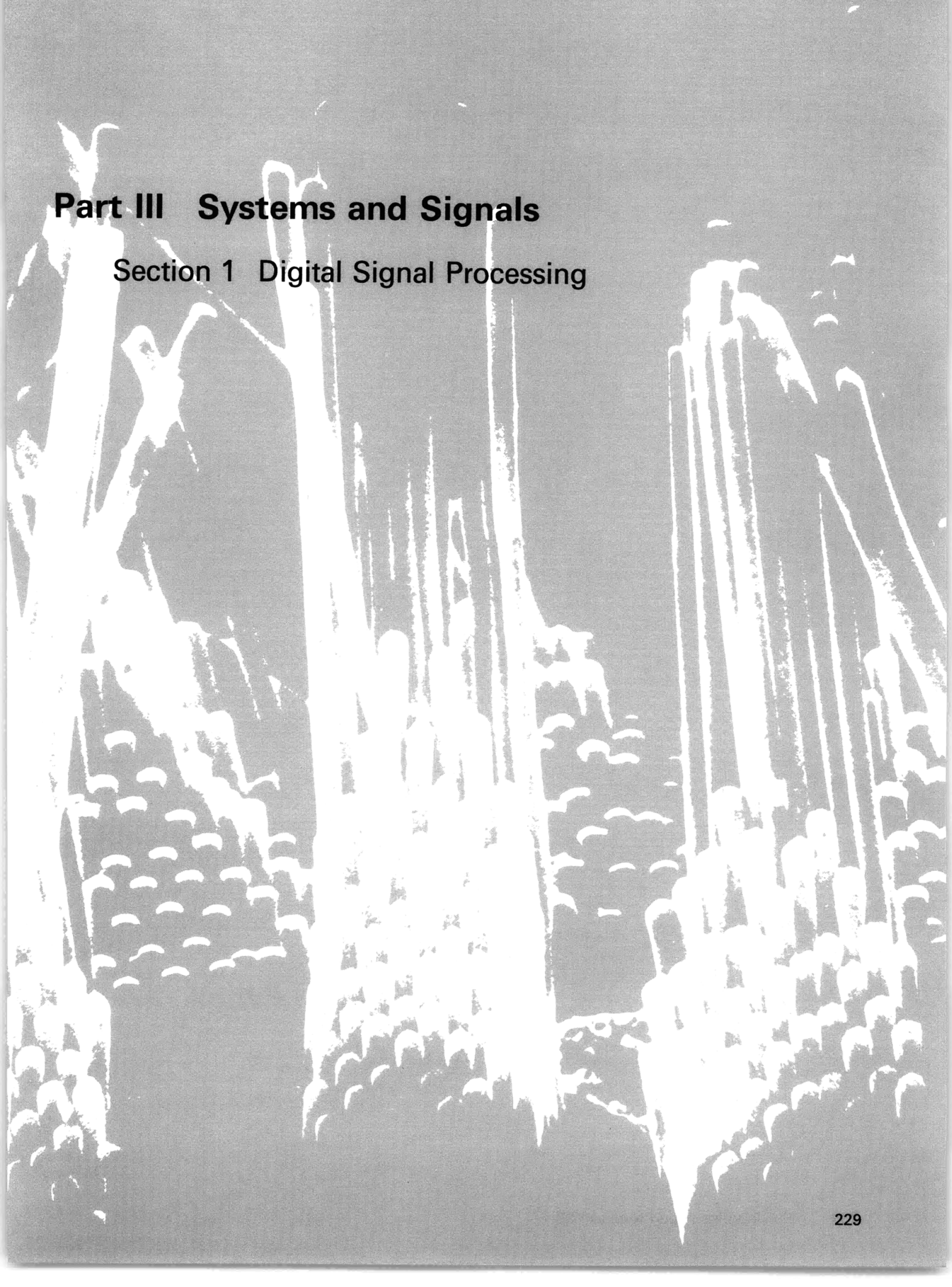
Part III Systems and Signals

Section 1 Digital Signal Processing



Section 1 Digital Signal Processing

Chapter 1 Digital Signal Processing Research Program

Chapter 2 Speech Processing Research Program

Chapter 3 Advanced Television Research Program

Chapter 4 Computer-Aided Fabrication System Structure

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Chapter 1. Digital Signal Processing Research Program

