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 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 use 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 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 multiple-access (CDMA) strategies, new techniques for exploiting

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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 Neural Signal Processing

Sponsor

National Science Foundation Graduate
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Project Staff

Anthony J. Accardi, Professor Gregory W. Wornell

In order to understand the detailed interworkings of many neurological processes, it is necessary to measure the firing patterns realized by individual neurons. Current measuring techniques involve inserting one or more electrodes into the region of interest, which make extracellular voltage recordings derived from the action potentials of nearby neurons. The difficulty is that firing patterns from many different neurons are superimposed at the electrodes, but we are interested in individual neuron behavior. Deriving this information from such measurements is referred to as separating multiple single-unit spike trains from a multi-unit recording.

The problem is therefore one of signal separation, and many approaches have been attempted based on pattern matching and feature clustering. In many of these approaches, the inaccurate assumption that different neurons exhibit action potentials with unique waveforms is made. A new instrument consisting of four very closely spaced electrodes, called the tetrode was developed in 1994. This instrument allows us to drop the assumption and therefore perform a more reliable separation. The best existing separation schemes for the tetrode are computer assisted; they present waveform parameters in a graphical manner so that a well-trained user can visually cluster the feature arising from separate neurons. These techniques necessarily prevent a full exploitation of the information available in the tetrode measurements, since decisions must be made in a low enough dimension for human visualization.

We hope to improve upon these existing multi-unit separation schemes and will then pursue related applications (e.g., action potential coding) of our new knowledge.

1.3 Dual-Channel Signal Processing

Sponsors

Sanders, a Lockheed-Martin Corporation
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Project Staff

Richard J. Barron, Professor Alan V. Oppenheim

Many models for signal estimation systems assume only statistical information about the source signal to be recovered and the channel through which the source is sent. In some scenarios, however, there also exists deterministic side information about the desired signal which can be used jointly with channel observations to assist recovery. For example, an existing full-band, noisy analog communications infrastructure may be augmented by a low-bandwidth digital side channel. Our research is a study of a hybrid channel that is the composition of two channels: a noisy analog channel through which a signal source is sent unprocessed and a secondary rate-constrained digital channel. The source is processed prior to transmission through the digital channel. Using a signal processing framework for low latency and low complexity, we derive optimal encoder and receiver structures for hybrid channels.

1.4 Batch-Iterative Channel Equalization

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Project Staff

Albert Chan, Professor Gregory W. Wornell

The goal of channel equalization is to minimize the probability of error by compensating for channel distortion. One basic approach to channel equalization is to use the decision-feedback equalizer (DFE). The portion of the DFE that cancels postcursor intersymbol interference (ISI) is nonlinear; as a consequence of this, the postcursor equalizer portion does not enhance noise. By contrast, the portion of the DFE that cancels precursor ISI is linear, limiting its capabilities and leaving behind a significant amount of residual precursor ISI.

We are currently working on a simple yet effective equalizer that cancels both precursor and postcursor ISI in a nonlinear fashion, based on the iterative equalizer described in Beheshti and Wornell (1997).[4] The equalizer suppresses both intersymbol and interuser interference in spread-signature code-division multiple-access systems,[5] but we have adapted that equalizer to the single-user, high ISI, fixed-channel scenario. We have shown theoretically and in simulations that our equalizer has a lower probability of error than the DFE. In fact, at high signal-to-noise ratio (SNR), our iterative equalizer requires 2.507 dB less power to achieve the same probability of error as the DFE.

We are now investigating the applicability of our equalizer to low-ISI channels and to the estimation of the channel in addition to the data symbols.

1.5 Information Embedding and Digital Watermarking

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Project Staff

Brian Chen, Professor Gregory W. Wornell

Digital watermarking and information embedding, which are also referred to as data hiding and steganography, refer to the process of embedding one signal, called the "embedded signal" or "digital watermark," within another signal, called the "host signal." The host signal is typically a speech, audio, image, or video signal, and the embedding must be done in such a way that the host signal is not degraded unacceptably. At the same time, the digital watermark must be difficult to remove without causing significant damage to the host signal and must reliably survive common signal processing manipulations such as lossy compression, additive noise, and resampling. Applications include copyright protection, authentication, transmission of auxiliary information, and covert communication.

