatically poorer bit-error performance than traditional additive white Gaussian noise channels when using uncoded transmissions.

Recent work[7] has suggested a technique known as spread-response precoding for combating signal fading found in wireless links. The idea behind this sort of precoding is to distribute the energy of each symbol in time to achieve the average effect of the channel rather than the instantaneous fade. A key element of the receiver is an equalizer which essentially inverts the effect of the fading channel. In our present work, we focus on a realistic equalizer structure derived from channel estimates.

We are evaluating a joint state and parameters estimator for the channel response based on Kalman filtering ideas and the Estimate Maximize (EM) algorithm. We are studying how to use estimates of the channel or channel inverse to best equalize the received signal, and how to take advantage of the special properties of the transmitted signal to allow the algorithm to perform blindly.

A flexible set of hardware has been assembled for demonstrating a variety of these signal processing algorithms indoors. Special purpose analog hardware has been built for modulating baseband signals to radio frequencies and back. The laboratory includes digital-to-analog (D/A) and analog to digital (A/D) converters for converting the baseband signal into discrete-time. Finally, four digital signal processors (DSPS) are used to perform the precoding and equalization of the baseband signals. The high computational power of the DSPS allow us to implement complex algorithms in real-time.

1.10 Properties of Approximate Parks-McClellan Filters

Sponsors

U.S. Air Force - Office of Scientific Research
  Grant F49620-96-1-0072
U.S. Army Research Laboratory
  Grant QK-8819
U.S. Navy - Office of Naval Research
  Grant N00014-93-1-0686

Project Staff

Li Lee, Professor Alan V. Oppenheim

For digital signal processing applications with real-time or low-power constraints, it is often desirable to use algorithms whose output quality can be adjusted depending on the availability of resources such as time or power. For this reason, recently there has been increased interest in approximate signal processing algorithms whose intermediate results represent successively better approximations to the desired solution. We have observed empirically that each coefficient in a Parks-McClellan filter converges to a steady state

[7] G.W. Wornell, "Spread-Response Precoding for Communication over Fading Channels," *IEEE Trans. Info. Theory* 42(2): 488-501 (1996); G.W. Wornell, "Spread-Signature CDMA: Efficient Multiuser Communication in the Presence of Fading," *IEEE Trans. Info. Theory* 41(5): 1418-1438 (1995).

value as the filter length increases. This suggests the possibility of obtaining filters that are near optimal while "re-using" filter coefficients from shorter filters in the design of longer filters. In the context of approximate processing this then allows a filtering operation to be done in stages. A paper demonstrating this observation and examining some of its implications will be presented in ICASSP '97.

1.11 Distributed Signal Processing

Sponsors

U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Army Research Laboratory
Grant QK-8819
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Li Lee, Professor Alan V. Oppenheim

We have initiated a project directed at implementing signal processing algorithms in a distributed network environment with unpredictable and dynamically changing resources. Our initial approach is to define a hierarchical set of signal processing modules with rules for describing the hierarchy. Our focus is on the signal processing aspects rather than the network aspects and consequently we represent the network issues statistically. Work on this project this year has involved defining the primitives and hierarchy and implementing a first stage simulation of the structure.

1.12 Approximate Signal Processing

Sponsors

Lockheed Sanders, Inc.
Grant N00014-93-1-0686
Contract BZ4962
U.S. Army Resarch Laboratory
Grant QK-8819
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Jeffrey T. Ludwig, Dr. S. Hamid Nawab

It is increasingly important to structure signal processing algorithms and systems to allow for flexibility in trading off between the accuracy or optimality of their results and their utilization of resources such as time, power, bandwidth, and physical space. Approximate signal processing algorithms are designed to incorporate such tradeoffs.

Our recent research indicates the enormous potential of approximate signal processing algorithms. These results show progress toward the ultimate objective of developing, within the context of signal processing and design, a more general and rigorous framework for utilizing and expanding approximate processing concepts and methodologies.

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.