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1.1 Introduction

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

1.2 A True Maximum Likelihood Method for Directional Wave Spectra Estimation and Matched-field Source Localization

Sponsor

U.S. Navy - Office of Naval Research
Contract N00014-90-J-1544

Project Staff

Professor Arthur B. Baggeroer

Most methods of estimating directional spectra involve a step wherein the cross spectral covariance of the signal field over the array elements must be estimated. When the arrays are large and the data sparse, this estimate is singular or poorly conditioned. Several methods of mitigating this, including diagonal loading, eigenvalue

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thresholding, and subspaces have been traditionally used to circumvent these singularities. The fundamental problem is that an arbitrary covariance matrix has many more degrees of freedom than the data can constrain. A new algorithm is introduced that starts directly from the data to form an estimate of the covariance matrix that is constrained by the wave equation describing the propagation of the directional signals. It is found that a "true" maximum likelihood estimate (not a minimum variance, distortionless filter in the guise of maximum likelihood) can be specified and an iterative algorithm for implementing it can be derived. The results are similar in structure to those derived by Snyder and Miller[6] for estimating power densities by imposing a Toeplitz constraint. The algorithm can be extended to matched-field processing for localizing-independent sources. One of the advantages is that *a priori* information about the sources can be used in estimating their distribution.

These findings were reported at the 120th Meeting of the Acoustical Society of America, San Diego, California, November 26-30, 1990.

1.3 Performance Bounds on the Passive Localization of a Moving Source for Ocean Acoustics

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Project Staff

Professor Arthur B. Baggeroer, Hee Chun Song[7]

Matched-field processing (MFP), presently used for locating a point acoustic source in the ocean using a vertical array, is extended to treat a moving source problem. The extension involves both temporally nonstationary and spatially inhomogeneous structure of the sound field generated by a time-harmonic point source moving uniformly in a stratified oceanic waveguide. Using normal mode description of the sound field, the focus was on the effect of source motion on MFP. An optimum receiver based on maximum likelihood method was developed in the presence of spatially and temporally white noise. The generalized ambiguity function (GAF) was used to analyze problems of accuracy, ambiguity, and resolution. The principal result is the demonstration that a moving source problem can be treated as a stationary source problem if the source travel distance (uncompensated speed times time window) is less than half the wavelength of trapped modes. Also, a closed-form expression for the optimum potential resolution is derived based on the Cramer-Rao bound. The lower bound provides physical insight into how each mode contributes to the localization process and can be easily evaluated for a wide range of source positions in any sound channel using sound channel eigenfunctions, eigenvalues, and the number of modes involved. Simulations of GAF and the bounds for Arctic environment illustrate the coupling of ocean environment to the localization performance.

These findings were reported at the 120th Meeting of the Acoustical Society of America, San Diego, California, November 26-30, 1990.

1.4 Fault-Tolerant Algorithms and Architectures for Digital Signal Processing

Sponsors

Charles S. Draper Laboratory
Contract DL-H-404158
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Project Staff

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

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

[6] D.L. Snyder and M.L. Miller, *Proc. IEEE* 75: 892-907 (1987).

[7] MIT Department of Ocean Engineering.

Recently, an alternative method of protecting signal processing operations called Algorithm Based Fault Tolerance (ABFT) has emerged.[8] ABFT combines the design of algorithms, architectures, and fault-tolerant systems, and results in more reliable, less costly systems. The regularity of operations in DSP algorithms is exploited, and in some applications, single fault correction may be achieved with only 30%-40% overhead.