In our work, we are developing a general framework for designing digital watermarking systems, evaluating their performance, and understanding their fundamental performance limits. In the process we have developed a class of digital watermarking techniques called quantization index modulation, along with a convenient realization called dither modulation, that have considerable performance advantages over previously proposed methods. More information can be found in Chen and Wornell (1998 and 1999).[6]

1.6 Multiple Descriptions for Soft Memory Systems

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Stark C. Draper, Professor Gregory W. Wornell

The multiple descriptions problem is a classic question of information theory in which a data source is encoded into two data streams. Each stream of data independently describes the original source to a certain fidelity, but together the streams describe the

4 S. Beheshti and G. Wornell, "Iterative Interference Cancellation and Decoding for Spread-signature CDMA Systems," *Procedure Vehicular Technology Conference,* Phoenix, Arizona, May 1997.

5 G.W. Wornell, "Spread-Signature CDMA: Efficient Multiuser Communication in the Presence of Fading," *IEEE Trans. Info. Theory*, September 1995.

6 B. Chen and G.W. Wornell, "Dither Modulation: A New Approach to Digital Watermarking and Information Embedding," *Proceeding of SPIE: Security and Watermarking of Multimedia Contents* (part of Electronic Imaging '99), San Jose, California, January 1999, forthcoming; B. Chen and G.W. Wornell, "Digital Watermarking and Information Embedding Using Dither Modulation," *Proceeding of 1998 IEEE Second Workshop on Multimedia Signal Processing (MMSP-98)*, Redondo Beach, California, December 7-9, 1998, pp. 273-78.

source to a higher fidelity. An example is an imperfectly packetized network where two packets are transmitted.

Either or both packets may be received. We want to encode the data into the packets so that if only one packet is received the most fundamental data is extractable but, if both arrive, further refinements are also available.

As described above, this problem was originally conceived of as a transmission problem with packet drops. We are extending the use of the multiple descriptions coding paradigm to memory systems. These systems can easily trade off storage space for quality of the stored signal. They facilitate the construction of lower-fidelity representations of the source and, in general, ease the computational requirements of memory management.

1.7 Algebraic and Probabilistic Structure in Fault-Tolerant Computation

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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. In the case of computations that can be represented as operations in a group, the doctoral dissertation by Beckmann[7] 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,[8] 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 have also illustrated a method of constructing fault-tolerant finite automata using a combination of group and semigroup homomorphic mappings. In our future work, we intend to fold probabilistic models for failures and errors into the design and analysis of ABFT systems.

1.8 Low-Complexity Diversity Transmission for Fading Channels

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Nicholas J. Laneman, Professor Gregory W. Wornell

In the mobile wireless communications setting, transmission quality is severely degraded by fading—fluctuations in received signal energy induced by multipath propagation and relative motion of the transmitter and receiver. Diversity transmission via multiple transmit antennas, bandwidth expansion, or coding mitigates the effects of fading by essentially repeating information on independent realizations of the channel and averaging the received signal. This most basic form of diversity transmission in space, frequency, or time corresponds to a repetition code, and more powerful codes for fading channels have been developed over the years. Unfortunately, the

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

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

complexity of maximum-likelihood decoding precludes the use of long codes and, hence, higher orders of diversity.

Motivated by two compelling examples, spread-response precoding[9] and signal space diversity[10] systems, our research explores low-complexity diversity transmission from primarily a signal processing viewpoint. Each of these examples creates temporal diversity by transmitting the output from a linear (unitary) transformation of an uncoded symbol sequence, and neither scheme requires additional power or bandwidth. In the case of spread-response precoding, a low-complexity receiver consisting of linear equalization followed by symbol-by-symbol detection can achieve very high orders of diversity with quite good performance. We are comparing and contrasting these examples, evaluating the performance and complexity tradeoffs associated with several receiver structures, and hoping to generalize these ideas to create more efficient schemes.

1.9 Distributed Signal Processing

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Li Lee, Professor Alan V. Oppenheim

Networks of distributed processors and sensors can offer significant advantages over traditional signal processing platforms through improving resource sharing and fault tolerance. However, for networks in which resource availability changes dynamically, it is important to have a flexible execution strategy which allows algorithms to adapt to the current conditions of the network. Our research studies a number of questions arising from designing signal processing algorithms for dynamically changing networks of processors and sensors. We propose a framework in which the system is allowed to dynamically and optimally choose an execution strategy for each data block in adaptation to the computing environment. Our formulation of this strategy derives from an interesting interpretation of algorithms as similar to communications networks. A software simulation has been developed to demonstrate these ideas.