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, adaptive beamforming, and image coding.

1.13 New Techniques for Communication with Feedback

Sponsor

U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686
Grant N00014-95-1-0834
Grant N00014-96-1-0930

Project Staff

James M. Ooi, Professor Gregory W. Wornell

Many communication links are inherently bidirectional, supporting the two-way exchange of voice, video, and other data between a pair of users. In such scenarios, a natural feedback path exists for each user's transmission, and as is well known this feedback path can generally be exploited to improve overall system performance in a variety of ways. Indeed, almost all duplex communication links in widespread use today exploit the availability of feedback, usually via an automatic repeat-request (ARQ) or related protocol.

In a typical feedback communication system, the transmitter sends data to a receiver over a noisy forward channel, and receives information about what the receiver actually observes via a feedback channel. The feedback path is often a relatively noise-free link, particularly when the receiver is feeding back information to the transmitter either at comparatively low rate or high power. Power asymmetry, specifically, arises naturally in a number of existing applications, including, e.g., Earthbound transmissions in satellite systems, and reverse (mobile-to-base) link transmission in a cellular mobile radio network. More generally, the noise-free feedback channel model is a good one for communication between portable, low-power transmitters and stationary, high-power receivers in a host of emerging systems for providing wireless personal communication services such as cellular telephony, paging, wireless local area networks, and wireless private branch exchanges.

While it is well established that feedback does not increase the capacity of discrete memoryless channels, it is well known that noise-free feedback can be used to substantially lower complexity and increase reliability of communication in practice.

We have developed a powerful framework for low-complexity, high-reliability communication over channels with feedback. We have used the framework to develop capacity-achieving coding schemes for arbitrary discrete memoryless and finite-state channels with noise-free feedback. We have also developed a universal communication scheme in which neither the transmitter nor the receiver need know the channel statistics to communicate reliably. We are exploring how the framework can be applied to coding for multiple-access channels as well as what the framework tells us about coding for channels with noisy feedback.

1.14 Analysis and Applications of Systems Exhibiting Stochastic Resonance

Sponsors

U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Haralabos C. Papadopoulos, Professor Gregory W. Wornell

Stochastic resonance is a phenomenon encountered in certain nonlinear systems when driven by noisy information-bearing signals. Specifically, increasing the input noise level in these systems often results in an enhancement of the information-bearing signal response, reflected for example, as output signal-to-noise ratio (SNR) enhancement, or as improved detection/estimation performance. Such systems are therefore 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 ages,[8] as well as for detection mechanisms in certain species, such as predator sensing by crayfish.[9] In terms of signal synthesis, the induced signal enhancement renders them attractive in a number of applications in signal communication and processing. In order to exploit 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 our research 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 research explores the phenomenon of stochastic resonance in the context of general signal processing problems which includes signal detection, classification, and enhancement.

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

9 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).

1.15 Modeling and Design of Approximate Digital Signal Processors and Approximate DSP Networks

Sponsors

U.S. Army Research Laboratory
Grant QK-8819
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Matthew J. Secor, Professor George C. Verghese, Professor Alan V. Oppenheim

This research investigates the area of approximate digital signal processing, studying the interactions among, and collective performance of multiple approximate 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 placed on a system comprised of approximate processors. 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. Another major area of our research is the development of new DSP algorithms and modification of existing DSP algorithms such that they exhibit incremental refinement properties and thus can be incorporated into larger approximate processing systems.

1.16 Sinusoidal Analysis Synthesis

Sponsors

Lockheed Sanders, Inc.
Contract BZ4962
U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Alan Seefeldt, Professor Alan V. Oppenheim

In our research, the use of sinusoidal analysis/synthesis (SAS) for the enhancement of noise corrupted speech is being explored. SAS approximates a digital speech waveform as a finite sum of time varying sinusoidal tracks. For the purposes of enhancement, the idea is to extract from the spectrum of the corrupted speech sinusoidal tracks that correspond to the speech alone. In order to attain an upper bound on the performance of SAS enhancement, the original uncorrupted speech is used as an aid in this track extraction procedure. Various processing techniques, such as spectral subtraction and amplitude smoothing, are then applied to the extracted tracks to reduce any remaining noise residual. The quality of speech enhanced with this SAS technique is being compared to that of previously developed enhancement procedures.