Applications of ABFT thus far examined have all been linear systems, and the data encoding and fault detection/correction techniques can be described using standard linear error correcting codes. In this research, we apply other encoding methods such as oversampling and cyclical error correcting codes. By using other encoding methods, we hope to protect a wider range of signal processing operations. Substantial progress has already been made, and ABFT has been successfully applied to two new operations: A/D conversion and convolution. Another goal of this research is to unify ABFT techniques into a general theory covering both linear and nonlinear systems.

1.5 Fault-Tolerant Round Robin A/D Converter System

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We describe a robust A/D converter system which requires much less hardware overhead than traditional modular redundancy approaches. A modest amount of oversampling is used to generate information which can be exploited to achieve fault tolerance. A generalized likelihood ratio test is used to detect the most likely failure and also to estimate the optimum signal reconstruction. The error detection and correction algorithm reduces to a simple form and requires only a slight amount of hardware overhead. We present a derivation of the algorithm followed by simulation results for both ideal and optimized FIR processing.

Publication

Beckmann, P.E., and B.R. Musicus. *Fault-Tolerant Round Robin A/D Converter.* RLE TR-561. Res. Lab. of Electron., MIT, 1990.

1.6 Implementation and Evaluation of a Dual-Sensor Time-Adaptive EM Algorithm

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Project Staff

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

Signal enhancement seeks to remove or reduce some corrupting signal or noise from a desired signal. Typical desired signals include speech or sonar. This work focuses on the implementation and the evaluation of a newly-formulated adaptive, time-domain, sequential implementation of the EM algorithm for two sensors. The algorithm assumes that the primary sensor receives the desired signal with Gaussian noise coupled in through a finite-impulse response filter. The secondary sensor receives noise with some filtered version of the desired signal. By iterating between a Kalman filter which estimates the signal and noise including the just-received data, and a maximum-likelihood parameter estimate, the algorithm converges to the uncorrupted signal while estimating the coefficients of the coupling filters.

Initial simulations indicate this sequential time-domain implementation performs comparably with Feder's block-processing frequency-domain implementation for the same problem. In addition, this implementation takes advantage of structural properties of the Kalman filter to minimize computation at each time step.

Future work will include implementing and examining a new formulation of the parameter estimation based on explicit calculation of the gradient, which should further reduce computational requirements significantly from the current maximum-likelihood parameter estimation. We also plan to examine the performance of the algorithm on more demanding "real-world" noise

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

sources, such as airplane noise, as opposed to white Gaussian noise.

1.7 Estimation and Correction of Geometric Distortions in Side-Scan Sonar Images

Sponsors

The Federative Republic of Brazil Scholarship
Lockheed Sanders, Inc.
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Grant MIP 87-14969
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Project Staff

Professor Alan V. Oppenheim, Daniel T. Cobra

This research introduces a new procedure for the enhancement of acoustic images of the bottom of the sea produced by side-scan sonar. Specifically, it addresses the problem of estimating and correcting geometric distortions frequently observed in such images as a consequence of motion instabilities of the sonar array. This procedure estimates the geometric distortions from the image itself without requiring any navigational or attitude measurements. A mathematical model for the distortions is derived from the geometry of the problem and is applied to estimates of the local degree of geometric distortion obtained by cross-correlating segments of adjacent lines of the image. The model parameters are then recursively estimated through deterministic least-squares estimation. An alternative approach based on adaptive Kalman filtering is also proposed, providing a natural framework in which *a priori* information about the array dynamics may be easily incorporated. The estimates of the parameters of the distortion model are used to rectify the image, and may also be used for estimating the attitude parameters of the array. A simulation is employed to evaluate the effectiveness of this technique, and examples of its application to high-resolution side-scan sonar images are provided.

1.7.1 Publication

Cobra, Daniel T. *Estimation and Correction of Geometric Distortions in Side-Scan Sonar Images.* RLE TR-556. Res. Lab. of Electron., MIT, 1990.

