XXII. 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, bloodflow, 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, speed transformation of speech, and enhancement of degraded speech. 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 a related area, the methods of speech compression using linear predictive encoding are being applied to the compression of data recorded in ocean bottom

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seismometers. These methods have been tested with data provided by the Woods Hole Oceanographic Institution and indicate that data recording requirements can be halved with little loss of fidelity.

For the problem of noninvasive 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 all-pass 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 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 and the estimation of formation parameters by first removing the modal behavior and other unwanted components from the measured signals. During 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 currently 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 and we are in the process of defining a research program which will involve a close collaboration between our group and Lincoln Laboratory.

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

Our new area of interest is the combination of signal processing and Artificial Intelligence techniques. There are a variety of problems in signal analysis and interpretation 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 beginning to explore this new area both generally and in the context of some specific applications.

1. THE PARABOLIC WAVE EQUATION

U.S. Navy — Office of Naval Research (Contract N00014-75-C-0852)
National Science Foundation Fellowship

Thomas E. Bordley, Arthur B. Baggeroer

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 reduces to a comparatively simple initial value problem for which the necessary initial conditions can be readily found. Second, since the split fields are decoupled approximants to the true transmitted and reflected fields, the approximate solution has a parabolic rather than a circular dispersion relation. Therefore, with increasing angle with respect to the direction of propagation, waves emanating from a point source become increasingly distorted and the absorption imposed on the signal becomes increasingly in error.

Such errors are of negligible importance in many marine problems because inhomogeneities in the water column normally are significant only when large distances are involved. Moreover, due to losses (primarily) associated with the presence of an attenuating bottom, only those waves which possess small grazing angles with respect to the bottom or, equivalently, which emanate at a nearly horizontal angle from the source, contribute substantially to the solution over large distances. Ordinarily then, those waves which contribute the most to the final solution are represented most accurately and those which contribute the least are represented least well.

When the signals propagating 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.

2. LINEAR SHIFT-VARIANT FORM-INVARIANT FILTERING

NATO Fellowship

Carlo Braccini

A system is defined to be form-invariant under a linear scaling of the input signal if the output is also linearly scaled, according to the same (or a different) scale factor of the input. This property is desirable in applications such as size-independent pattern recognition, tracking of moving objects, Doppler signal processing, and restoration of special optical distortions. Moreover, it characterizes the processing taking place in the peripheral auditory and visual system of man.

Form invariance cannot be achieved by means of linear shift-variant systems, so that the broader area of linear shift-variant processing must be considered. Within this area, there is a unique class guaranteeing the form invariance property, and it is the one characterized by an impulse response depending essentially on the ratio of input and output variables. For these systems, the representation tool, equivalent to the Fourier transform for LSI systems, is the Mellin transform.

Besides the theoretical properties, issues of implementing the Mellin transform and form-invariant systems will be investigated. One can exploit and generalize the concept that the Mellin transform of f(t) can be computed as the Fourier transform of $f(e^t)$. The effects of the approximated nonlinear mapping must then be accounted for in the discrete case, particularly in 2-D and inversion problems, for which more robust techniques should be explored.

Among various applications in pattern recognition and coding problems, there is one connected to the phase-only reconstruction techniques. These, in fact, may be improved by retaining some amount of amplitude spectrum information, determined on the basis of the space and frequency-variant smoothing performed by the visual system. Both the smoothing and its inverse should be implemented by form-invariant algorithms.

3. EVENT COMPRESSION WITH RECURSIVE LEAST SQUARES SIGNAL PROCESSING

U.S. Navy — Office of Naval Research (Contract N00014-75-C-0951)
National Science Foundation (Grant ECS79-15226)

Webster P. Dove, Alan V. Oppenheim

In various signal processing problems, signals arise which contain pulses or events that must be located or detected. Problems of this type include RADAR and SONAR rangefinding, speech pitch detection, and seismic data analysis. All of these fields share the need for a signal processing technique which can compress events occurring in the raw data into shorter events in the processed data. This type of processing can reduce the overlap between successive events leading to improved detectability and by increasing the impulsiveness of each event make locating the events easier. Such a signal processing procedure, which reduces the duration of events in an input sequence without changing their relative separations, is an event compression algorithm. Examples include homomorphic deconvolution, matched filtering, linear predictive deconvolution.

Approaching the problem of event compression typically entails modeling the physical situation in appropriate fashion and then designing an algorithm which is expected to solve the problem for data which fits that model. Subsequent investigation of the performance of the algorithm in a realistic environment may then lead to alterations in the model, the algorithm or both.

In this work we apply a recursive least square (RLS) adaptive linear prediction algorithm to event compression, using synthetic data from a data model based on an abstraction of a sonic well logging problem. Because the events in this data have independent spectra, event compression by linear time invariant filtering would not be effective (the inverse filter for each arrival would have to be different). By using an adaptive algorithm we perform time varying event compression on the input data sequence.