1.10 Transmit Antenna Arrays for Multiple-User Wireless Communication

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Project Staff

Michael J. Lopez, Professor Gregory W. Wornell

In wireless communication, a signal can reach a destination in different ways by reflecting off buildings and other obstacles. Multipath fading, caused by interference when two or more such rays combine at the receiver, is thus a major issue. One method of combating fading is with diversity, i.e., by inserting redundancy in different time, frequency, or spatial slots. We have been investigating spatial diversity using multiple antennas at the transmit side.

Certain strategies have been proposed in the literature when the transmitter does not know the channel properties, broadcasts to all possible receivers, or communicates with one user on a known channel. We are studying the case in which the transmitter attempts to communicate along known channels to a finite number of receivers. We have looked at some of the inherent tradeoffs involved in the problem and have calculated performance measures for a few potential algorithms. It is interesting to look at the unknown channel or broadcast scenarios as asymptotic cases of this problem when the number of receivers is large and to compare the advantages of using broadcast versus multicast strategies when the number of users is moderate.

9 G.W. Wornell, “Spread-Response Precoding for Communication over Fading Channels,” *IEEE Trans. Info. Theory*, 42(5): 488-501 (1996).

10 J. Boutros and E. Viterbo, “Signal Space Diversity: A Power- and Bandwidth-Efficient Diversity Technique for the Rayleigh Fading Channel,” *IEEE Trans. Info. Theory.* (1998).

We are also interested in opportunities to exploit other kinds of diversity and in the various duality-type relationships that exist between different kinds of diversity.

1.11 Applications of Digital Watermarking via Quantization Index Modulation

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Project Staff

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We have been exploring the application of digital watermarking to authenticating identification photos and copy protection for digital movies. Digital watermarking involves adding one signal (the watermark) to another signal, (the host) such that the watermark is difficult to remove without damaging the host.

Specifically, we have been studying quantization index modulation (QIM), a watermarking technique developed by graduate student Brian Chen and Professor Gregory W. Wornell. We have developed a bound on the bit error rate in embedding a watermark in a host subjected to Gaussian noise. Using this analytic bound as well as numerous simulations, we are studying how many bits can reliably be embedded in an ID photograph which is printed and scanned. The benefit of embedding a watermark in an ID photograph would be to prevent a forger from using a photograph of a person in a forged ID for another person.

Recently, we have also become interested in using digital watermarking for content protection. We are exploring some ways to design a secure digital video disk (DVD) player and secure digital movies which would prevent pirates from viewing movies distributed in DVD format without paying the required fee.

1.12 Efficient Digital Encoding and Estimation of Noisy Signals

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Haralabos C. Papadopoulos, Professor Gregory W. Wornell

In many applications in science and engineering, one must rely on coarsely quantized and often unreliable noisy measurements in order to accurately and reliably estimate quantities of interest. This scenario arises, for instance, in distributed wireless sensor networks where measurements made at remote sensors need to be fused at a host site in order to decipher an information-bearing signal. Resources such as bandwidth, power, and hardware are usually limited and shared across the network. Consequently, each sensor may be severely constrained in the amount of information it can communicate to the host and the complexity of the processing it can perform.

In this research, we develop a versatile framework for designing low-complexity algorithms for (1) efficient digital encoding of the measurements at each sensor and for (2) accurate signal estimation from these encodings at the host. We show that the use of a properly designed and often easily implemented control input added prior to signal quantization can significantly enhance overall system performance. In particular, efficient estimators can be constructed and used with optimized pseudo-noise, deterministic, and feedback-based control inputs, resulting in a hierarchy of practical systems with very attractive performance-complexity characteristics.

1.13 Efficient Sampling Rate Alteration Using IIR Filters

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Andrew I. Russell, Professor Alan V. Oppenheim

Multirate processing of signals is used in many applications, such as converting between differing digital formats. This research looks at the problem of sampling rate conversion and explores the use of IIR filters. We have shown that there are implementations of IIR filters in a multirate context which are computationally efficient, even though this was previously thought to be impossible. The research is also focused on improving filter design techniques for equiripple filters. Specifically, techniques are being developed for the design of very-high-order near-optimal single- and multirate filters. These filters will be more efficient than existing filters which are used for sampling rate conversion.