1.8 Signal Processing Applications of Chaotic Dynamical Systems

Sponsors

AT&T Bell Laboratories Doctoral Program
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Project Staff

Professor Alan V. Oppenheim, Steven H. Isabelle

Unlike linear time invariant systems which display only simple periodic or fixed point behavior under zero input, nonlinear dynamical systems display a much broader range of behavior. The realization that this complex behavior may result from even very simple nonlinear difference or differential equations has caused much excitement in the physics community. Chaotic dynamics, it is speculated, may provide a simple explanation for complicated phenomena observed in nature, such as turbulence in fluid flow and the erratic orbits of the planets. Chaos has also caused excitement in the signal processing community where it may provide a rich new set of tools for signal analysis and modeling.

In moving from linear to nonlinear signal models, entirely new classes of signal processing problems may be addressed. The goal of this research is to explore applications of nonlinear systems operating in the regime of chaotic behavior to problems of signal description. One fundamental issue to be addressed is determining when these models are appropriate. To resolve this issue, we are currently exploring techniques of detecting chaotic behavior in observed signals.

1.9 High-Resolution Direction Finding for Multidimensional Scenarios

Sponsors

Bell Northern Research Ltd.
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Project Staff

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There has been considerable interest recently in High-Resolution Techniques for direction finding

(DF) and for time series analysis. Recent results[9] have improved understanding of High-Resolution direction finding techniques in the following areas:

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

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

In practice, common applications of DF techniques to non-planar far-field and near-field problems are multidimensional in nature, requiring estimation of a vector of parameters. For example, two angular parameters are necessary in 3-D far-field problems (e.g., azimuth, elevation). Extension of 1-D approaches to multi-D is not always direct, as several high-resolution techniques, including MinNorm and minimum dimension Beamspace algorithms, fail to uniquely locate sources for multi-dimensional scenarios. This research explores the multidimensional Direction Finding problem, and extends recent 1-D results to multi-D scenarios.

1.10 Signal Processing for Ocean Acoustic Tomography

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Project Staff

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This work involves the signal processing techniques used to develop a thermal map of a 1000 km-long section of the ocean through Ocean Acoustic Tomography. Acoustic tomography is a technique in which travel time measurements over large distances in the ocean are used to infer ocean properties such as temperature and tidal motion. Large-scale (> 500 km) fluctuations of temperature in the ocean are important in determining weather, climate, ocean circulation, and the distribution of marine organisms. The temperature spectrum over horizontal distances exceeding 500 km is usually obtained from point measurements. Point measurements contain both large-scale and smaller-scale (meso and fine) spectral components. And, often, they are not obtained simultaneously in time. Tomographic measurements of large-scale signals, however, are both virtually instantaneous and integral, attenuating signals from the smaller scales.[10]

More specifically, pulse compression of the maximal length sequence signals, wide band Doppler correction to account for source-receiver motion, cross-correlation and coherent averaging to increase signal to noise ratio will be performed. To initialize the problem, multipath identification will be done using ray theory. Finally, a Kalman filter will be used to tomographically invert for the temperature profile. Using a 900 m-long vertical acoustic array, this is the first experiment in which acoustic time fronts have been observed at long range. We expect the number and the diversity of multipaths to yield high resolution.

1.11 Structure Driven Multiprocessor Compilation of DSP and Linear Algebra Problems

Sponsors

Defense Advanced Research Projects Agency
Contract N00014-87-K-0825
IBM Corporation
Sloan Foundation

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

[10] J.L. Spiesberger, "Basin-Scale Tomography: A New Tool for Studying Weather and Climate," *J. Geophys. Res.*, forthcoming.

U.S. Navy - Office of Naval Research
Grant N00014-89-J-1489

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Professor Bruce R. Musicus, Professor Anant Agarwal,[11] G.N. Srinivasa Prasanna

In this thesis,[12] we explore issues related to the automatic compilation of linear algebra and digital signal processing problems onto multiprocessors. The highly regular structure, the high computation, and the data-independent control of these problems makes them ideally suited for automatic compilation onto multiprocessors. In this research, shared memory multiprocessors like the MIT Alewife Machine[13] are targeted.