1.17 Signal Processing and Communication with Solitons

Sponsors

U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Army Research Laboratory
Grant QK-8819
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Andrew C. Singer, Professor Alan V. Oppenheim

Traditional signal processing algorithms rely heavily on models that are inherently linear. Such models are attractive both for their mathematical tractability and their applicability to the rich class of signals that can be represented with Fourier methods. Nonlinear systems that support soliton solutions share many of the properties that make linear systems attractive from an engineering standpoint. Although nonlinear, these systems are solvable through inverse scattering, a technique analogous to the Fourier transform for linear systems. Solitons are eigenfunctions of these systems which satisfy a nonlinear form of superposition and display rich signal dynamics as they interact. By using solitons for signal synthesis, the corresponding nonlinear systems become specialized signal processors which are naturally suited to a number of complex signal processing tasks. Specific analog circuits can generate soliton signals and can be used as natural multiplexers and demultiplexers in a number of potential soliton-based wireless communication applications. These circuits play an important role in investigating the effects of noise on soliton behavior. Finally, the soliton signal dynamics can provide a mechanism for decreasing transmitted signal energy while enhancing signal detection and parameter estimation performance.

1.18 A Parametric Framework for Non-Gaussian Signal Processing

Sponsors

U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Army Research Laboratory
Grant QK-8819
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Shawn M. Verbout, Professor Alan V. Oppenheim

Recent work has been aimed at developing a framework for analyzing and processing discrete-time non-Gaussian autoregressive (AR) random signals. As in the conventional linear-Gaussian AR time series model, the signal equation consists of two components: a regression component and a driving noise component. In the newly proposed model, the regression parameters are taken to be constant, but the parameters of the Gaussian driving noise are subject to abrupt changes over time that occur according to a finite-state Markov chain. Important problems in statistical signal processing that are being analyzed under this new model include (1) identification of unknown signal parameters, and (2) filtering of signals (with known parameters) in additive noise.

The classical linear AR time series model has long been used for the statistical analysis of experimental observations. Though the AR model is a rather restricted version of the general linear model, it has gained wide acceptance in disciplines such as economics, biology, geophysics, and engineering for several reasons: (1) it is inherently simple, both mathematically and computationally; (2) it is capable of representing a wide range of correlation patterns with a relatively small number of parameters; and (3) it is ideally suited to representing many natural phenomena, which tend to be strongly temporally correlated and thus exhibit sharp spectral peaks, particularly in low-frequency bands.

In many practical situations, the aspect of the conventional AR time series model that limits its performance is not the assumption that the underlying process is autoregressive; rather, it is the assumption that the process is Gaussian. Although the Gaussian model is adequate in certain cases, it is not flexible enough to represent populations that are, for example, heavy-tailed or strongly skewed. In fact, since the Gaussian pdf falls off sharply at values that are even moderately far from the mean, the Gaussian assumption is clearly inappropriate when the driving noise is characterized by gross fluctuations in amplitude, sudden random bursts, or frequently occurring spikes or impulses. These kinds of driving inputs are typically encountered in problems such as speech processing, exploration seismology, low-frequency communication systems, and underwater signal detection.

In recent research, we have considered a modified version of the standard AR time series model in which the driving noise is no longer taken to be i.i.d. and Gaussian. Instead, we assume that the driving noise is characterized, at each time step, by one of a finite set of underlying regimes, and that it switches randomly from one regime to another according to a Markov chain with constant transition probabilities. At a given time step, the regime of the driving process is selected by a discrete Markovian regime variable, and the input noise sample at that time is drawn from a Gaussian pdf whose mean and variance are dependent on the current value of the regime variable. This model includes the case in which the driving noise is an i.i.d. Gaussian sequence; clearly, this case is represented by letting the Markov chain have only one state. However, the model is much more general than the conventional model in that it allows for a wide range of densities and temporal correlation structures for the input samples.