1.14 Linear Models for Randomized Sampling of Discrete-Time Signals

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Project Staff

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Efficient computation in digital signal processing is a practical problem that has been approached in various ways. Traditional solutions involve down-sampling the signal by a constant factor. Instead, in this project we explore the use of randomized sampling of the discrete-time signal as a method for reducing multiplications in the convolution sum. We have designed several algorithms based on randomized sampling of the input signal, the system function, and also iterative randomized sampling of the system function. In addition, we have formulated additive noise models for each algorithm which allowed linear filtering techniques to be applied.

1.15 Approximation and Autonomy in Large-Scale Signal Processing Computations

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Project Staff

Matthew J. Secor, Professor George C. Verghese, Dr. Hamid S. Nawab, Professor Alan V. Oppenheim

This research investigates issues related to the operation, dynamics, and control of networks of processors performing large-scale digital signal processing (DSP) computations. Two key aspects of our framework are (1) the introduction of approximate processing to provide flexibility in managing system resources effectively for enhanced performance and (2) the use of market-based models for the computational environment on which to build the communication and control algorithms.

In a typical scenario, an individual user partitions a large problem into task packets that can be distributed on the network. Other processors attend to these task packets as well as they can, and then approximate results are gathered back at the user's node for re-assembly into an approximate answer to the problem of interest. Associated with such a scenario are research challenges related to (1) characterizing DSP algorithms appropriately for this form of approximate processing, (2) developing partitioning and re-assembly methods that are robust to differential delays and varying levels of completion in the returned task packets, and (3) managing the communication and computation on the network so that congestion and delay instabilities are avoided. Our research focuses on the latter set of problems, dealing with the dynamics and control of the network.

Specifically, we are reformulating queuing network models and load balancing strategies for networks of processors using richer models for jobs. Our models include the ability to use approximate processing methods to trade off the resources required to process a job against the quality of the results. We have used such models to investigate the relationship between the stability of a queue and the distribution of the quality of the jobs after being processed by the queue.

Additionally, we have investigated the use of economic models and mechanisms for the efficient allocation of resources of computations. We have developed analytical models for the dynamic behavior of an existing implementation of an auction-based resource allocation system to determine the fairness and stability properties of the scheme. Some numerical pitfalls in naive approaches to the simulation of such systems have also been exposed. We are also exploring the use of game-theoretic models to model the strategic aspects of the interaction between independent participants in the computational marketplace in their attempt to acquire computational resources.

1.16 Sinusoidal Analysis Synthesis

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Project Staff

Alan J. Seefeldt, Professor Alan V. Oppenheim

The problem of separating voice from background noise or sounds in a recording is being investigated. In particular, we are examining the use of an uncorrupted facsimile of the corrupted voice as an aid in the separation process. Possible directions include using the facsimile to generate a probabalistic model from which processing is derived and applying transformations to the facsimile to make it sound like the voice imbedded in the noise. These techniques will hopefully be applied to the separation of an operatic voice from the accompanying orchestra.

1.17 Speech Enhancement with Side Information

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Project Staff

Charles K. Sestok, Professor Alan V. Oppenheim

Common approaches to noise reduction in speech signals use statistical information about the signal only. In some cases, more accurate characterizations of the clean speech exists. For example, a communications system consisting of a noisy analog channel and a low bandwidth digital channel could transmit a complete but corrupted version of the speech on the analog link and an error free side information signal on the digital link. This research examines the performance that could be achieved by employing such deterministic information about the original speech waveform.

Since spectral shaping parameters provide a good model for speech, linear prediction (LP) coefficients and zero phase impulse response coefficients were considered as side information. Experiments were performed using both sets of side information to enhance speech corrupted with additive white Gaussian noise. An approximate maximum likelihood (ML) estimator for the original speech using the LP side information and an exact ML estimator using the zero phase impulse response coefficients have been implemented. Both algorithms were able to reduce the audible noise at signal-to-noise ratios of 0 dB. Future directions for this research include comparison of the results with other noise cancellation procedures and extensions of the algorithm to single channel scenarios.

1.18 Communications Using Chaotic Systems

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Project Staff

Wade P. Torres, Professor Alan V. Oppenheim

Chaotic signals are potentially useful for spread-spectrum communications because they have noise-like characteristics, are generally broadband, and are difficult to predict. In particular, because chaotic signals are difficult to predict and noise-like, they are possibly well suited for situations in which it is desirable to have a low probability of intercept, or covert operation. Since it has been shown that chaotic systems can be synchronized through a weak, one-way coupling, many chaotic transmitter/receiver designs are possible, both in discrete-time and continuous-time. We are investigating an approach in which the information signal modulates the rate at which the chaotic transmitter system evolves. Conceptually, this can be viewed as a generalization of traditional phase- and frequency-modulation techniques. The major problem in designing this type of communication system is robust demodulation of the information signal and maintaining synchronization in the presence of noise.

