20. Digital Signal Processing

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20.1 Introduction

The Digital Signal Processing Group is carrying out research in the general area of digital signal processing with applications to speech, biomedical, image and geophysical data processing. In addition to specific projects being carried out on campus, there is close interaction with Lincoln Laboratory, with the Woods Hole Oceanographic Institution, with the Schlumberger-Doll Research Center, and with the Non-Invasive Diagnostic Laboratory at Massachusetts General Hospital.

In the area of speech processing, the Digital Signal Processing Group has been working on the development of systems for bandwidth compression of speech, parametric modeling of speech using pole-zero models, time-scale modification of speech, enhancement of degraded speech, and speech reconstruction from spectrograms. Our current work in the speech area involves the problem of enhancing degraded speech and a related problem, that of the development of algorithms for robust speech compression in the presence of additive noise. In addition, we are investigating a variety of conditions under which a speech signal can be reconstructed from spectrograms.

In a related area, the methods of speech compression using linear predictive encoding are being applied to the compression of data recorded in ocean bottom seismometers. These methods have been tested with data provided by Woods Hole Oceanographic Institution and indicate that data recording requirements can be halved with little loss of fidelity.

For the problem of non-invasive measurement of bloodflow characteristics two projects are being carried out. The first involves spectrum analysis of sounds generated by the heart (e.g., by artificial heart valves). The applicability of MEM spectrum analysis to these (rather short-duration) signals is being studied. It is hoped that with precise spectral measurements, it will be possible to detect and quantify degradations in artificial heart valve performance. The second project area involves the use of an active ultrasonic measurement system. Measurement of blood velocity can be accomplished via the doppler principle, while imaging is done in a pulse-echo mode. The objective in this project is to measure regions of disturbed flow (i.e., turbulence) which are associated with arterial disease. Spectrum analysis of the doppler signal is a key part of such measurement, but more sophisticated processing to combine the doppler information with a B-mode image are being considered.

The areas of image and geophysical data processing in general both involve the processing of multidimensional signals. The theoretical projects in 2-D signal processing include filter design (e.g. 2-D allpass design to match phase response), the synthesis of good 2-D filter implementations, 2-D spectrum analysis, and 2-D deconvolution. Specifically related to geophysical data processing we have been pursuing a number of projects. One project, which is being carried out in collaboration with the Woods Hole Oceanographic Institution, is the development of an algorithm for data processing to measure the acoustic reflection coefficient from the ocean bottom. Out of this work has come a new method for generating synthetic data. Currently a data extension technique is being developed. Another problem area is the determination of formation parameters from acoustic well logs. The estimation procedure consists of first removing the modal behavior and other unwanted components from the measured signals.

From March to May, 1980, we led a large acoustics and geophysics experiment, FRAM II, in the Eastern Arctic. In another application of velocity analysis, we have applied adaptive array processing to measure the reverberation of acoustic signals in the Arctic Ocean as well as the phase velocity of the seismic paths within the seabed for refraction studies. Some of the acoustic experiments involve sonar signal processing for operating over a spread channel. Acoustic imaging from a submersible often generates an image dominated by strong highlights because of the specular reflections introduced by the relatively long wavelengths. We are working on an adaptive array processing method to suppress the deleterious effects of these highlights in the image.

There are also a number of projects related to image processing that we are currently pursuing. One project is restoration of images degraded by additive noise, multiplicative noise and convolutional noise. Out of this project, we have developed a new image restoration system which is applicable to restoring images degraded by various different types of degradation. Our current work in this project involves evaluation of the new restoration system and development of other image restoration systems. Another project is reconstruction of images from its Fourier transform phase, and its application to blind deconvolution and image coding. In this project, we have developed a new set of conditions under which an image can be restored from its Fourier transform phase alone. We are currrently working on the application of these theoretical results to practical problems such as blind deconvolution. A third project is development of new image coding techniques by reducing quantization noise in PCM image coding. Our approach to first decorrelate the quantization noise and then reduce the quantization noise by a noise reduction system led to a noticeable improvement in the performance of a simple PCM image coding system. We are currently working on the extension of these results to a more complex PCM image coding system. Our work on image processing is being carried out in collaboration with Lincoln Laboratory.

