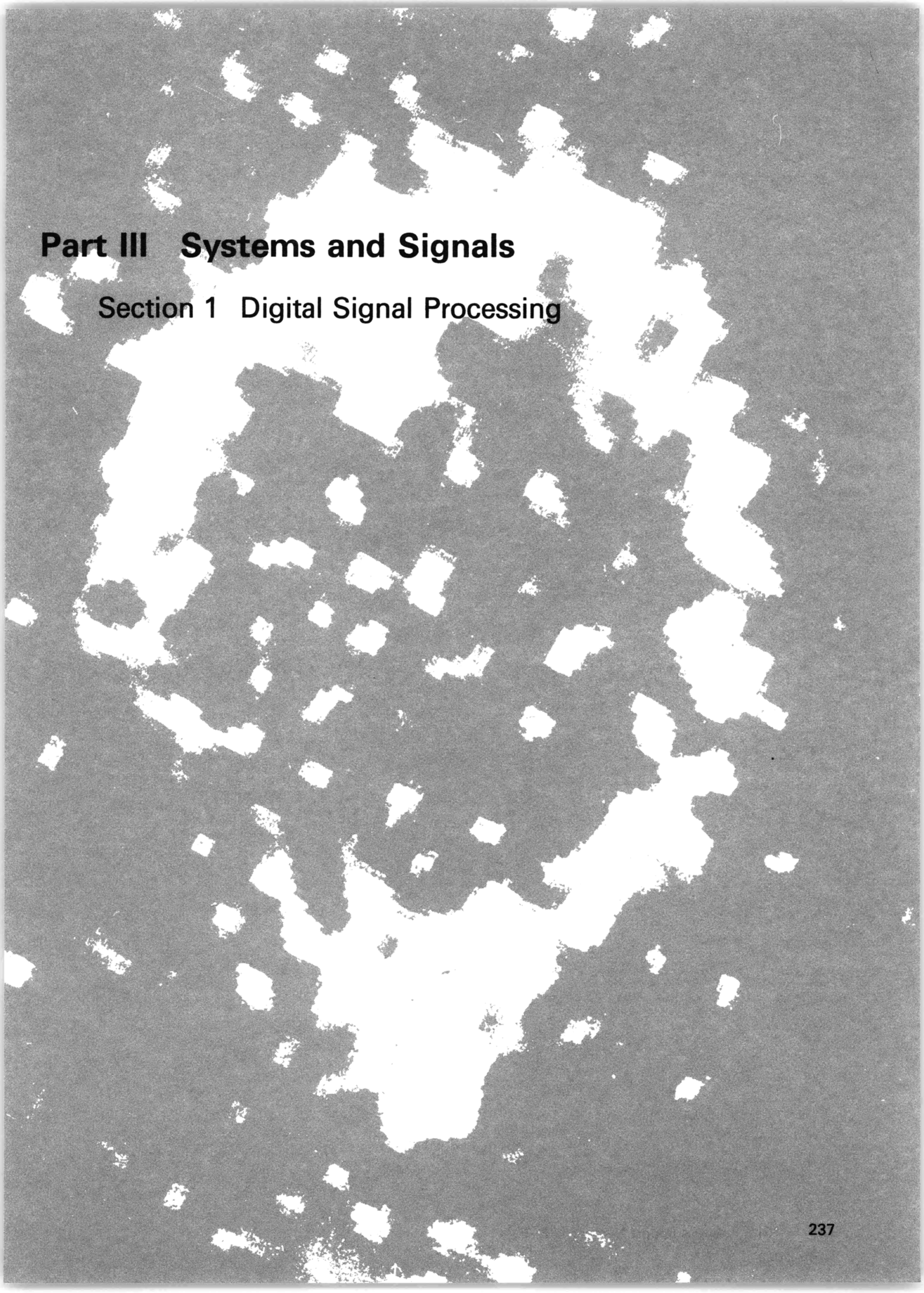
# Part III Systems and Signals

## Section 1 Digital Signal Processing



# Section 1 Digital Signal Processing

# Chapter 1. Signal Processing Research Program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## 1.1 Introduction

The Signal Processing Research Program develops new algorithms and their applications in a variety of areas including speech, image, and underwater acoustic signal processing. In addition, the program deals with issues related to algorithm implementation. We are following a new direction in signal processing, which we refer to as knowledge-based signal processing. While, historically, signal processing has principally emphasized numerical techniques, knowledge-based processing involves a combination of numerical and symbolic processing techniques.