Although the proposed AR model will undoubtedly yield a more accurate representation of many physical systems, it does not enjoy the mathematical simplicity associated with the conventional Gaussian model. Moreover, widely used methods that are based on the Gaussian assumption–methods such as least squares (for parameter estimation) or Kalman filtering (for signal estimation)–can not be expected to yield optimal or even near-optimal performance for many signals that are accurately characterized by the proposed model. Thus, the main challenges of this work lie in the creation of new algorithms for signal and parameter estimation, and more generally in the development of a unifying parametric framework for handling non-Gaussian signals.

1.19 Array Processing Techniques for Broadband Mode Estimation and Modal Tomography

Sponsor

U.S. Navy - Office of Naval Research
Grant N00014-95-1-0362

Project Staff

Kathleen E. Wage, Professor Arthur B. Baggeroer

This research focuses on advanced array processing techniques for underwater applications such as ocean acoustic tomography. Specifically, the goal of this research is to develop a signal processing framework for estimating the normal mode decomposition of low-frequency, broadband receptions. Modal representations are useful in seismo-acoustic modeling since they are directly related to a solution of the wave equation and because mode amplitudes and phases contain valuable information about the source and the propagation medium. Modal signal processing is an important area of research for several reasons. First, verification of mode propagation and scattering theories requires accurate estimates of the field to be determined from measurements made using mode-resolving arrays. A second motivation is that acoustic tomography applications rely on very precise measurements of mode travel times. The Acoustic Thermometry of Ocean Climate (ATOC) project is an ongoing research project designed to demonstrate that tomographic systems can be used to measure ocean climate variability over ranges of 3,000 to 10,000 km in the North Pacific. Incorporating normal mode data from long-range propagation studies into an inversion for environmental parameters requires a thorough understanding of the resolution of the underlying estimators. The purpose of this research is to develop and evaluate methods of processing modal signals for projects such as ATOC.

Typically, acoustic modes are used for describing and analyzing the temporal/spatial structure of narrowband signals. The overall objective of this research is to develop a general framework for broadband modes and, using this framework, to develop algorithms for detecting and analyzing broadband modal signals. Specifically, there are two related signal processing problems that this research will address. The first is the estimation of a time series of modal excitation coefficients based on measurements from a hydrophone array. Previous work on mode estimation has concentrated primarily on modal decompositions for narrowband sources, often in the context of source localization problems. Studies of broadband sources, such as those used for tomography, are rather limited. This research will define a framework for broadband mode estimation which can be used to explore the time/frequency resolution tradeoffs inherent in the processing of transient or non-stationary signals. Within this framework research efforts will focus on developing robust data-adaptive methods for modal beamforming, using concepts similar to those of matched field processing. Performance of conventional mode estimators is directly linked to how well the hydrophone array samples the water column. The intent of the robust techniques is to mitigate the sensitivity of the estimator to single-sensor failures and to aid in the design of shorter arrays with minimal loss in resolution capabilities. Mode estimation is closely related to adaptive beamforming and linear inverse theory, thus the results of this research may be relevant to a broader class of signal processing problems.

The second issue this research will address is the detection of broadband pulse arrivals in the acoustic modes and the estimation of the associated travel times. 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. An objective of the proposed research is to develop optimal receivers for the modal signals and to characterize their performance. The receiving strategies must account for the dispersive nature of the ocean waveguide and extend to channels that have random coupling and fading due to internal waves.

1.20 Multiscale State-Space Algorithms for Processing 1/f Signals

Sponsors

U.S. Air Force - Office of Scientific Research
Grant F49620-96-1-0072
U.S. Navy - Office of Naval Research
Grant N00014-93-1-0686

Project Staff

Alex Che-Wei Wang, Professor Gregory W. Wornell

Natural landscapes, noise in electrical devices, and fluctuations in the stock market are among the extraordinary variety of phenomena that exhibit fractal structure. The prevalence of fractal geometry in nature indicates the value of algorithms for processing fractal signals.