In both the context of image processing and array processing we have obtained some significant results in the multi-dimensional high resolution spectral estimation problem. Specifically, we have developed new algorithms for maximum entropy power spectrum estimation which are computationally simple and applicable to both equally spaced and non-equally spaced data for both one-dimensional and two-dimensional signals. This algorithm is currently being applied to investigate the characteristics of multi-dimensional maximum entropy power spectrum estimation solutions.

One new area of interest is the combination of signal processing and artificial intelligence techniques. There are a variety of problems in signal analysis that can be approached either from the analytical viewpoint characteristic of signal processing, or the symbolic viewpoint characteristic of knowledge-based systems and artificial intelligence. There is also the potential for combining these two viewpoints into what perhaps could be referred to as knowledge-based signal processing. We are currently exploring this new area both generally and in the context of some specific applications.

20.2 The Parabolic Wave Equation

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Arthur B. Baggeroer, Thomas E. Bordley

In underwater acoustics, the parabolic (Schroedinger) wave equation is often used as an approximation to the hyperbolic wave equation when solutions are sought in regions with slowly varying inhomogeneities. Such inhomogeneities prevent separation-of-variables or other simple solution techniques from being employed. The rationale behind the parabolic approximation is that if the inhomogeneities are weak, then the component of the field which is due to reflections ought to be small. Thus, it should be legitimate to split the wave equation into a pair of coupled equations in the "transmitted" and "reflected" fields and then solve for the transmitted field with the reflected field set to zero.

This approximation has two significant effects. First, since the equation is reduced to first order in the direction of propagation or range, the original problem is reduced to a comparatively simple initial value problem for which the necessary initial conditions can be readily found. Second, the split fields are decoupled approximants to the true transmitted and reflected fields, possessing a parabolic rather than a circular dispersion relation. Therefore, both the shape and scaling of waves emanating from the source at large angles with respect to the nominal direction of propagation are wrong.

Such errors are of negligible importance in many marine problems. Inhomogeneities in the water column normally are significant only when large distances are involved. Moreover, since the attenuation in the bottom is very heavy, only those waves which possess a small grazing angle with respect to the bottom, or equivalently, which emanate at an angle close to horizontal with respect to the source, contribute substantially to the solution over large ranges. Ordinarily then, a good estimate of the field can be obtained even for a significantly inhomogeneous media, because those waves which contribute the most to the final solution are represented well while only those which contribute little are represented poorly.

When the signals propagation through the bottom are not small or are of interest in themselves, this approach is less satisfactory. Local variations in the acoustic field with range can be rapid due to the presence of distinct material layers in the earth. Thus, the approximation can be very poor. In this research, we are concerned with extending the parabolic approximation so that the fields in both the earth and the sea are modelled well.

20.3 Knowledge-Based Signal Processing

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National Science Foundation (Grant ECS80-07102)
Alan V. Oppenheim, Randall Davis, Webster P. Dove

We are working on problems in the area of knowledge-based signal processing. In particular, we are presently implementing a version of the Gold-Rabiner pitch detector on a computer designed at the MIT Artificial Intelligence Lab known as the LISP machine. Ultimately, we hope to integrate into our implementation more knowledge about the pitch detection problem in order to achieve better performance of the pitch detector.

20.4 MEM Spectral Estimation for Non-uniformly Sampled Signals

U.S. Army Research Office (Contract DAAG29-81-K-0073)
James H. McClellan, Farid U. Dowla

The standard maximum entropy algorithms require that the signal or its autocorrelation function be known on a uniform grid. However, there are many situations where the signal might be sampled non-uniformly. Spatial sampling for wave-number (bearing) analysis of geophysical, radar, or oceanographic data is sometimes non-uniform. As a solution to the above problem, McDonough[1] proposed interpolating the non-uniform signal samples onto a uniform grid and then employing the standard MEM. He further suggested that a bandlimited minimum-energy interpolation be performed on the known signal samples to obtain the uniform samples.