In addition to working on specific on-campus projects, our group interacts closely with MIT Lincoln Laboratory and Woods Hole Oceanographic Institution.

## 1.2 Algorithmic Fault Tolerance in Digital Signal Processing


### Sponsors

Charles S. Draper Laboratory
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Conventional methods for achieving fault tolerant computer architectures rely on triplicating computational resources and using voter circuitry to reject incorrectly computed results. From an information theory viewpoint, however, using coding techniques would achieve a high level of error recovery with minimal overhead. Essentially, a coder distributes information across a noisy channel bandwidth, so that individual noise spikes may destroy a portion of many bits, but not an entire bit. A decoder at the receiver could combine information from the full channel bandwidth to reconstruct the original message with a very high degree of reliability.

High-performance memory systems and communication networks use coding techniques heavily. These techniques work well to protect modules in which data enters at one end and is expected to arrive intact and unchanged at the other. However, coding techniques are not usually used to protect actual computation. Instead, high levels of fault tolerance within CPUs are traditionally achieved by duplicating or triplicating processor resources and voting on the results. Also, the coding and decoding procedure adds to the latency of the channel, slowing down any machine using the protected component.

We are developing a new approach to fault tolerance that protects certain types of linear computation against processor failure. In this approach, a small number of redundant processors protect each other and the "real" processors. One design employs a bank of analog-to-digital converters operating in round-robin fashion to achieve an overall sampling rate somewhat above the Nyquist rate for the signal. A dither system and digital low-pass filter combine to reduce quantization errors in the front end. We can use this same low-pass, however, to detect and correct temporary or permanent errors in any of the converters without substantially increasing the total amount of computation time. The system trades off additional hardware for greater accuracy and higher levels of fault protection. When converters fail, an increase in the effective quantization error is the only result.

Another application of algorithmic fault tolerance is the FFT processor system used in range and velocity doppler sonar processing. In this processor system, we use a stack of processors to process multiple scans of sonar data from a phased-array antenna. Each processor does the same linear FFT processing, but on different sets of range cells. Adding extra processors working on linear combinations of the inputs to the other processors allows simple fault detection and correction. Regardless of the number of processors in the system, detecting K simultaneous failures requires only K extra processors; detecting and correcting K simultaneous failures requires only 2 K extra processors. When conventional truncation or rounding arithmetic is used, however, then the error checking is only approximate.

In this case, adding more processors improves the accuracy of the fault checking and correction. We use generalized likelihood ratio tests to select the most likely failure hypothesis and to perform the most likely fault correction. Realistic systems result, using comparatively small computational overhead (<50 percent) to achieve 100 percent single fault detection and correction.

We are presently working with Draper Laboratories to design a prototype hardware system incorporating 16 DSP-32C processors and a 68000 controller for testing a sonar application of this theory. We are also currently trying to extend the idea to cover nonlinear computation.

### *Publications*

Beckmann, P., and B.R. Musicus. "Fault-Tolerant Round Robin A/D Converter System." *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, April 1990.

Song, W.S. *A Fault-Tolerant Multiprocessor Architecture for Digital Signal Processing Applications.* Ph.D. diss. Dept. of Electr. Eng. and Comput. Sci., MIT, 1989.

## 1.3 Active Noise Cancellation Using the EM Algorithm

**Sponsors**

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**Project Staff**

Professor Alan V. Oppenheim, John R. Buck

Active noise cancellation seeks to eliminate or reduce unwanted noise by creating a signal which will destructively interfere with the noise. Active cancellation is especially attractive for low-frequency noise, which is particularly resistant to traditional passive methods. Problems in active noise cancellation fall into two categories: removing noise from a desired signal and cancelling noise in an environment. This work will focus primarily on the former problem, although it has applications to the latter as well.

Specifically, the research will further investigate an adaptive, recursive time-domain implementation of the EM (Estimate-Maximize) algorithm for removing additive noise from a corrupted signal. This research focuses primarly on the two-microphone case. One microphone receives primarily speech with some filtered noise component, while the other receives primarily noise with some filtered speech components. With each additional new sample of information, the algorithm attempts to refine its estimate of the speech and noise signals, and then uses this estimate to further estimate the filters through which the speech and noise are passed through before reaching the microphone. By repeating this process, the algorithm should converge to the original speech, the filter coefficients, and the variance of the noise.

