XIX. DIGITAL SIGNAL PROCESSING

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The Digital Signal Processing Group is carrying out research in the general area of digital signal processing with applications to speech, image, and geophysical data processing. In addition to specific projects being carried out on campus, there is close interaction both with Lincoln Laboratory and with the Woods Hole Oceanographic Institution.

In the area of speech processing, over the past several years 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, and enhancement of degraded speech. Our work in the speech area is currently heading toward an increasing involvement with 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 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 are being tested with data provided by the Woods Hole Oceanographic Institution.

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 all-pass design to match phase response), the synthesis of good 2-D filter implementations, 2-D spectrum analysis, and 2-D deconvolution. We have been pursuing a number of projects specifically related to geophysical data processing. We are applying some of the filter design results to seismic-wave migration by implementing a program on our MAP processor. Another project, which has been 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 Hankel transform algorithm which has potential applications to a number of other problems. Another problem area is that of velocity analysis on array data. The specific application that we are considering is that of velocity analysis on well logging data. We are also pursuing a number of other problems associated with the analysis of well logging data, including the development of techniques for event detection. 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. 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. The work on image processing is being carried out in collaboration with Lincoln Laboratory, and we are in the process of defining a research program which would involve close collaboration between our group and Lincoln Laboratory. Projects that we are contemplating include enhancement of degraded images, and reconstruction of images from phase-only information. In both the context of image processing and array processing, we are also beginning to explore such topics as high resolution, multidimensional spectral estimation, and two-dimensional short-space signal processing.

1. LINEAR PREDICTIVE ENCODING OF SEISMIC DATA

National Science Foundation Fellowship

Thomas E. Bordley, Arthur B. Baggeroer

If marine seismic traces are stored in their original digital form, large quantities of data storage are necessary because of the broad dynamic range of these signals. Since these signals are not sample waveforms of a white noise process, it is known that the data can be presented more efficiently (e.g., via entropy encoding). Thus, ocean-bottom seismometers with their limited available storage are artificially constrained in the number of signals which they can record, if the incoming data are simply stored without processing. Since the retrieval of these sensors is difficult and expensive, it is of significant interest to determine a processing scheme suitable for use by these minicomputer-controlled seismometers. This research examines the effectiveness of Linear Predictive Encoding (LPE) in reducing the amount of storage required for data gathered in ocean-bottom seismology.

The essence of this technique is to characterize a waveform in terms of the parameters of a stationary rational digital model, i.e., as the output of a reverberative system, and then to store the parameters of this system and a correction signal instead of storing the original signal. The rationale behind this approach is that if a signal is sufficiently predictable in terms of the model, the energy in the error signal will be much less than the energy in the original signal. Thus, the total number of bits required to represent the signal as an error signal and a set of parameters will be much less than needed to represent the waveform directly.

At present, we are engaged in empirically testing this technique on data supplied through Dr. Graham M. Purdy of the Woods Hole Oceanographic Institution.

2. IMPLEMENTATION OF MULTIDIMENSIONAL DISCRETE SYSTEMS FOR SIGNAL PROCESSING

Joint Services Electronics Program (Contract DAAG29-78-C-0020)

David S. K. Chan, James H. McClellan

This research has established a "state-space" representation for studying the implementation of a general class of multidimensional discrete systems. This formulation extends to cases other than the first-quadrant causal filters that are usually studied. Using this framework, the minimization of coefficient sensitivity and round-off noise under structure transformations can be studied. The analog network technique known as continuously equivalent networks has been adapted to the multidimensional realization problem. Work is progressing on the realization algorithm to improve its performance, especially for the two-dimensional case.

References

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2. J. H. McClellan and D. S. K. Chan, "Realization of Two-Dimensional Systems," <u>Proceedings of the 1978 European Conference on Circuit Theory and Design</u>, September 4-8, 1978, Lausanne, Switzerland, pp. 381-388.