The second solution, implied in Burg's thesis,[2] would be to solve directly for the maximum entropy spectrum of the non-uniformly sampled signal. The theory and implementation for this approach has recently received closer attention by several authors.[3,4,5,6]

Algorithms of Lang and McClellan[7] and of Lim and Malik[4] which solve for the ME spectrum for arbitrarily spaced correlation samples are well suited to this problem.

An empirical investigation on the performance of these methods on signals which were non-uniformly sampled is being conducted. Within the context of specific estimation problems, i.e., determining peak frequencies and resolving closely spaced peaks for sinusoidal and autoregressive processes in white Gaussian noise background, the performance of a true maximum entropy solution seems superior. A paper summarizing this is being written.

The comparison between the above algorithms is also being performed.

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20.5 Geophysical Inversion by Velocity Spectra Migration and Joint Travel-time and Amplitude Techniques

U.S. Navy - Office of Naval Research (Contract N00014-77-C-0266)

Arthur B. Baggeroer, Gregory L. Duckworth

This work examines the resolution and uniqueness properties of several velocity function inversion techniques for multichannel seismic refraction/wide angle reflection and ocean bottom interaction data. The performance of an array processing technique for velocity spectra, or frequency–

wavenumber. estimation is quantified as a preprocessor for an iterative wave equation migration algorithm, a recursive travel time technique, and a joint travel-time and amplitude inversion procedure.

The migration maps the arrival time, phase velocity, and frequency decomposition of the data from the velocity spectral estimate directly into a velocity-depth section. In contrast to commonly used techniques, the method allows data collected on a small array at a limited number of offsets to be inverted in a manner that preserves both amplitudes and the resolution inherent in the entire processing chain. All multiple arrivals are retained and used simultaneously in the inversion.

The relatively simple travel time inversion is quantified in terms of its resolution and stability for data sets in which higher order multiples as well as first arrivals are used to generate the tau-p data on which the inversion is carried out. This technique is of interest because of its light computational load and its direct exposition of data sets which are physically infeasible. This feedback can then be used to edit the data set to ensure feasibility by eliminating erroneous picks or processing biases. (Near feasibility in the event of random errors due to noise.)

The joint travel-time and amplitude inversion is an iterative procedure which uses the WKBJ approximation to the wave amplitude function along with the travel time functions to enhance the uniqueness properties of the result. This technique is useful when experimental limitations disallow dense spatial coverage in the data acquisition.

It is the ability of the adaptive velocity spectral estimator to resolve late breaking arrivals which allows the multiples to be discerned and included in all the inversion techniques mentioned. The simultaneous inversion of the multiples allows a simplified experimental procedure to be more fully exploited. The frequency resolution of the technique allows frequency dependent propagation effects to be observed.

The procedure is demonstrated on a refraction survey shot during the Fram II experiment in the eastern Arctic Ocean. Many propagation phenomena are observed, including modal propagation within the ocean bottom and crust and compressional to shear conversions at impedance discontinuities.

20.6 A Comparison of Power Spectrum Estimation Techniques for Array Data

Hughes Aircraft Co. Fellowship

National Science Foundation (Grant ECS80-07102)

U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)

Jae S. Lim, Michelle J. Glaser

The data gathered from a planar array of microphones may be analyzed via power spectrum estimation (PSE) techniques to estimate the azimuth and elevation of a sound source. This type of problem arises in applications such as the detection and location of a low-flying aircraft. As this problem is inherently three dimensional and it is computationally impractical to do PSE in more than two dimensions, the estimation has been done in two steps. First, the spatial covariance matrix is estimated at a specific temporal frequency using block averaged periodograms. Then, this covariance is used to do the two dimensional spatial spectrum estimation.