After initial investigation of the algorithm, the research will pursue a streamlined version that removes much of the unnecessary matrix manipulation in the original description of the algorithm. We will utilize known structural properties of the matrices used by the algorithm in its successive estimations.

### 1.3.1 Iterative Algorithms for Stochastic Estimation

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**Project Staff**

Professor Bruce R. Musicus, Dr. Meir Feder, Dr. Ehud Weinstein

Maximum Likelihood (ML) is a well-known technique for generating asymptotically efficient estimates of unknown system parameters using noisy and incomplete observations. Classic ML algorithms include Wiener and Kalman filtering for estimating signals from noisy observations and auto-regressive, moving-average algorithms for estimating pole-zero models from clean data.

Unfortunately, when the signal model has unknown parameters and the observations are corrupted by noise, the optimal ML algorithm is often difficult to compute. We have been working on a variety of iterative algorithms for solving these difficult ML problems, similar to the Estimate-Maximize (EM) algorithm. The key idea is to decouple the estimation of the unknown internal signal from the estimation of the unknown parameters. We iteratively estimate the signal, use the signal to estimate the parameters, and then use the parameters to build a better filter for estimating the signal. Each step is quite similar to a classical ML filter or parameter estimation calculation, and convergence can be guaranteed to a stationary point of the likelihood function.

We have explored in depth the time-delay estimation on an array of receivers and with a single source. Our algorithms iteratively estimate the time of arrival at each sensor, along with the attenuation of the signal at each sensor, the signal power spectrum, the noise gain at each sensor, and the signal itself. After developing a novel convergence rate analysis for this EM algorithm, we used it to develop hybrid EM-ML algorithms capable of achieving superlinear convergence in some or all of the parameters with only a modest boost in computational effort. Extensive simulation has verified the performace of our algorithms. Future work will extend the technique to multiple signals and the convergence analysis to cover other EM algorithms, such as pole-zero estimation.

### Publications

Liou, C.-Y., and B.R. Musicus. "A Separable Cross-Entropy Approach to Power Spectral Estimation." *IEEE Trans. Acoustics, Speech Signal Proc.* 38(1): 105-113 (1990).

Musicus, B., and E. Weinstein. *Iterative Maximum Likelihood Time Delay and Doppler Estimation Using Stationary Signals.* RLE Technical Report No. 545. MIT, 1989.

Segal, M., E. Weinstein, and B.R. Musicus. "Estimate-Maximize Algorithms for Multi-Channel Time Delay and Signal Estimation." *IEEE Trans. Acoustics, Speech Signal Proc.* (1989), forthcoming.

## 1.4 Iterative Maximum Likelihood Time Delay and Doppler Estimation Using Stationary Signals

#### Sponsors

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#### Project Staff

Professor Bruce R. Musicus, Dr. Ehud Weinstein

We develop computationally efficient iterative algorithms for the joint Maximum Likelihood (ML) estimation of the time delays, Doppler shifts, and spectral parameters of stationary Gaussian signals radiated from a stationary or moving point source and observed in the presence of uncorrelated additive noise at two or more spatially distributed receivers. A particularly important feature of these algorithms is that they decompose the estimation of the signal spectral parameters from the estimation of the delay and Doppler parameters, leading to a considerable simplification in estimator structure and computation. The proposed algorithms converge to the set of stationary points of the likelihood function, and each iteration increases the likelihood. Because all algorithms are derived from a common iterative framework related to the Estimate-Maximize algorithm, we analyze their convergence rates both theoretically and via simulation.

### Publication

Musicus, B., and E. Weinstein. *Iterative Maximum Likelihood Time Delay and Doppler Estimation Using Stationary Signals.* RLE Technical Report No. 545. MIT, 1989.

### 1.4.1 Research in Digital Signal Processing

#### Project Staff

Dr. Charles S. Burrus

This research has been in two areas: digital filter design and efficient DFT and convolution algorithms. In the first area, we have developed a new method for the least squared error design of linear phase FIR filters. This method introduces a spline transition band in the ideal frequency response so that we can derive an analytical design formula that is as simple as the Window method, yet which retains optimality and explicit control over the band edges. We have developed a numerical method that allows error weighting which analytical methods cannot achieve. In the area of algorithms, the use of tensor and Kronecker product formulation of the DFT shows some promise for implementation on parallel and vector computer architectures.