1.19 Parameter Estimation for Autoregressive Gaussian-Mixture Processes

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Project Staff

Shawn M. Verbout, Professor Alan V. Oppenheim

This research addresses the problem of estimating parameters of discrete-time non-Gaussian autoregressive (AR) processes. We are considering the subclass of processes whose driving noise samples are statistically independent and identically distributed according to a Gaussian-mixture probability density function (PDF). Because the likelihood function for this problem is typical of not having bounds in the vicinity of undesirable, degenerate parameter estimates, the maximum likelihood approach cannot be used. Hence, an alternative approach has been taken in which a finite local maximum of the likelihood surface is sought. This approach, which is termed the quasimaximum likelihood (QML) approach, has been used to obtain estimates of the AR parameters as well as the means, variances, and weighting coefficients that define the Gaussian-mixture PDF. A technique for generating solutions to the QML problem has been derived using a generalized version of the expectation-maximization principle. This technique, which is referred to as the EMAX algorithm, has been applied in several illustrative cases; its performance in these cases has been favorable to that of previously proposed algorithms based on the same data model and to that of conventional least-squares techniques.

1.20 Array Processing Techniques for Broadband Mode Estimation

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Project Staff

Kathleen E. Wage, Professor Arthur B. Baggeroer

The goal of this research is twofold: (1) to develop a signal processing framework for estimating the normal mode decomposition of low-frequency, broadband underwater acoustic signals and (2) to apply this framework to analyze data from a long-range propagation experiment.

Normal modes are useful because they are an orthonormal basis derived from the wave equation. Also the associated set of mode coefficients contains valuable information about the ocean waveguide. The strong connection of modes to the propagation environment motivates their use in applications such as matched field source localization and tomography. Measurements of modal phase/group velocity perturbations are used in tomography to infer changes in the sound speed (and thereby temperature) over long ranges. In either case, success relies on the ability to accurately resolve the mode signals as well as a thorough understanding of how modes propagate in the ocean.

Previous work has focused primarily on methods for analyzing narrowband sources, but there is growing interest in using wider bandwidths, especially for tomographic applications. This research focuses explicitly on the issues associated with broadband mode estimation. Specifically, this involves designing mode filters that can accommodate variations in the modal wavenumbers and addressing the issue of mode coherence across frequency bands. While much theoretical work has been done regarding the effects of random ocean fluctuations on mode coherence, there have been few opportunities to compare theoretical predictions with experimental measurements. This study investigates the temporal and frequency coherence of broadband mode receptions using data from the Acoustic Thermometry of Ocean Climate (ATOC) experiment.

1.21 Coding for Transmission over Unknown Channels

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Project Staff

Huan Yao, Professor Gregory W. Wornell

In certain communication applications such as wireless communications the channel could be unknown and could be time variant. In this scenario, one approach is to repeatedly estimate the channel using training sequences known to the receiver, taking up time and/or frequency allowance. But this approach also becomes unsuitable when the channel is highly time variant. Another possible approach in the unknown channel scenario is for the receiver to estimate the channel and decode the transmitted message simultaneously. This class of approaches often has the shortcoming of low spectral efficiency and slow convergence in channel estimation. We are currently exploring possible approaches related to the latter case, but with specific structures built into the transmitted signal to facilitate the receiver to estimate the channel and decode the transmitted message. We are currently studying the manner in which the channel distorts the signal to find features of the signal that are robust enough to channel distortion.

1.22 Publications

1.22.1 Journal Articles

Beheshti, S., S.H. Isabelle, and G.W. Wornell. “Joint Intersymbol and Multiple-Access Interference Suppression Algorithms for CDMA Systems.” *European Trans. Telecommun., Special Issue on CDMA Techniques for Wireless Comm. Systems*. 9(5): 403-18 (1998).

Chen, B., and G.W. Wornell. “Analog Error-Correcting Codes based on Chaotic Dynamical Systems.” *IEEE Trans. Commun*. 46(7): 881-90 (1998).