The 1/f processes are an important class of fractal random processes. Due to the wide range of phenomena modeled as 1/f processes, many useful applications based on processing these signals can be envisioned. Algorithms for predicting future values of a 1/f signal given observations of the process over a finite time interval could have applications in economic forecasting, for instance.

This research develops signal processing algorithms involving 1/f processes both as the signal of interest and as an obscuring noise process. We develop a linear time-invariant (LTI) multiscale state-space model for discrete 1/f processes. This model leads naturally to computationally efficient algorithms for processing 1/f signals with traditional linear LTI state-space methods such as the Kalman filter and Kalman smoother.

1.21 Publications

1.21.1 Journal Articles

Buck, J.R., J.C. Preisig, M. Johnson, and J. Catipovic. "Single Mode Excitation in the Shallow Water Acoustic Channel Using Feedback Control." *IEEE Oceanic Eng.* Forthcoming.

Ludwig, J.H., S.H. Nawab, and A.P. Chandrakasan. "Low Power Digital Filtering Using Approximate Processing." *IEEE Solid-State Circuits* (31)3: 395-400 (1996).

Ooi, J.M., S.M. Verbout, J.T. Ludwig, and G.W. Wornell. "A Separation Theorem for Periodic Sharing Information Patterns in Decentralized Control." *IEEE Trans. Automat. Contr.* Forthcoming.

Winograd, J., J. Ludwig, H. Nawab, A. Chandrakasan, and A.V. Oppenheim. "Approximate Signal Processing." *VLSI Signal Process.* Forthcoming.

Wornell, G.W. "Emerging Applications of Multirate Signal Processing and Wavelets in Digital Communications." *Proc. IEEE Special Issue on Applications of Wavelets* (84)4: 586-603 (1996).

Wornell, G.W. "Spread-Response Precoding for Communication over Fading Channels." *IEEE Trans. Info. Theory* (42)2: 488-501 (1996).

1.21.2 Conference Papers

Chen, B., and G.W. Wornell. "Efficient Channel Coding for Analog Sources using Chaotic Systems." *Proceedings of Globecom '96*, London, England, November 18-22, 1996.

Isabelle, S.H., and G.W. Wornell. "Recursive Multiuser Equalization for CDMA Systems in Fading Environments." *Proceedings of the Allerton Conference on Communication, Contr., and Computing*, Allerton, Illinois October 1996.

Lam, W., and G.W. Wornell. "Multiscale Analysis of Fractal Point Processes and Queues." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

Lee, L., and R.C. Rose. "Speaker Normalization Using Efficient Frequency Warping Procedures." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

Ludwig, J.T., S.H. Nawab, and A.P. Chandrakasan. "Convergence Results on Adaptive Approximate Filtering." *Proceedings of the International Symposium on Optical Science, Engineering and Instrumentation*, Denver, Colorado, August 1996.

Narula, A., M.D. Trott, and G.W. Wornell. "Information-theoretic Analysis of Multiple-Antenna Transmission Diversity for Fading Channels."*Proceedings of the International Symposium on Information Theory and Applications*, Victoria, Canada, September 1996.

Ooi, J.M., and G.W. Wornell. "Decentralized Control of a Multiple Access Broadcast Channel: Performance Bounds." *Proceedings of the International Conference Decentralized Control*, Japan, December 1996.

Oppenheim, A.V., K. Cuomo, and R. Barron. "Channel Equalization for Self-Synchronizing Chaotic Systems." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

Papadopoulos, H.C., and G.W. Wornell. "A Class of Stochastic Resonance Systems for Signal Processing Applications." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

Ramamurthy, M., S.H. Nawab, J.M. Winograd, and B.L. Evans. "Integrated Numeric and Symbolic Signal Processing Using a Heterogeneous Design Environment." *Proceedings of the International Symposium on Optical Science, Engi-*

neering and Instrumentation, Denver, Colorado, August 1996.