Most algorithms in signal processing and inner loops of linear algebra algorithms can be expressed in terms of expressions composed of matrix operators. The matrix operator dataflowgraphs have regular data and control flow, and regular communication structure. The nodes of the matrix expression dataflowgraph are matrix operators themselves (macro-nodes). The matrix expression dataflowgraph (also called a macro dataflowgraph), in general, exhibits little structure, but has data-independent control.

The basic compilation paradigm explored in the research is to exploit the structure in the dataflowgraphs of such numeric problems. These numeric problems can be conveniently represented as compositions of basic matrix operators. If good compilations for the basic matrix operators are known, and good techniques to compose these operator algorithms are used, then a good compilation for the numeric problem can be derived. The strategy will be fast if each of the two steps is fast. This is a hierarchical compilation strategy.

This thesis has two major parts. First, it shows that it is possible to analyze and derive good algorithms for the basic matrix operators, thus deriving a parallel operator library. Second, it shows that one can quickly compose these library routines to get good algorithms for the complete matrix expression. This yields a speedy hierarchical compilation strategy for such structured problems.

Specifically, we demonstrate how the structure present in matrix operator data flowgraphs can be used to derive close to optimal routines for them. Our techniques can be used to develop a parallel library of routines for these operators.

We have developed theoretical insights into effectively composing matrix operator algorithms to yield algorithms for the matrix expression. These insights have been obtained by viewing scheduling from the perspective of optimal control theory. Minimal time scheduling strategies can be identified with time optimal control strategies. Under certain conditions, the scheduling problem is equivalent to shortest path and flow problems. The approach has been used to derive simple heuristics for composing matrix operator algorithms to form algorithms for the expression.

We have implemented these ideas in the form of a prototype compiler producing Multilisp code from a matrix expression in Lisp-like syntax. Timings statistics on the MIT Alewife Machine[13] have been obtained that, in general, verify our ideas.

1.12 Robust Non-planewave Array Processor Development Using Minmax Design Criteria

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Project Staff

Professor Alan V. Oppenheim, James C. Preisig

Underwater acoustic array processors often must operate in environments whose characteristics are not completely specified at the time of design of the processor. These unknown characteristics may be characteristics of (1) the signal which the processor is attempting to receive, (2) the interfering signals which it also receives, (3) the environment in which the signal propagates, or (4) the array geometry.

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[11] MIT Department of Electrical Engineering and Computer Science.

[12] G.N.S. Prasanna, *Structure Driven Scheduling of Linear Algebra and Digital Signal Processing Problems,* Ph.D. diss. proposal, Dept. of Electr. Eng. and Comput. Sci., MIT, 1988; M.M. Covell, *Representation and Manipulation of Signal Processing Knowledge and Expressions,* Ph.D. diss. proposal, Dept. of Electr. Eng. and Comput. Sci., MIT, 1987; C.S. Myers, *Signal Representation for Symbolic and Numerical Processing,* Ph.D. diss., Dept. of Electr. Eng. and Comput. Sci., MIT, 1986.

[13] A. Agarwal, "Overview of the Alewife Project," Alewife Systems Memo, Lab. for Comput. Sci., MIT, 1990.

This research focuses on developing efficient array processors which are robust with respect to these kinds of uncertainties. For some applications, such as far-field beamforming in a known homogeneous propagation environment, there are established efficient algorithms for developing minmax beamformers which are robust with respect to the location of interfering signal sources. Unfortunately, many underwater acoustic array processing problems require detection and filtering or estimation of parameters, that, associated with signals propagating through inhomogeneous media, are poorly modeled as propagating planewaves. Novel formulations have been proposed for some of these problems, the solutions of which are optimal array processors with respect to minmax design criteria and which are robust with respect to specified uncertainties. Mathematically, these problems are posed as complex Chebyshev approximation problems on multidimensional spaces. The available algorithms to solve these problems are inefficient and limit the practical applicability of the resulting array processors. Efficient methods of finding optimal and near-optimal solutions to these problems are being explored, and the salient characteristics and the performance capabilities of the resulting array processors are being investigated.