3. EVENT DETECTION IN SONIC WELL LOGGING

National Science Foundation (Grant ENG76-24117)

Webster P. Dove, Alan V. Oppenheim

Oil wells are analyzed by acoustically testing at many places along their depth, from which a sound velocity profile can be developed. For each test a pulse of sound is generated at the bottom of a 13 meter long test probe and received at four microphones spaced at one meter intervals at the top of the probe.

The signal received at each microphone is the sum of many overlapping dispersed pulses, each of which has travelled a different path. To find the velocities in the paths of interest the arrival time of each related pulse must be determined accurately. That requires a method of reducing the dispersion of the pulses so they become distinct separate arrivals.

To do this signal processing, we are using Recursive Least Squares prediction (the covariance method) to cancel the filtering effect of the different paths. Then the arrival time of the first and second pulses (which are the ones of interest) should be apparent either in the output of the predictor or the behavior of the predictor coefficients.

4. ADAPTIVE ARRAY PROCESSING FOR HIGH-RESOLUTION ACOUSTIC IMAGING

National Science Foundation Fellowship

Gregory L. Duckworth, Arthur B. Baggeroer

Determination of the internal structure of a medium opaque or ill-suited to electromagnetic radiation is a problem encountered in many different applications. High-resolution visualization of underwater objects through turbid seawater is the problem currently being dealt with; however, other applications include real-time viewing of internal movements of the human body without x-ray's potentially harmful effects, nondestructive testing of metallic and low x-ray contrast objects, and determination of the earth's subsurface structure.

Because of their analogous behavior to electromagnetic radiation with respect to reflection, diffraction, and refraction, but differing attenuation and physiological properties, short wavelength acoustic pressure waves can be used to perform the above tasks, but with a new set of inherent advantages and difficulties. For example, in the context of the undersea environment, acoustic imaging has an advantage over optical imaging in that the attenuation of the acoustic-pressure waves is dependent primarily on the temporal frequency, and relatively independent of the density of suspended solids, whereas light is subject to intense backscattering from cloudy water. A result of this is that the "range-to-reverberation" limit is larger for acoustic imaging, and although absorption at wavelengths adequate for reasonable resolution is high, we can theoretically increase the illuminating power and obtain the desired range capabilities.

The problems with acoustic imaging stem from the need to keep the wavelengths long enough for adequate range, SNR, and power consumption, and short enough for good resolution and small receiver apertures. These considerations ultimately lead to systems with small numerical apertures with the diffraction field undersampled in space and hence, poor resolution and aliasing problems. Resolution seems to be the most problematic issue since the large point-spread function generated by classical (Fresnel transform) processing is subject to tremendous amounts of sidelobe leakage from specular reflections. Typically a great deal of specularity is encountered since the illuminating wavelengths are large compared to the surface roughness of the objects to be imaged.

These problems lead us to the thrust of the current research – application of the "Maximum Likelihood" technique of spectral analysis to adaptive array processing of the diffraction-pattern samples. It has been found that the adaptive point-spread function of a system incorporating this technique yields better resolution for distributed objects as long as care is taken in estimation of the spatial-covariance function. The subtlety involves making the spatial covariance look like it was formed by reflections from statistically independent incremental areas. The work also involves determination of the statistics of the estimators when an inadequate number of data vectors are used to ensure that the spatial-covariance matrix is distributed in a complex Wishart manner. Two-dimensional arrays that are optimized in some sense for good resolution and aliasing reduction with the minimum number of sensors are also examined.

5. DESIGN OF TWO-DIMENSIONAL ALL-PASS FILTERS

U. S. Navy – Office of Naval Research (Contract N00014-75-C-0951)
National Science Foundation (Grant ENG76-24117)

David B. Harris, James H. McClellan

The objective of this research project is to develop methods for designing all-pass filters with specified phase characteristics. Little previous work has been done on the design of phase functions, particularly in two dimensions, since the problem is highly nonlinear.