A paper on the FIR filter design method will be published in *IEEE Transactions on Acoustics, Speech and Signal Processing,* and another will be presented at the 1990 International Conference on Acoustics, Speech and Signal Processing (ICASSP). A paper on tensor formulations of the FFT will also be presented at ICASSP '90.

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

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**Project Staff**

Professor Alan V. Oppenheim, Daniel T. Cobra

Since its introduction in the early 60s, side-scan sonar has proved to be an important tool for underwater exploration and, in particular, for marine geological research. Its applications include surveying the seafloor, searching for and locating objects on the bottom of the sea, and prospecting for deepsea mineral deposits.

The information contained in reflected sound waves is used by side-scan sonars to produce acoustic images, called sonographs. These sonographs constitute a composite representation of the topographic features and the relative reflectivity of the various materials on the seabed. However, sonographs do not precisely depict seafloor topology. The sonar image can suffer from radiometric interferences such as those caused by dense particle suspension in the water, shoals of fish, or ultrasonic waves generated by passing ships. Large-scale geometric distortions may be caused by deviations in the ship's trajectory from the ideal straight path. Small-scale distortions may be caused by motion instability of the towed body on which the transducers are mounted because of underwater currents or ship sway. As a result, the interpretation of sonographs often requires extensive practice and can be a tedious and time-consuming task.

We have successfully corrected radiometric distortions and large-scale geometric distortions through standard image-processing techniques. Using digital post-processing of sonographs, our goal is to develop techniques for detecting and correcting small-scale geometric distortions due to sonar motion instabilities. Previously, there has been inadequate information available in the literature. We are exploiting the cross-correlation between segments of adjacent image lines as a measure of the local degree of geometric distortion in sonographs. With a mathematical model derived from the geometry of side-scan sonars, we then employ these measures of geometric distortion to estimate the attitude parameters of the sonar array. Finally, the geometric distortions are corrected by resampling the image to compensate for the estimated motion instabilities of the array.

This project is being conducted under MIT's Joint Program with the Woods Hole Oceanographic Institution and with the cooperation of the U.S. Geological Survey.

## 1.6 An Algorithm Design Environment for Signal Processing

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**Project Staff**

Professor Alan V. Oppenheim, Michele M. Covell

Signal processing uses the computer primarily for numerical calculations. Recently, a number of signal-processing environments have evolved which simplify the initial creation of a signal-processing algorithm from a series of low-level signal-processing blocks. Despite this progress, the majority of the design process is generally completed without computer support: Analyses of the properties of the selected algorithm are generally completed by hand, as is the manipulation of the algorithm to find efficient, input/output equivalent implementations. This study explores some of the issues involved in providing software tools for the symbolic analysis and rearrangement of signal-processing algorithms as well as for the initial algorithm selection.

We have developed a software environment that supports numeric signal-processing computations as well as the symbolic analysis and manipulation of signal-processing expressions. This study is primarily involved with the symbolic manipulation of signal-processing expressions. To allow for the efficient manipulation of a variety of "regular" algorithms such as polyphase and FFT structures, we introduce correspondence constraints and use them to guide the rearrangement of these structures. We develop detailed cost descriptors to allow the environment to accurately compare the costs of the various equivalent implementations. We then use these comparisons to reduce the number of implementations presented to the user by removing the uncomputable and computationally inefficient forms.

We have demonstrated the potential of constrained algorithm manipulation with two examples. First, we considered briefly the problem of noninteger sampling rate conversion. Second, we explored in detail the more complex problem of detecting and discriminating FSK codes in sonar returns. Thirdly, we used an example on the recovery of in-phase and quadrature samples of an RF signal to highlight some of the limitations of the design tools developed in this study, which was completed in December 1989.