Hadjicostis, C.N., and G.C. Verghese. “Structured Redundancy for Fault Tolerance in LTI State-Space Models and Petri Nets.” *Kybernetika*. Forthcoming.

Lee, L., and R. Rose. “A Frequency Warping Approach to Speaker Normalization.” *IEEE Trans. Speech Audio Process.* 6(1): 49-60 (1998).

Narula, A., M.J. Lopez, M.D. Trott, and G.W. Wornell. “Efficient Use of Side Information in Multiple-Antenna Data Transmission over Fading Channels.” *IEEE Selected Areas Commun.* (Signal Processing for Wireless Communications). 16(8): 1423-36 (1998).

Ooi, J.M., and G. W. Wornell. “Fast Iterative Coding Techniques for Feedback Channels.” *IEEE Trans. Info. Theory*. Forthcoming.

Papadopoulos, H.C., G.W. Wornell, and A.V. Oppenheim. “Signal Encoding from Noisy Measurements using Quantizers with Dynamic Bias Control.” Submitted to *IEEE Trans. Info. Theory*.

Singer, A.C., A.V. Oppenheim, and G.W. Wornell. “Detection and Estimation of Multiplexed Soliton Signals.” Submitted to *IEEE Trans. Signal Process*.

Winograd, J.M., S.H. Nawab, and A.V. Oppenheim. “FFT-based Incremental Refinement of Suboptimal Detection.” Submitted to *IEEE Trans. Signal Process*.

1.22.2 Meeting Papers Published

Barron, R.J., C.K. Sestok, and A.V. Oppenheim. "Speech Enhancement Using Spectral Envelope Side Information." Paper published in the *Proceedings of International Conference on Acoustics, Speech, and Signal Processing-98*, Seattle, Washington, May 1998.

Chen, B., and G.W. Wornell. "Digital Watermarking and Information Embedding using Dither Modulation." Paper published in the *Proceedings of the IEEE Workshop on Multimedia Signal Processing*, Redondo Beach, California, December 1998.

Laneman, J.N., and G.W. Wornell. "Robust Equalization for Spread-Response Precoding Systems." Paper published in the *Proceedings of the International Conference on Acoustics, Speech, and Signal Processing-98*, Seattle, Washington, May 1998.

Lee, L., and A.V. Oppenheim. "Distributed Signal Processing." *Proceedings of the Federated Laboratory Symp*osium, College Park, Maryland, February 1998.

Lee, L., and A.V. Oppenheim. "Distributed Signal Processing." Paper published in the *Proceedings of the International Conference on Acoustics, Speech, and Signal Processing-98*, Seattle, Washington, May 1998.

Lopez, M.J., and G.W. Wornell. "Transmitter Antenna Array Broadcasting with Side Information." Paper published in the *Proceedings of the IEEE International on Symposium Information Theory*, MIT, Cambridge, Massachusetts, August 1998.

Ooi, J.M., and G.W. Wornell. "Further Results on Fast Iterative Coding for Feedback Channels: Multiple-Access and Partial-Feedback." Paper published in the *Proceedings of the IEEE International Symposium on Information Theory,* MIT, Cambridge, Massachusetts, August 1998.

Papadopoulos, H.C., G.W. Wornell and A.V. Oppenheim. "A Class of High-Performance Low-Complexity Wireless Sensor Networks." Paper published in the *Proceedings of the 2nd Annual Advanced Telecommunications and Information Distribution Research Program Conference*, Maryland, February 1998.

Papadopoulos, H.C., G.W. Wornell, and A.V. Oppenheim. "Low-Complexity Digital Encoding Strategies for Wireless Sensor Networks." Paper published in the *Proceedings of International Conference on Acoustics, Speech, and Signal Processing-98*, Seattle, Washington, May 1998.

Papadopoulos, H.C. and G.W. Wornell. "Distributed Estimation Techniques for Wireless Sensor Networks." Paper published in the *Proceedings of Allerton Conference on Communication, Control, and Computing*, Monticello, Illinois, September 1998.

Wang, A.C., and G.W. Wornell. "Prediction and Estimation for Fractal Processes Using Multiscale State-Space Algorithms." Paper published in the *Proceedings of International Conference Acoustics, Speech, and Signal Processing-98*, Seattle, Washington, May 1998.

Wornell, G.W. "Creating and Exploiting Diversity in Wireless Systems: A Signal Processing Perspective." Paper published in the *Proceedings International Symposium on Circuits and Systems*, Monterey, California, May 1998.