Several applications for two-dimensional phase-only filters await the development of satisfactory design techniques. The most important is in simulation of acoustic-wave propagation. In a two-dimensional spatial geometry involving depth and lateral offset, the wave field recorded (for all time and offset) at a particular depth is related to the wave field at another depth by a two-dimensional filtering operation. The implied filter is specified by one of two solutions to the two-dimensional acoustic-wave equation. The filter characteristic is all-pass with the phase given by the dispersion relation of the wave equation. A broadband digital filter approximation to this ideal response is being sought to enable numerical extrapolation of measured wave fields from one depth to another. Wave-field extrapolation of this sort is an integral part of the wave-equation method of seismic time-section migration.

Another possible application of all-pass filter design is phase compensation of two-dimensional recursive filters designed for magnitude only.

Recently, progress in design of all-pass has been obtained. A method whereby the phase-design problem is transformed to a more tractable 2-D spectral-factorization problem has been used effectively. And a new form of 2-D linear prediction has been developed to perform the spectral factorization.

6. TIME-SCALE MODIFICATION OF SPEECH

U. S. Navy – Office of Naval Research (Contract N00014-75-C-0951)

Samuel Holtzman, Michael R. Portnoff

We have implemented an analysis-synthesis system on our PDP-11 computer that performs uniform-rate speed transformations on speech signals. The problem of spectral degradation and introduction of noise, which usually occur in similar systems, are not found in this one. This is because the system performs the transformations in the frequency domain by means of the discrete short-time Fourier transform rather than in the time domain.

In order to achieve an even more natural-sounding result, we are at present introducing nonuniformities into the speed transformations to incorporate a dependency of the system on local features of the speech signal being transformed.

Our work has been directed toward the development of an algorithm to automatically segment the speech signal into a sequence of passages for which an expected level of degradation, caused by uniform time-scaling, can be determined. The purpose of segmenting the signal in this manner is to allow the degree of local time-scale modification to be decreased whenever the expected level of degradation is high.

The algorithm uses a statistical analysis of the speech signal to determine a level of local quasi-stationarity which, based on our model of speech production, is highly correlated with the expected level of local degradation.

7. PARAMETER ESTIMATION FROM SEISMIC DATA

Schlumberger-Doll Research Center Fellowship

Andrew Kurkjian, Alan V. Oppenheim

The purpose of this research is to develop signal processing for a borehole (oil well) sonic tool which takes into account more physics than is found in a simple nondispersive model. The problem is to estimate certain parameters of the rock surrounding the borehole from signals received at an array of sensors in the borehole. The actual physics of the situation is very complicated due to the variety of seismic-wave phenomena which are present. Aside from compressional and shear wavelets, the pressure field also contains a water (tube, mud) wave, a head (lateral, conical) wave, a pseudo-Rayleigh wave, and a Stoneley wave. This work will examine the physical nature of these seismic waves and then develop processing to estimate the parameters of interest from the received signals based on this physical nature.

8. PERFORMANCE OF MAXIMUM ENTROPY SPECTRAL ESTIMATORS

Hertz Foundation Fellowship

Steven W. Lang, James H. McClellan

Many problems in estimation can be considered as the estimation of some feature of a power spectrum. The development of new methods of spectral estimation, such as various "maximum entropy" techniques, gives us new tools with which to attack such problems.

This research is concerned with an analysis of the performance of various maximum entropy spectral estimators when the random process being observed is composed of sinusoids in additive noise. In particular, the problem of estimating the sinusoid frequencies is considered. This problem is interesting and important in its own right; the measurement of Doppler shifts in radar or the search for periodicities in geophysical data might be so modeled. It also points up some differences between various "maximum entropy" spectral estimators. Thus, the results obtained should provide some insight into the performance of these estimators in other situations.

9. PARAMETRIC MODELLING OF THE LUNGS FROM ACOUSTIC SIGNALS

National Science Foundation Fellowship

David C. LeDoux, James H. McClellan

This is part of a project attempting to develop new diagnostic techniques for detecting pulmonary disease in infants and children. In these techniques, a sound wave is injected into the patient's respiratory system through the mouth. The sound is reflected from various points within the lungs, and the reflected signal is recorded as it emerges from the mouth. If we model the lung as a linear time-invariant system, we can determine its transfer function (frequency response or impulse response) from the input and output signals. This transfer function contains information about the air passages within the lungs and can hopefully be used to detect such problems as blocked or constricted airways.