## 1.7 Compiling Signal Processing Algorithms into Architectures


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### Project Staff

Professor Bruce R. Musicus, G.N. Srinivasa Prasanna, Dennis C. Fogg

We are currently studying several important and difficult problems which arise when compiling signal processing algorithms into either prespecified architectures or into custom hardware designs. In general, the problem of optimized high-level mapping of algorithms into flexibly defined architectures is extraordinarily difficult, requiring a search over an enormous algorithmic and architectural design space and matching arbitrarily structured algorithms onto arbitrary architectures. Focusing on signal processing has several advantages including: (1) a large literature on efficient algorithms for many signal processing problems is available, and (2) often there are clear strategies for converting a signal processing calculation into a variety of alternative forms that may be better suited for particular implementations. These algorithms also typically rely on a large amount of matrix algebra and are, therefore, composed of high-level modules with regular computation, data access, and control structure. Often, much of the branch control inside these high-level modules can be precompiled or otherwise anticipated, thus allowing for highly efficient pipelining and parallelism. Because we can rely on static analysis of the program, effective optimizing compilers and optimizing Computer-Aided Design packages can be developed. We can use special techniques which exploit knowledge of signal processing to reorganize the computation into appropriate large-scale modules and map the large regular computation modules onto specialized hardware. Finally, the real-time constraints and huge computational burden often associated with signal processing applications such as radar, sonar, speech or image processing, often justify investment in special purpose, parallel and pipelined systems.

Our efforts have focused on a small set of issues whose solution is critical to the development of high-level algorithm compilers. In one project, we are writing a parallel algorithm expert for doing matrix computations and Fast Fourier Transforms (FFT) on linear or general arrays of processors. Because these computations are highly-structured, we can prove theorems setting lower bounds on the computation time for a given architecture with fixed computation, communication, and memory bandwidths. Furthermore, we can derive optimal algorithmic transformations of these problems for a given ratio of computation speed to memory speed to communication speed. Systematic sweep strategies based on Critical Path Method (CPM) scheduling can achieve execution times very close to the optimal bounds. We have written a compiler in Multi-LISP which incorporates our strategies, and are testing the code on an Encore Multimax processor and on a new computer architecture being developed by Professor Anant Agarwal.

Our second focus is on mapping a given dataflow graph into custom VLSI. We are attempting large numbers of possible designs for solving the given problem, rating each according to multiple performance objectives. The result is a scatter plot illustrating the achievable system design tradeoffs between speed and cost. The human designer can then choose a particular performance range for further exploration. The architectural exploration proceeds in two phases. In the first, the system proposes a variety of possible hardware block diagrams for solving the problem. In the second phase, we assign pipelining and instantiate the modules from a library of possible parts, combining decoupled design techniques with sample search methods and heuristics for pruning the search space to reduce the complexity of searching through the design space. We have also found a fast linear programming algorithm for solving approximately for the optimal system tradeoff of speed versus area. We have extended this algorithm into a mixed integer-linear programming technique that simultaneously chooses components and places pipeline registers. Our current efforts focus on enriching the user interface, allowing dynamic exploration of the design space, and developing good heuristically-limited enumeration techniques to restrict the search through the design space and guide the linear programming solution.

Graduate students Jim Olsen, Kevin Peterson and Tom Dennedy, who are not members of the Signal Processing Research Program, are also involved in this study, but are not funded by the above-listed sponsors.

## 1.8 Vector Quantization with Adaptive Structured Codebooks

**Sponsors**

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**Project Staff**

Professor Alan V. Oppenheim, Jacek Jachner

In this research project, we investigate a class of adaptive structured codebooks for Vector Quantization (VQ), for which both codebook design and codeword matching are sufficiently fast to be performed on-line. Recent work in low bit rate speech coding has applied VQ to code the residual signal after both short-term linear prediction (LPC) and long-term (pitch) prediction. For example, in Code Excited Linear Prediction (CELP) coders, each block of residual signal is matched with one codeword vector from a codebook set to minimize a linearly weighted mean square error (WMSE) criterion. The noise weighing criterion is time-varying with the LPC model of short-term speech characteristics.

The performance of a codebook depends on its design, or the choice of codewords to populate it, and on the error criterion. Performance-optimal design requires complex clustering techniques not suitable for on-line computation. Since optimality depends on the error criterion, the CELP coder would require time-varying codebooks for optimal performance. As multiple codebook storage is not practical, a non-optimal fixed codebook is commonly used instead. The designs proposed here are also not optimal, but may be simply adapted to the time-varying criterion to enhance performance relative to fixed codebook systems.