Singer, A.C. "Detection and Estimation of Soliton Signals." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

Winograd, J.M., S.H. Nawab, and A.V. Oppenheim. "FFT-Based Incremental Refinement of Suboptimal Detection." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

Winograd, J.M., J. Ludwig, S.H. Nawab, A.V. Oppenheim, and A. Chandrakasan. "Flexible Systems for Digital Signal Processing." *Proceedings of the AAAI Fall Symposium on Flexible Computation*, Cambridge, Massachusetts, November 9-11, 1996.

Wornell, G.W., and M.D. Trott. "Signal Processing Techniques for Efficient Use of Transmit Diversity in Wireless Communications." *Proceedings of ICASSP '96*, Atlanta, Georgia, May 7-10, 1996.

1.21.3 Technical Reports

Chen, B. *Efficient Communication over Additive White Gaussian Noise and Intersymbol Interference Channels Using Chaotic Sequences.* RLE TR-598. Cambridge: MIT Research Laboratory of Electronics, 1996.

Ooi, J.M., and G.W. Wornell. *Fast Iterative Techniques for Feedback Channels* RLE TR-613. Cambridge: MIT Research Laboratory of Electronics, 1996.

Singer, A.C. *Signal Processing and Communication with Solitons.* RLE TR-599. Cambridge: MIT Research Laboratory of Electronics, 1996.

Verbout, S.M., J.M. Ooi, J.T. Ludwig, and A.V. Oppenheim. *Parameter Estimation for Autoregressive Gaussian-Mixture Processes: The EMAX Algorithm.* RLE TR-611. Cambridge: MIT Research Laboratory of Electronics, 1996.

1.21.4 Books

Buck, J.R., M.M. Daniel, and A.C. Singer. *Computer Explorations in Signals and Systems Using Matlab.* Upper Saddle River, New Jersey: Prentice Hall, 1997.

Oppenheim, A.V., A.S. Willsky, and S.H. Nawab. *Signals and Systems.* 2d ed. Upper Saddle River, New Jersey: Prentice Hall, 1997.

1.21.5 Theses

Asavathiratham, C. *Digital Audio Filters Design Using Warped Filters.* M. Eng. thesis. Dept. of Electr. Eng. and Comput. Sci., MIT, 1996.

Barron, R.J. *Channel Equalization for Self-Synchronizing Chaotic Systems.* M. Eng. thesis. Dept. of Electr. Eng. and Comput. Sci., MIT, 1996.

Beheshti, S. *Techniques for Enhancing the Performance of Communication Systems Employing Spread-Response Precoding.* M. Eng. thesis. Dept. of Electr. Eng. and Comput. Sci., MIT, 1996.

Buck, J.R. *Single Mode Excitation in the Shallow Water Acoustic Channel Using Feedback Control.* Ph.D. diss., Dept. of Electr. Eng. and Comput. Sci., MIT, 1996.

Chen, B. *Efficient Communication over Additive White Gaussian Noise and Intersymbol Interference Channels Using Chaotic Sequences.* M. Eng. thesis. Dept. of Electr. Eng. and Comput. Sci., MIT, 1996.

Lee, L. *A Frequency Warping Approach to Speaker Normalization.* M. Eng. thesis. Dept. of Electr. Eng. and Comput. Sci., 1996.

Singer, A.C. *Signal Processing and Communication with Solitons.* Ph.D. diss. Dept. of Electr. Eng. and Comput. Sci., MIT, 1996.