Since the air passages in the lungs form a branching network of tubes, we would expect the frequency response to exhibit dips due to resonances of single tubes and combinations of tubes. This is, in fact, what is observed from actual data. However, the human lung contains several million tubes and the frequency response is correspondingly complicated. The object of this research is to apply the techniques of pole-zero modeling to represent the dominant features of the frequency response by a small number of parameters (poles and zeros). It is hoped that some future workers might be able to use the parameters as a basis for differentiating between healthy and diseased lungs.

10. SPEECH ENHANCEMENT

U. S. Navy – Office of Naval Research (Contract N00014-75-C-0951)

Jae S. Lim, Alan V. Oppenheim

Degraded speech occurs in a variety of contexts, and its enhancement is desirable for many practical applications. In our past research on this problem, we have developed several systems for enhancement and bandwidth compression of noisy speech by attempting to estimate the parameters of a specific underlying speech model based on the Maximum A Posteriori (MAP) estimation procedure. When the systems were implemented and applied to real speech data, they performed well as enhancement and potential bandwidth compression systems of noisy speech at various S/N ratios. Our future research in this area will include investigation of methods to improve our current speech-enhancement systems and development of new systems.

11. MAXIMUM LIKELIHOOD ESTIMATION WITH NOISY DATA

U. S. Navy – Office of Naval Research (Contract N00014-75-C-0951)

Bruce Musicus, Jae S. Lim

Maximum Likelihood (ML) Estimation is a powerful tool for estimating model parameters or signals from observed system output. Not only does it yield estimates with nice theoretical properties, but the estimates are also easily calculated for many useful signal models. Unfortunately, when both the parameters of the system as well as the system output must be estimated from observations corrupted by noise, Maximum Likelihood Estimation usually requires a difficult nonlinear optimization. Three different ML approaches have been proposed for estimating the signal and parameters of a system from noisy observations. We have found iterative algorithms for solving each of the three problems, which effectively decouple the uncertainty in the parameter and signal values, thus simplifying the calculation required. When applied to a particular pole-zero model, all three algorithms iterate back and forth between linearly filtering the observations to estimate the signal and fitting parameters to the signal estimate by solving linear equations. The theoretical properties of the algorithms, their relationship to methods previously proposed by Lim, and their application to a variety of signal models have been studied. Testing of the algorithm's performance on real data has just begun, and results are not yet available. However, because of their conceptual and algorithmic simplicity, as well as their solid theoretical basis, these algorithms promise to be a useful tool for signal and parameter estimation in the presence of noise.

12. WINOGRAD FOURIER TRANSFORM ALGORITHM (WFTA) IMPLEMENTATION

National Aeronautics and Space Administration (Grant NSG-5157)

Syed H. Nawab, James H. McClellan

Bounds on the minimum number of data transfers (i. e., loads, stores, and copies) required by WFTA and FFT programs have been derived. The analysis is applicable to those general-purpose computers with at least 4 general processor registers (e. g., the IBM 370, PDP-11, etc.). It was shown that the 1008-point WFTA requires about 21% more data transfers than the 1024-point radix-4 FFT; on the other hand, the 120-point WFTA has about the same number of data transfers as the mixed-radix ($4 \times 4 \times 4 \times 2$) version of the 128-point FFT and 22% fewer than the radix-2 version. Finally, comparisons of the "total" program execution times (multiplications, additions, and data transfers, but not indexing or permutations) were made.

Arithmetic concurrences, such as those found in special-purpose fast Fourier transforms (FFT) hardware, were surveyed and categorized. Similar structures were then derived for the Winograd Fourier transform algorithm (WFTA). Relative time-efficiency plots were obtained for the 1024-point radix-4 FFT and the 1008-point WFTA as a function of the number of real arithmic operations executable in parallel. This comparison showed that the relative time efficiency of the two algorithms in sequential computations generally carries over to cases where arithmic parallelism is exploited.