Our proposed design approach is to optimize a structurally constrained codebook for each time-varying error criterion. A rotationally symmetrical distribution is assumed for the prediction residual. In the Karhunen-Loeve transform domain of the error criterion weighing matrix, we use adaptive bit allocation (ABA) to assign levels to sample-by-sample quantizers for the high-energy coefficients, representing the remainder of the coefficients by two-level quantizers. The assignment of sign for each codeword coefficient is controlled by a binary error-correcting code. This design provides a continuous range of good performance from binary error-correcting codes suitable for unweighted MSE, to simple adaptive bit allocation when only a few coefficients are dominant.

The complexity of codeword matching depends on the structural properties of the codebook; the proposed design allows fast matching. Both optimal and random codebooks (which are designed by random selection), require a computationally intensive search to match the speech residual to one of the codewords. Known structured codebooks based on trees or lattices accelerate the computation, but do not simply adapt to time-varying error criteria. For the proposed codebook structure, we achieve efficient codeword search by a novel combination of fast methods for soft-decision decoding of binary error-correcting codes with the sample-by-sample independent property of the multi-level quantizers.

Computer simulation is currently being used to evaluate the performance of the proposed technique.

### 1.8.1 Detection Statistics for Multichannel Data

**Sponsors**

Amoco Foundation Fellowship
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**Project Staff**

Professor Alan V. Oppenheim, Tae H. Joo

In our research, we have developed a new detection statistic for a class of multichannel detection problems for which, in the presence of an emitter, a narrowband signal exists in all channels in addition to wideband noise. When an emitter is absent, the received data may contain narrowband noise components in some, but not all of the channels, as well as wideband noise. A detector which tests each channel separately for the existence of the narrowband component does not perform as well as the detectors which use all channels collectively.

To collectively use the data from different channels, the average has been previously used as a detection statistic. However, because the average only tests the total energy, its detection performance noticeably degrades when a narrowband component exists in many channels. As an alternative detection statistic, we considered the semblance, which measures the coherence between the channels. The receiver operating characteristic curves show that the average performs better than the semblance if more than half of the channels contain only wideband noise when the emitter is absent, while the semblance performs better if more than half of the channels contain narrowband components when the emitter is absent. Therefore, we could improve the detection performance of both the average and the semblance.

We developed an improved detection statistic by combining the average and the semblance, determining a combining function and satisfying a set of constraints which ensure that the average and the semblance contribute equally to the detection statistic. Before being combined, the average is transformed to make its probability density function match the probability density function of the semblance. The receiver operating characteristic curves show that the combined statistic performs better than other statistics including the average and the semblance.

We have applied this new detection statistic to the gravitational wave signal detection problem and developed a new algorithm which computes the Fourier transform magnitudes at the exact frequencies using the chirp z-transform. Examples of the gravitational wave signal detection demonstrate that the new algorithm performs better than the previously developed algorithm. This study was completed in December 1989.

## 1.9 The Application of Complex Approximation Algorithms to the Design of Robust Range-Dependent Beamformers

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### Project Staff

Professor Alan V. Oppenheim, James C. Preisig

Beamformers often operate in environments having characteristics that could not be specified at the time of design. These unknown environmental characteristics can relate to signals received, interfering signals, the environment in which the signal propagates, or the beamformer itself. Given a criterion that is a measure of beamformer performance under a given fixed set of conditions, the smaller the value of the criterion, the better the beamformer performs. Therefore, by minimizing the maximum value of the criterion evaluated over the range of possible environmental characteristics, we can design beamformers that cope with these uncertainties.

Mathematically, this type of design problem can be posed as a complex approximation problem. For some applications, such as far-field beamforming in a known homogeneous propagation environment, efficient algorithms already exist that can solve this design problem. The goal of our research is to develop efficient algorithms for range-dependent beamformers operating in non-homogeneous propagation environments.

We are performing this research under the auspices of the MIT-Woods Hole Oceanographic Institution Joint Program.

## 1.10 Analysis and Applications of an Adaptively Trained Recurrent Neural Network

### Sponsors

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### Project Staff

Professor Alan V. Oppenheim, Michael D. Richard

Neural networks have proven useful in many signal processing applications, particularly in those involving pattern classification. In fact, nonparametric neural-network classifiers perform better than traditional ones, such as parametric Bayesian classifiers, on many problems. However, neural-network research has focused on the properties and uses of feedforward networks. Because these networks are only capable of performing memoryless input-output mappings, they are ineffective for problems requiring exploitation of temporal properties of the input data. One of these problems is the analysis of sequential speech patterns for performing speech recognition.