1.13 Shadowing and Noise Reduction in Chaotic Systems

Sponsors

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Fellowship and Grant AFOSR-91-0034
U.S. Navy - Office of Naval Research
Grant N00014-89-J-1489

Project Staff

Professor Alan V. Oppenheim, Michael D. Richard

Chaotic systems are nonlinear dynamical systems characterized by extreme sensitivity to initial conditions. A signal generated by a chaotic system may appear random, despite its having been generated by a low-order, deterministic dynamical system. Both random and chaotic signals lack long-term predictability; but, in contrast to truly random signals, chaotic signals exhibit short-term predictability. Evidence of chaotic behavior has been reported in many diverse disciplines, including physics, biology, engineering, and economics.

An interesting research question arises when simulating a chaotic orbit by iterating a set of difference equations. Specifically, how quickly does the simulated orbit deviate from the actual orbit with the same initial conditions, because of noise induced by round-off errors and finite precision arithmetic – so-called dynamical noise—at each iteration? Also, does an actual orbit, perhaps with different initial conditions, stay close to or shadow the simulated orbit? A related question that arises when measuring a chaotic signal is whether or not knowledge that the signal is chaotic can be exploited to reduce noise due to measurement errors—so-called observational noise.

This research is (1) exploring techniques to reduce noise, both dynamical and observational, in chaotic systems and (2) establishing the utility of these techniques to typical noise-reduction problems in signal processing. An iterative technique for noise reduction, which was proposed in the literature, was implemented and is being studied. This technique was derived from a mathematical proof of the Shadowing Lemma which established the existence of actual, shadowing orbits to noisy, simulated orbits for a certain class of non-chaotic dynamical systems. The use of an Estimate-Maximize (EM) algorithm for noise reduction is also being considered. Finally, the value of various noise reduction techniques in improving the short-term predictability of chaotic signals will be ascertained.

1.14 Causal Filters with Negative Group Delay

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Project Staff

Professor Alan V. Oppenheim, Stephen F. Scherock

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

Currently, the scope of the research is restricted to discrete time FIR filters. Several design algorithms have been successfully implemented, and new error measures have been developed to judge the quality of the filters. The current algorithms are fast and direct (not iterative), but are impractical for large filters. Future work will explore new algorithms, and algorithms for FIR filters.

1.15 Signal Prediction Based on Nonlinear and Chaotic System Models

Sponsors

Lockheed Sanders, Inc.
U.S. Navy - Office of Naval Research
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Project Staff

Professor Alan V. Oppenheim, Gregory W. Wornell, Andrew C. Singer

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

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

1.16 Signal Enhancement Using Single and Multisensor Measurements

Sponsors

Lockheed Sanders, Inc.
U.S. Navy - Office of Naval Research
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Project Staff

Professor Alan V. Oppenheim, Dr. Meir Feder, Dr. Ehud Weinstein

A time-domain approach to signal enhancement based on single and multiple sensor measurements is developed. The procedure is based on the iterative Estimate-Maximize (EM) algorithm for maximum likelihood estimation. On each iteration, in the M step of the algorithm, parameter values are estimated based on the signal estimates obtained in the E step of the prior iteration. The E step is then applied using these parameter estimates to obtain a refined estimate of the signal. In our formulation, the E step is implemented in the time domain using a Kalman smoother. This enables us to avoid many of the computational and conceptual difficulties with prior frequency domain formulations. Furthermore, the time domain formulation leads naturally to a time-adaptive algorithm by replacing the Kalman smoother with a Kalman filter. In place of successive iterations on each data block, the algorithm proceeds sequentially through the data with exponential weighting applied to permit the algorithm to adapt to changes in the structure of the data. A particularly efficient implementation of the time-adaptive algorithm is formulated for both the single- and the two-sensor cases by exploiting the structure of the Kalman filtering equations. In addition, an approach to avoiding matrix inversions in the modified M step is proposed based on gradient search techniques.