13. EVALUATION OF CIRCULARLY SYMMETRIC TWO-DIMENSIONAL FOURIER TRANSFORMS AND ITS APPLICATION TO THE MEASUREMENT OF OCEAN-BOTTOM REFLECTION COEFFICIENTS

U.S. Navy – Office of Naval Research (Contracts N00014-75-C-0951 and N00014-77-C-0196)

Alan V. Oppenheim, George V. Frisk, David R. Martinez
[G. V. Frisk is with the Woods Hole Oceanographic Institution.]
[D. R. Martinez is with the M.I.T.-W.H.O.I. Joint Program in Oceanography/Oceanographic Engineering.]

In a variety of applications the need arises for the evaluation of the two-dimensional Fourier transform of circularly symmetric functions. Because of the circular symmetry, the two-dimensional Fourier transform reduces to the Fourier-Bessel or Hankel transform. This research considers a method for evaluating this transform using the "projection-slice" theorem for multidimensional transforms. The method is applied specifically to the measurement of the plane-wave reflection coefficient of a horizontally stratified ocean bottom using the fact that, for a point source, the bottom-reflected field and the plane-wave reflection coefficient are circularly symmetric and are related through a two-dimensional Fourier transform.

14. SHORT-TIME FOURIER ANALYSIS

Michael R. Portnoff

Short-time Fourier analysis is based on the notion of a multidimensional representation for a one-dimensional signal. Specifically, a one-dimensional time signal, $x(t)$, is represented by a two-dimensional function of time and frequency, $X(t, \omega)$, called a short-time Fourier transform (STFT). In its simplest form, the STFT $X(t, \omega)$ is defined as

$$X(t, \omega) = \int_{-\infty}^{\infty} x(\tau)\, w(t-\tau)\, e^{-j\omega\tau}\, d\tau,$$

where $w(t)$ is a window function that is, in some sense, narrow in time or frequency, or both. In its more general form, the STFT is defined using a window that is allowed to depend on both time and frequency.

Short-time Fourier analysis is particularly useful for studying "slowly time-varying" phenomena such as speech, music, and other acoustic signals, because rapidly varying local features appear as functions of frequency in the STFT, whereas slowly-varying global features appear as functions of time. Thus, the STFT is a formal mathematical description for our notion of a "time-varying spectrum." Furthermore, short-time Fourier analysis is, in many ways, analogous to the acoustic processing performed by the human auditory system.

At present, short-time Fourier analysis is not well understood. The objective of our research, therefore, is to develop a better understanding of this method of signal analysis, both by investigating the mathematical properties of the STFT and studying the STFT for specific signal models.

15. ESTIMATION OF UNWRAPPED PHASE

U. S. Navy – Office of Naval Research (Contract N00014-75-C-0951)

Thomas F. Quatieri, Jr., Alan V. Oppenheim

The unwrapped phase estimation problem for discrete-time sequences was first encountered in the development of a mixed-phase homomorphic vocoder, where the smooth-phase estimate of the vocal-tract impulse response led to harsh-sounding synthetic speech. The sensitivity of the unwrapped-phase envelope to time-domain perturbations was observed to be greater than that of the log-magnitude spectrum. These observations initiated a number of questions and answers in magnitude/phase properties and relations, which are common to numerous areas of signal processing.

In particular, a theoretical framework was developed for unwrapped-phase estimation from harmonic spectra (voiced speech) through smoothing real and imaginary spectral components. Short-time homomorphic analysis and a short-time harmonic model have led to pitch-adaptive duration and alignment requirements on time-domain windowing. The underlying phase envelope is consequently preserved so that cepstral windowing can be applied. The result is a mixed-phase homomorphic vocoder of somewhat higher quality than its minimum-phase counterpart.

In addition, two alternative mixed-phase vocoders were considered: the first is based on linear interpolation of complex harmonic peaks, and the second on Lim's homomorphic spectral-root deconvolution scheme.

Current research encompasses the following three major topics:

(1) A general framework of phase estimation for a number of signal-processing applications, including speech, seismic, and oceanographic problems.

(2) More reliable methods of obtaining unwrapped phase from sampled and random data.

(3) Investigation of magnitude/phase relations and their relative sensitivities to perturbations.