Several neural networks, collectively referred to as recurrent neural networks, have been proposed to deal with "temporal" problems. Most of these consist of traditional feedforward networks, which have feedback connections to transform the networks into nonlinear dynamical systems. Because they have feedback connections, recurrent neural networks are difficult to "train"; and because they are nonlinear dynamical systems, they are difficult to analyze. Therefore, until recently, recurrent neural networks have been primarily curiosity factors rather than practical, useful problem-solving tools.

This research investigates a novel recurrent neural network and its utility for speech recognition. The research is addressing unique methods for adaptively and continuously training this network as input data is received. In addition, we will explore various methods for analyzing recurrent networks to improve the current, superficial understanding of the properties of these networks.

## 1.11 A Code-Division, Multiple Beam Sonar Imaging System

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### Project Staff

Dr. Jules Jaffe, Professor Alan V. Oppenheim, Dr. Timothy Stanton, John M. Richardson

In this research project, we are exploring the development of a new active sonar imaging concept using the principle of code-division and the simultaneous transmission of multiple-coded signals. The signals are sixteen symbol, four-bit, non-linear, block Frequency-Shift Keyed (FSK) codes, each of which is projected into a different direction. Upon reception of the reflected waveform, each signal is separately detected and the results are inverted to yield an estimation of the spatial location of an object in three dimensions. The code-division sonar is particularly effective, operating in situations where the phase of the transmitted signal is perturbed by the propagation media and target.

Most imaging techniques presently used rely on preserving the phase of the received signal over the dimension of the receiving array. In code-division sonar, we use the combined effects of code-to-code rejection and the a priori knowledge of the direction from which each code was transmitted to obtain spatial resolution. We show that the coded signals are highly tolerable of phase distortion over the duration of the transmission. The result is a high-resolution, three-dimensional image that we can obtain in a highly perturbative environment. Additionally, the code-division sonar is capable of a high frame rate due to the simplicity of the processing required.

We have presented two algorithms to estimate the spatial coordinates of an object in the ensonified aperture of the system, comparing their performance for different signal to noise levels. Finally, employing the concept of code-division imaging in a series of experiments, we used code-division sonar image objects under a variety of conditions. We presented the results of the experiments, showing the resolution capabilities of the system.

This research was conducted under the auspices of the MIT-Woods Hole Oceanographic Institution Joint Program and completed in August 1989.

## 1.12 Back-projection with Fourier Series Expansion and FFT

### Project Staff

Makoto Tabei

In this research, we are investigating a computationally efficient procedure for CT reconstruction. The reconstruction procedure that is most commonly used on CT scanners is based on the convolution back-projection (CBP) algorithm. The CBP algorithm consists of two steps: first, projection data is linearly convolved with filter function (convolution); then, the convolved projection is back-projected onto the image frame (back-projection). Although this algorithm is accurate and can be easily implemented, the back-projection is a very time-consuming process, especially when higher-order interpolation is employed for best accuracy. This computational complexity is due to the fact that the contribution of the convolved projection at each projection angle must be evaluated and summed at all pixels in the image frame.

We are proposing an alternative approach to interpolating and back-projecting the convolved projection onto the image frame. The technique is based on the synthesis of arbitrary frequency 2-D sinusoidal functions using the Gaussian function and a 2-D inverse FFT. The procedure is as follows: We transform the convolved projection using a 1-D FFT and replicate it at higher frequencies to perform upsampling (the interpolation generally requires upsampling before filtering out higher-frequency components). Then we multiply the projection with the frequency response of the interpolation function. The resultant sequence provides the Fourier series coefficients of the filtered projections to be back-projected. We use the Gaussian function to project each of the coefficients given along the radial line in polar coordinates and onto rectangular grids. When the filtered projections for all projection angles are projected in the frequency domain, a 2-D IFFT is evoked, correcting its result by division with the reciprocal Gaussian function to produce the reconstructed image.

The computation required in this procedure is one-sixth of conventional back-projection with linear interpolation for the image of 512 x 512. The use of the modified cubic spline or any other higher-order interpolation functions does not affect the computational complexity, requiring only replacement of the tables representing the frequency response of the interpolation function. The proposed algorithm also produces images as accurate as those of any known algorithm.