The first goal of this project is to compare the performance of three different PSE techniques in this setting. The three techniques are: the Maximum Entropy Method (MEM), the Maximum Likelihood Method (MLM), and the Bartlett Method. This assumption will be useful in determining the extent to which algorithms recently developed to solve the two-dimensional MEM problem will have an impact in real world problems. Performance will be judged on the basis of such factors as peak location and minimum peak separation for resolution. In addition, the effect of varying the number of blocks averaged in estimating the covariance matrix will be examined in conjunction with each of the three algorithms above. The second goal is to determine whether the MEM estimate is a power spectrum or a power spectral density estimate.

20.7 Characterization of a Atherosclerosis Using Pulse Doppler Ultrasound

American Edwards Labs. Grant
Whitaker Health Sciences Fund
James H. McClellan, William A. Harrison

Detection of mild stenosis in the carotid arteries is important in reducing the incidence of stroke. Pulse doppler ultrasound offers a non-invasive method of detecting stenosis in superficial arteries. But, current doppler systems are not always able to detect mild stenosis. The approach of this research is to quantify the factors that affect the frequency spectrum of the received backscattered ultrasound. It is known that the spatially varying velocity of the blood and the finite dimension of the ultrasound beam cause a broadening of the frequency spectrum. The effect of stenosis is to alter the velocity profile of the blood. Currently, an attempt is being made to derive the frequency spectrum for arbitrary blood flow profile. The goal is then to generate the frequency spectrum of the return signal and correlate it with the degree of stenosis. This work is being done in collaboration with Dr. Robert S. Lees of the Arteriosclerosis Research Center at M.I.T.

20.8 Pole-Zero Modelling and Classification of Phonocardiograms

Pfeiffer Foundation Grant

Whitaker Health Sciences Fund

James H. McClellan, Tae H. Joo

Detection of failure of heart valves is an important medical problem. One non-invasive diagnosis method is analysis of phonocardiograms. In this work, phonocardiograms of porcine bio-prosthetic valves are analyzed to classify them into normal or abnormal. The classifier is designed using the features selected from the power spectrum which is derived by autoregressive moving-average signal modelling methods. This work is done in collaboration with Dr. Robert S. Lees of Massachusetts General Hospital Non-Invasive Diagnosis Laboratory.

20.9 Reflection Coefficient Estimation in the Borehole Environment

Schlumberger-Doll Research Center Fellowship

U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)

National Science Foundation (Grant ECS80-07102)

Alan V. Oppenheim, Andrew L. Kurkjian

Given an acoustic source and an array of receivers within the borehole, this research explores the estimation of the cylindrical wave reflection coefficient as a function of frequency and the axial component of wavenumber (i.e. angle). The estimation of the reflection coefficient is analogous to performing reflection seismology within a borehole. The reflection coefficient is useful for studying the radial variation of a rock formation due to formation fluids and gasses displaced by the drilling muds. The processing problem is one dereverberating the received array of data and an iterative algorithm is explored which converges to the maximum likelihood estimate of the reflection coefficient. This iterative solution to the nonlinear ML problem is considered a major contribution of this research. A second contribution of this work is in the rapid evaluation of arrays of synthetic acoustic well logging waveforms.

20.10 One and Two Dimensional Maximum Entropy Spectral Estimation

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U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)
Jae S. Lim, Naveed A. Malik

This research has investigated the Maximum Entropy (ME) power spectrum estimation problem for two-dimensional signals. This problem is important since it is known that the ME spectral estimates have high resolution characteristics for one-dimensional signals and it is expected that this property will extend to two-dimensional signals. The problem involves obtaining a correlation matching estimate for the power spectrum that can be represented as an autoregressive model with the same order as that of the known correlations. A new algorithm for solving the highly non-linear problem has been developed. The algorithm uses Fourier transform operations only and can be implemented on a main-frame computer with large on-line memory, as well as on a mini-computer with limited storage. The properties of the 2-D ME spectra have been characterized and the results have been compared to the Maximum Likelihood and Bartlett estimates. The algorithm has also been shown to be applicable to one-dimensional signals with missing correlation values.