1.22 Network-Driven Motion Estimation for Wireless Video Terminals

Sponsors

National Science Foundation
Graduate Fellowship
U.S. Army Research Laboratory/ARL Advanced
Sensors Federated Lab Program
Contract DAAL01-96-2-0001

Project Staff

Professor Anantha P. Chandrakasan, Wendi B. Rabiner

Video is becoming an integral part of many portable devices such as wireless cameras, personal com-

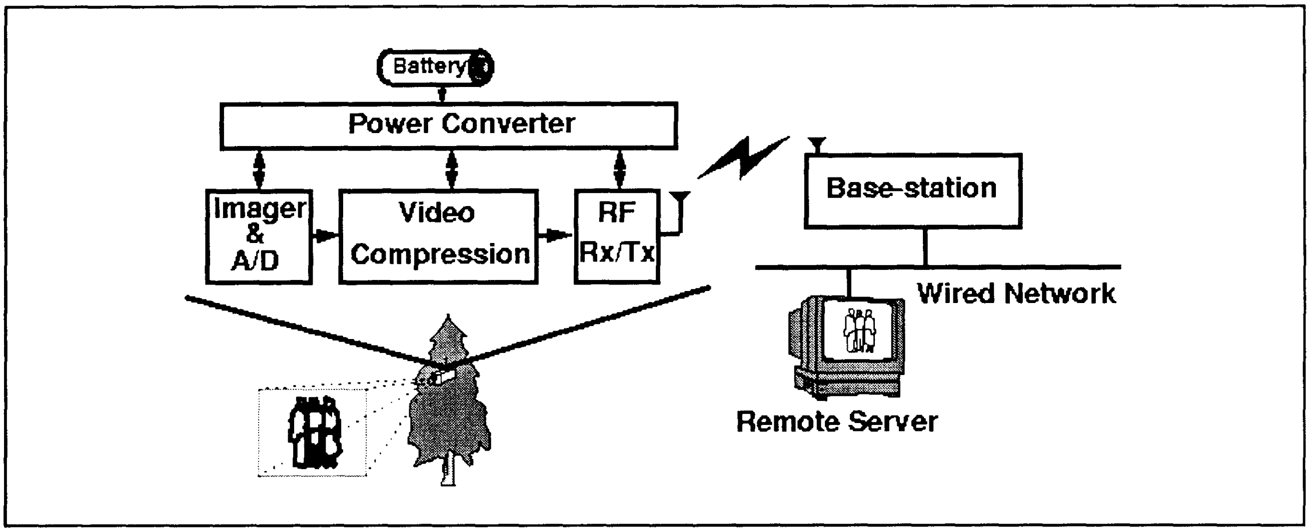

Figure 1. Wireless video terminals in a networked environment.

munication devices, and video cellular phones. Due to the massive amount of data contained in video signals and the limited bandwidth of the wireless channel, developing compression techniques for these applications is extremely important. Conventional video systems use some form of scene motion estimation/motion compensation at the encoder to reduce the temporal correlation inherent in most video signals and hence achieve high compression ratios. However, since most motion estimation algorithms require a large amount of computation, it is undesirable to use them in power constrained applications, such as battery operated wireless video terminals. Minimizing power dissipation is the key to maximizing battery lifetime and thus should be one of the driving forces when designing motion-estimation algorithms for portable video encoders.

Figure 1 shows a low-power wireless camera in a networked environment. The goals for the design of this system include a long battery lifetime as well as high video compression ratios. We have developed a motion-estimation algorithm, termed network-driven motion estimation, which reduces the power dissipation of wireless video devices in a networked environment by exploiting the predictability of object motion. Since the location of an object in the current frame can often be predicted accurately from its location in previous frames, it is possible to optimally partition the motion-estimation computation between battery operated portable devices and high powered compute servers on the wired network.

In network driven motion estimation, a remote high powered resource at the base-station (or on the wired network) predicts the motion vectors of the current frame from the motion vectors of the previous frames. The base-station then sends these predicted motion vectors to a portable video encoder, where motion compensation proceeds as usual. Network-driven motion estimation adjusts the coding algorithm based on the amount of motion in the sequence. This technique uses motion prediction to code portions of the video sequence which contain a large amount of motion and conditional replenishment to code portions of the sequence which contain little scene motion. Network driven motion estimation achieves a reduction in the number of operations performed at the encoder for motion estimation by over two orders of magnitude, while introducing minimal degradation to the decoded video compared with conventional full search encoder-based motion estimation, as shown in figure 2.