The results of this study were presented at the IEEE International Conference on Acoustics, Speech, and Signal Processing, Albuquerque, New Mexico, April 3-6, 1990.

## 1.13 Iterative Algorithms for Parameter Estimation from Incomplete Data and Their Applications to Signal Processing

**Sponsors**

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**Project Staff**

Dr. Ehud Weinstein

The Estimate-Maximize (EM) algorithm is an iterative method for finding the Maximum Likelihood (ML) or the Maximum-A-Posteriori (MAP) parameter estimates with incomplete data. The method has been found useful in a variety of signal processing problems including medical imaging, parameter estimation of superimposed signals, speech enhancement in multiple microphone environments, and signal reconstruction from spatial information.

We have developed an extension of the EM algorithm, termed the Cascade EM (CEM) algorithm, that is useful in accelerating the rate of convergence of the algorithm and in simplifying the computations involved. This research was developed with Mordechai Segal of Tel-Aviv University.

Using the EM and the CEM algorithms, we have considered the problem of simultaneous state estimation and parameter identification in linear dynamical continuous/discrete systems. The main result is that we can use Kalman smoothing equations for ML identification of the system parameters. We also developed a new method for calculating the log-likelihood gradient (score), the Hessian, and the Fisher's Information Matrix (FIM), that we used for efficient implementation of gradient-search algorithms and for assessing the mean square accuracy of the ML parameter estimates. This research was developed with Mordechai Segal.

We have also developed a computationally efficient algorithm for estimating the spatial and spectral parameters of multiple source signals using radar/sonar arrays. The most attractive feature of the proposed algorithm is that it decouples the spatial parameter optimization from the spectral optimization, leading to a significant reduction in the computations involved.

This research was developed with Mordechai Segal of Tel-Aviv University and Professor Bruce R. Musicus.

## 1.14 Equalization (Identification) of Non-Minimum Phase Systems

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**Project Staff**

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The problem considered in this study is the following: We observe the output of a discrete time-invariant linear possibly nonminimum phase system H with input being a realization (sample function) from a stochastic process. We want to recover the input signal or equivalently to identify the magnitude and phase of the inverse of H using a tap-delay line (Equalizer). This problem, referred to as self-recovering or blind equalization, is of prime interest in data communications and acoustical and geophysical signal processing.

We have developed necessary and sufficient conditions for equalization. Based on these conditions, we proposed several equalization criteria and proved that their solution corresponds to the desired response. These criteria are universal in the sense that they do not impose any restrictions on the probability law of the input (unobserved) process. The resulting algorithms only involve the computation of a few moments of the system output, implying a simple tap update procedure.

This research was developed with Mr. Ofir Shalvi of Tel-Aviv University.

## 1.15 Signal Processing with 1/f Processes using Wavelets

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Recently, we have developed a wavelet-based Karhunen-Loeve expansion for a family of nearly-1/f processes. Our subsequent work focused on exploiting this expansion in the solution of some rather general signal processing problems involving these processes. Since 1/f models are applicable to a wide range of natural and man-made phenomena, there are many applications of this work.

In the area of 1/f signal modeling, we are investigating two problems. The first involves determining the wavelet which leads to a Karhunen-Loeve expansion for exactly-1/f processes. Secondly, we have defined a discrete-time counterpart to the continuous-time 1/f process. At present, we are developing useful characterizations of this process and studying its properties.

Some identification problems involving 1/f processes are also of interest. The first involves hypothesis testing, i.e, determining whether a process under investigation is a sample function of a 1/f process. In solving the second problem, we estimated the parameters of a 1/f process from possibly noisy observations. We have obtained a number of results for this problem using Maximum Likelihood theory.

In terms of detection problems, we are working on the development of likelihood-ratio tests for the detection of signals in additive 1/f-noise environments. Also of interest is the design of signals for low probability of detection in 1/f-noise backgrounds.

Finally, we are considering a variety of estimation problems for 1/f processes. In general, we are studying the problems of smoothing, filtering, and prediction of 1/f processes based on noisy observations.

In all of the above cases, we are finding not only computationally efficient solutions that exploit the self-similar structure of these processes, but intuitively satisfying interpretations as well.