1.22.3 Doctoral Dissertation

Papadopoulos, H.C. *Efficient Digital Encoding and Estimation of Noisy Signals*. Ph.D. diss. Department of Electrical Engineering and Computer Science, MIT, 1998.

1.22.4 Book

Poor, H.V., and G.W. Wornell, eds. *Wireless Communications: Signal Processing Perspectives*. Upper Saddle River, N.J.: Prentice-Hall, 1998.

1.23 Vibration-to-Electric Energy Conversion

Sponsors

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Professor Anantha P. Chandrakasan, Professor Jeffrey H. Lang, Scott E. Meninger, Rajeevan Amirtharajah, José O. Mur-Miranda

The trend in modern VLSI design towards low-power digital signal processing (DSP) and remote sensing applications creates an opportunity for the exploitation of novel energy sources. The extremely low duty cycle of such systems pushes power requirements of a source into the µW range. Self-powered systems based on harvesting ambient energy become viable alternatives, eliminating the need for batteries and creating low-maintenance, autonomous systems.

Several different ambient sources have already been exploited. These include solar, electromagnetic, radio frequency (RF), and mechanical vibration sources. With advances in microelectromechanical (MEMS) technology, it is possible to implement a self-powered system with the MEMS device acting as an electromechanical transducer in the form of a variable capacitor, with conversion governed by employing low-power digital control techniques.

By placing charge on the capacitor plates and then moving the plates apart, mechanical energy can be converted into electrical energy which can then be stored and utilized by a load. The system is depicted in Figure 1. The mechanical system is modeled as a vibration source that couples into the electrical system through the MEMS transducer. A low-power controller directs energy conversion and supplies power to the load. The controller consists of (1) a power electronics subsystem that excites the transducer through its energy conversion cycle and is optimized to minimize losses, and (2) a digital control core which generates the timing pulses that drive the gates of the power FETS in the power electronics subsystem.

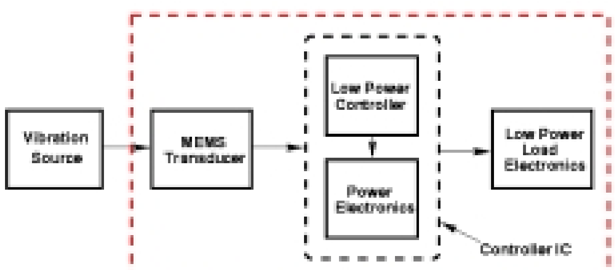

Figure 1. System block diagram

In order to validate the proposed system, a controller IC has been designed and fabricated in a 0.6 µm CMOS technology. Figure 2 on page 292 is a die photograph.

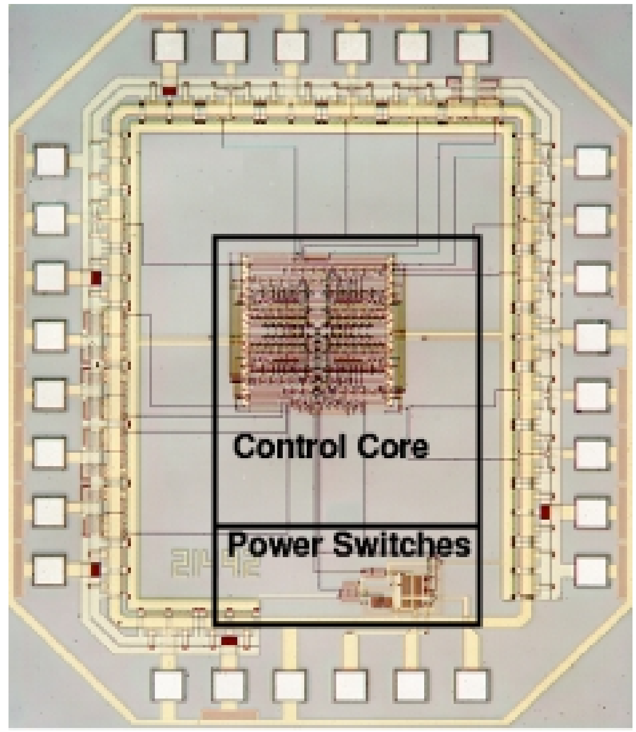

Figure 2. Controller block diagram

The controller has been verified to operate correctly and measured for losses. The MEMS device is currently being fabricated so constant value capacitors were used in its place during testing.