Publication

Weinstein, E., A.V. Oppenheim, and M. Feder. *Signal Ehancement Using Single and Multi-Sensor Measurements.* RLE TR-560, Res. Lab. of Electron., MIT, 1990.

1.17 Synthesis, Analysis, and Processing of Fractal Signals

Sponsors

Canada, Natural Science and Engineering Research Council Scholarship
U.S. Air Force - Office of Scientific Research Grant AFOSR-91-0034
U.S. Navy - Office of Naval Research Grant N00014-89-J-1489

Project Staff

Professor Alan V. Oppenheim, Gregory W. Wornell

Recently, we have developed a novel and highly useful framework for addressing a broad class of signal processing problems involving fractal signals and systems. This framework is based upon an efficient wavelet-based expansion for the 1/f family of fractal processes in terms of uncorrelated coefficients.[14] Since 1/f processes are inherently well-suited for modeling a wide range of natural and manmade phenomena, there are many potentially important applications of this work.

Subsequent work has focused on exploiting this framework in the solution of some rather general signal processing problems involving these processes. For example, in Wornell and Oppenheim,[15] we consider the problem of parameter estimation and signal estimation (smoothing) for fractal signals embedded in white noise. The whitening and discretization achieved by the wavelet representation leads to a highly tractable analysis and computationally efficient algorithms.

Continuing this work, we have been studying the related problems of prediction and interpolation of fractal processes. We have also considered the extension of this work to more general classes of fractal processes.

Additional work has addressed a number of general detection problems involving signals in fractal backgrounds. Here we consider both coherent and incoherent detection scenarios. This work has provided a foundation for our more recent work in developing signaling strategies for communication over fractal channels[16] and in designing signals for low probability of detection in fractal backgrounds.

1.18 Active Noise Cancellation

Sponsors

Texas Instruments, Inc.
U.S. Navy - Office of Naval Research Grant N00014-89-J-1489

Project Staff

Professor Alan V. Oppenheim, Kambiz C. Zangi

Unwanted acoustic noise is a by-product of many industrial processes and systems. With active noise cancellation (ANC), one introduces secondary noise sources to generate an acoustic field that interferes destructively with the unwanted noise to eliminate it. Examples of unwanted noise include: machinery, aircraft cabin, and fan noise.

Traditional active noise cancellation systems assume that the statistical characteristic of the primary noise is known *a priori.* Furthermore, almost all of the existing systems use two microphones, and, as a result, suffer from an acoustic feedback between the cancelling speaker and the input microphone.[17]

We are currently studying an active noise cancellation system which uses only one microphone and therefore has no feedback problem. In addition, we are looking at various algorithms to adapt the system to the time-variations of the primary noise source. To this end, we have been able to use the estimate maximizing algorithm to find the maximum likelihood estimate of the time-varying noise statistics.[18]