Figure 2 also shows that, even though network-driven motion estimation and conditional replenishment require the same number of operations at the encoder, network driven motion estimation greatly improves the quality of the decoded images compared with conditional replenishment. Thus network-driven motion estimation obtains the power efficiency of conditional replenishment while maintaining the high quality reconstructed images of encoder-based full search motion estimation.

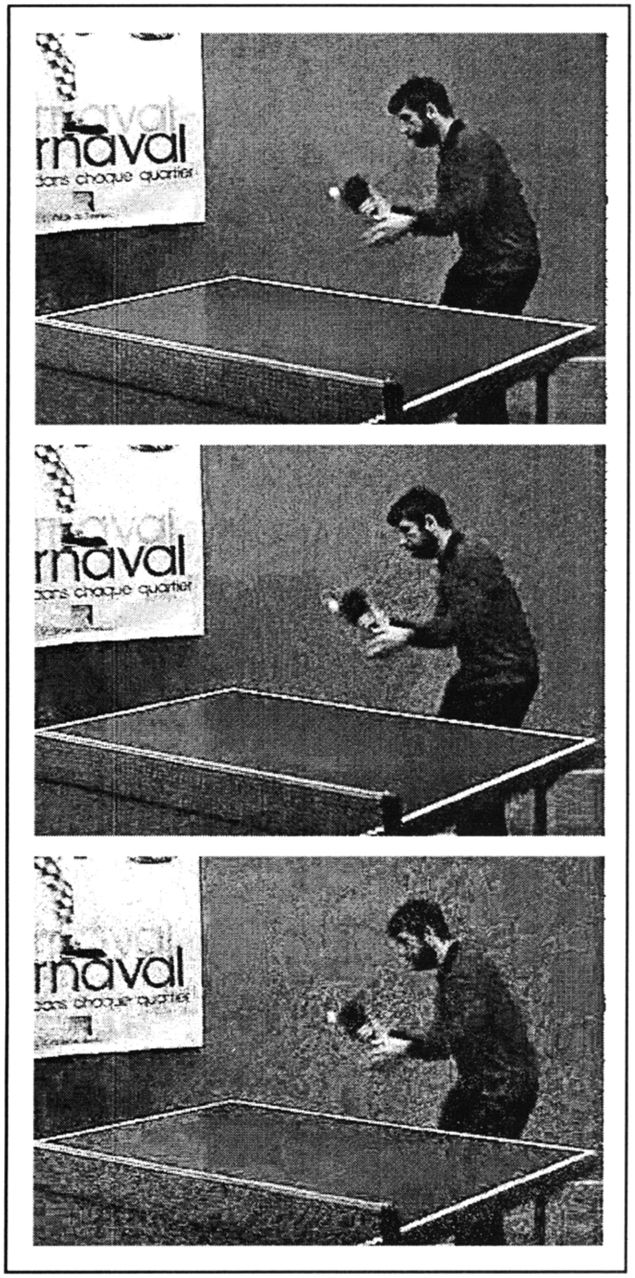

Figure 2. Tennis sequence coded with a constant bit rate. (a) Encoder-based motion estimation, SNR = 27.3 dB. (b) Network driven motion estimation, SNR = 25.4 dB. (c) Conditional replenishment, SNR = 22.3 dB.

1.22.1 Publications

Rabiner, W.B., and A.P. Chandrakasan. "Network Driven Motion Estimation for Portable Video Terminals." Paper presented at the International Conference on Audio, Speech, and Signal Processing, Munich, Germany, April 21-24, 1997.

Rabiner, W.B. *Network Driven Motion Estimation for Wireless Video Terminals.* S.M. thesis. Dept. of Electr. Eng. and Comput. Sci., MIT, 1997.