20.11 Investigation of the Importance of Phase of the Residual Error Signal in the Linear Predictive Model of Speech

National Science Foundation (Grant ECS80-07102)
U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)
Alan V. Oppenheim, Evangelos E. Milios

Linear predictive modelling of speech signals has been applied successfully so far for speech parameter estimation and representation of speech for low bit rate transmission. Our aim is to check experimentally if the phase-only version of the long-time residual sequence contains sufficient information to produce high-quality speech in the output. The project also involves the implementation of a modified version of Singleton's algorithm for the computation of long FFT's using disc memory. Efficient coding of the phase-only version of the residual sequence will finally be examined.

20.12 The Determination of the Acoustic Reflection Coefficient of the Ocean Bottom from its Response to a Point Source

U.S. Navy - Office of Naval Research (Contracts N00014-77-C-0196 and N00014-81-K-0742)
National Science Foundation (Grant ECS80-07102)
Alan V. Oppenheim, George V. Frisk[9], Douglas R. Mook

The determination of the plane wave reflection coefficient for real and complex angles from acoustic measurements is continuing. Analysis involves computation of the Hankel transform. Associated issues related to projection, interpolation and windowing are also being studied. The feasibility of the method for determining the reflection coefficient is being examined by studying its performance in an environment subject to real world sources of degradation. These studies are being done as a combination of theoretical work and computer experiments on synthetically generated data. A theoretically exact method for generating the necessary synthetic data for all ranges has been developed that employs both a numeric and an analytic component.

20.13 Optimal Signal Reconstruction and ARMA Model Identification Given Noisy and Incomplete Observation Data

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U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)
Jae S. Lim, Bruce R. Musicus

There are a large number of signal processing applications in which signals and/or model parameters must be estimated from noisy or incomplete observation data. For example, we may need to reconstruct a finite length signal from noisy measurements of its phase or magnitude. We may want to filter and extrapolate a finite interval of stationary data given an ARMA or a bandlimited model of the signal process. We may wish to optimally fit an ARMA model to a signal given a finite interval of noisy measurements, possibly with missing data points. We may wish to design a multi-dimensional FIR filter to meet certain complicated time and frequency domain constraints. Surprisingly, all these problems have a similar structure, and can be solved in a similar manner. We start with a probabilistic model describing the relationship between the signal, the noisy system output, and any unknown system parameters. The given observation information defines constraints on the range of possible signal, output, and parameter values. A variety of Maximum Likelihood criteria can then be applied to estimate the unknowns. Each choice of an optimality criterion leads to a different set of estimates. Despite this, they can all be solved by similar iterative algorithms. Starting from initial estimates of the

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[9]Woods Hole Oceanographic Institution

unknowns, we optimally filter the noisy output and project the result onto the signal constraint space in order to estimate the signal. Next we estimate the output by projecting this signal estimate onto the output constraint space. Given the new signal and output estimates, the parameters are then reestimated. Appropriate compensation is included at each step for the uncertainty of our estimates. The next iteration uses the improved outpt and parameter estimates to improve the next filtered signal estimate, and the cycle continues. Each iteration increases the likelihood function, and convergence can be guaranteed under mild assumptions. Closed form solutions can also be derived in certain cases.

Another major focus of our research is the development of new algorithms for solving almost-Toeplitz sets of linear equations. Equations of this type arise quite commonly in the filtering and signal reconstruction problems we have been investigating, and efficient solution methods are especially critical because the problems must be solved repetitively as the algorithms iterate. Levinson-style recursions have been most commonly discussed in the literature. Fast Choleski methods for solving the same problems can be derived as "mirror images" of the Levinson-style recursion. Using both forward and backward recursions to minimize storage, these fast Choleski methods are particularly valuable for solving the band diagonal, almost-Toeplitz problems that arise when filtering noisy ARMA signal data. These algorithms also strongly resemble "inside out" Euclidian polynomial algorithms, an observation which directly leads to new "doubling algorithms" for solving almost-Toeplitz problems in only $0(N\log^2 N)$ operations. Connections between Euclidian polynomial algorithms and Levinson recursion algorithms are also being explored in other contexts as well; in particular, we have developed a new Euclidian algorithm with "partial pivoting" for evaluating inverse Z-transforms and inverse Laplace transforms of rational polynomials.