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

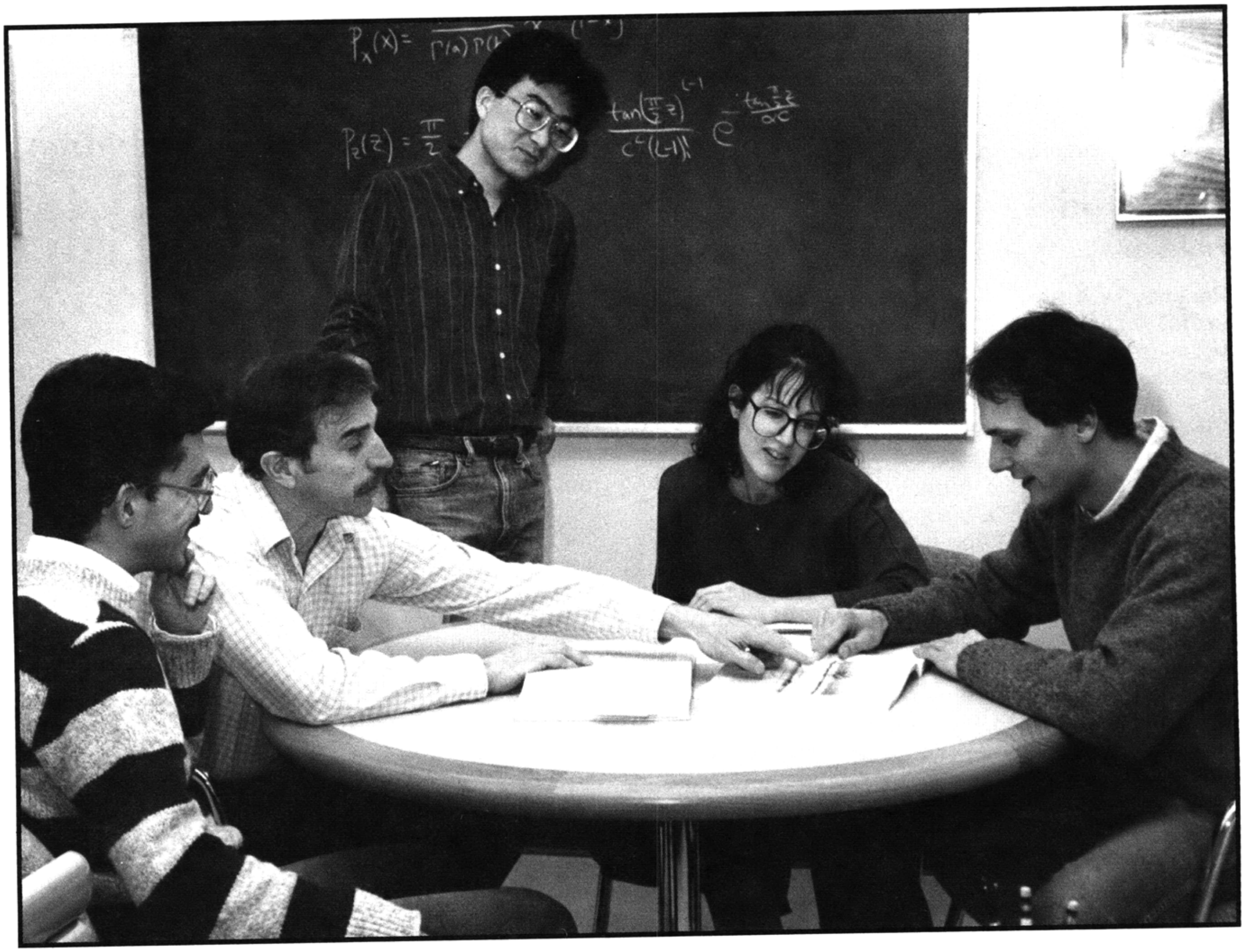
[15] G.W. Wornell and A.V. Oppenheim, "Estimation of Fractal Signals from Noisy Measurements Using Wavelets," *IEEE Trans. SP,* forthcoming.

[16] G.W. Wornell, "Communication over Fractal Channels," *Proc. ICASSP* (1991), forthcoming.

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

[18] M. Feder, A. Oppenheim, and E. Weinstein, "Methods for Noise Cancellation Based on the EM Algorithm," *Proc. ICASSP* 1987, pp. 201-204.

A problem of immediate interest is to develop similar algorithms with faster rates of convergence. We are also working on algorithms with better computational efficiency.[19]



Professor Alan V. Oppenheim with his graduate students. From left, Daniel T. Cobra, Professor Oppenheim, Tae H. Joo, Michele M. Covell, and Gregory W. Wornell.

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19 E. Weinstein, A. Oppenheim, and M. Feder, *Signal Enhancement Using Single and Multi-Sensor Measurements*, RLE TR-560. Res. Lab. of Electron., MIT, 1990.