20.14 Artificial Intelligence and Signal Processing in Scene Analysis

Schlumberger-Doll Research Center Fellowship
U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)
National Science Foundation (Grant ECS80-07102)
Alan V. Oppenheim, Cory Myers

The problem of computerized scene analysis has been approached by both the signal processing and the artificial intelligence communities. The purpose of this research is to investigate those systems which have been used in the domain of scene analysis and to determine how signal processing and artificial intelligence techniques have been combined in these systems. The ultimate goal of this research is to determine if there are good methods for combining artificial intelligence and signal processing techniques which may be extended to various problem domains.

20.15 Signal Reconstruction from the Short-Time Fourier Transform Magnitude

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National Science Foundation (Grant ECS80-07102)
Alan V. Oppenheim, Hamid Nawab

In the context of short-time signal restoration, it has been shown that under very mild restrictions a discrete-time signal can be reconstructed from its short-time Fourier transform magnitude. Furthermore. reconstruction can also be carried out when the STFT magnitude has been adequately sampled in time or frequency or both. Based on these results. restoration algorithms can be developed for one-dimensional as well as multi-dimensional signals degraded by additive noise. The performance of such algorithms is currently under investigation. Another important application under consideration is signal coding with the sampled STFT magnitude.

20.16 MEM Power Spectral Estimation with Uncertainty in Correlation Measurements

National Science Foundation (Grant ECS79-15226)
James H. McClellan, Jean-Pierre Schott

Traditionally in power spectral estimation, one uses MEM to apply the matching constraint property, i.e. the correlations obtained from the computed power spectrum match the given correlations. In many instances, the given correlations are noisy, and consequently the justification of matching the correlation measurements is less obvious. The first goal of this research is to derive a new MEM algorithm where the matching equation is replaced by an inequality constraint in the form of a quadratic involving the given correlations and a weighting matrix W. In the course of this research, it appeared that a good candidate for the inverse of W was the covariance of the correlation estimates of the signal. Currently, the theoretical properties of this matrix W, as well as the performance of the algorithm with synthetic data, are under investigation.

20.17 Unimportance of Phase in Speech

Hughes Aircraft Company Fellowship
National Science Foundation (Grant ECS80-07102)
U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)
Jae S. Lim, David L. Wang

The enhancement of speech degraded by additive noise has recently received considerable attention in the literature. Most of the systems proposed have been based on combining an estimate of the short-time spectral-magnitude of the speech signal with the Fourier transform phase of the degraded speech. Some recent research results, however, suggest that more accurate phase estimation might be useful in improving the performance of these schemes. Our concern is to test this hypothesis, by considering the importance of the Fourier transform phase in speech enhancement. Our results indicate that an effort to estimate the phase more accurately from the noisy speech is unwarranted for speech enhancement schemes based on the phase-only reconstruction techniques or for schemes based on combining the phase estimate with an independently estimated magnitude.

20.18 Comparison of Fast Hankel Transforms

Hertz Foundation Fellowship
National Science Foundation (Grant ECS80-07102)
U.S. Navy - Office of Naval Research (Contract N00014-81-K-0742)
Alan V. Oppenheim, Michael S. Wengrovitz

Numerical computation of Hankel (Fourier-Bessel) transforms is required in a number of applications including optics, electro-magnetics and acoustics. This research project is intended to be a preliminary investigation and comparison of current algorithms which perform the fast Hankel transform. Advantages of several techniques will be investigated and computer simulations will be performed. The effect of interpolation will be investigated and an improved algorithm will be specified and simulated.