This work was completed in July 1980.

4. SEISMIC SIGNAL MODELING AND PROCESSING

U.S. Navy — Office of Naval Research (Contract N00014-77-C-0257)

Gregory L. Duckworth, Arthur B. Baggeroer

In the spring of 1980 a large-scale undersea and seabottom acoustic experiment was mounted from three ice camps in the East Arctic ocean. The results of the experiment are extremely high-quality multichannel digital data for long-range acoustic transmission and reverberation through the water column and seismic refraction studies. The current research work involves analysis of the refraction data, including the use of adaptive frequency/wavenumber spectral estimation techniques to obtain phase velocity spectra. A research goal is the application of parabolic approximation techniques to the elastic wave equation in the water column and sub-bottom strata in order to compute path transmission losses for a given geologic model. A comparison of these predicted amplitudes and those measured by the velocity analysis will aid in the inversion of the refraction data to obtain the geologic structure.

5. A NEW DESIGN OF PCM SPEECH CODING SYSTEM

Cengiz Esmersoy, Jae S. Lim

PCM coding is widely used in speech processing as well as in other areas, such as picture processing. Since a high bit rate requires more channel bandwidth, some techniques have been used to improve the quality without increasing the bit rate. In this research, a PCM speech coding system, which combines nonlinear quantization, adaptive quantization, and noise reduction techniques will be developed.

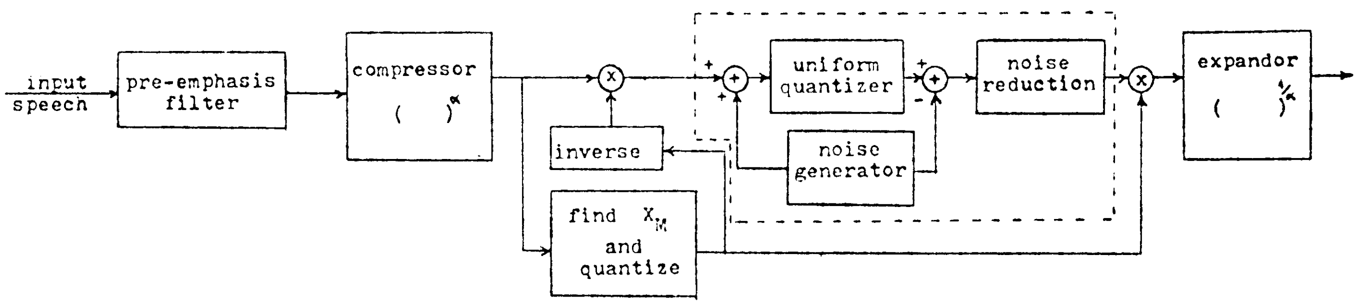

Fig. XXII-1. Block diagram of the overall system.

The system has a simple encoder. A high-class pre-emphasis filter precedes the whole system to prevent the loss of information in the high-frequency components of speech. From the probabilistic analysis of speech signals, it is well known that the lower amplitude levels are more likely to occur. Hence, by using nonlinear quantization, low amplitude levels can be represented with smaller errors by allowing larger errors for large amplitude levels. To perform nonlinear quantization, a compressor following the pre-emphasis filter and an expander at the end of the system are installed. One approach to accommodate the nonstationary nature of the speech waveform is adaptive quantization. The adaptation proposed in this research is somewhat different than the conventional schemes. The speech signal is divided into equal length segments each of 20-40 msec duration. The maximum absolute value of each segment is found and quantized. Then the segment is run once more and each sample of the segment is multiplied by the inverse of the quantized maximum. Hence, a comparatively more stationary signal is obtained which is to be PCM coded. Quantized maximum values are sent to the receiver attending the PCM signal, and each point of the segment is multiplied by the quantized maximum at the decoder. Since it is required that the gain parameter be sent only once for each segment, this will not increase the bit rate considerably. The subsystem in the dashed line is static, meaning that none of the parameters are changed for adaptation. The noise generator creates pseudo-random white noise in the range [-Q/2; Q/2] where Q is the quantization step size. At the output of the Roberts' system, signal independent additive noise is obtained. A noise reduction system, such as spectral subtraction, follows the Roberts' system. Different spectral subtraction techniques will be applied to achieve the best performance. Development of the optimum parameters is a topic of research. After the system is developed with optimum parameters, a subjective evaluation of the system performance relative to direct PCM will also be performed.

6. EFFECT OF NOISE IN SIGNAL RECONSTRUCTION FROM ITS FOURIER TRANSFORM PHASE

Bell Laboratories

Carol Y. Espy, Jae S. Lim

In general, a sequence cannot be recovered from its Fourier transform phase alone. However, under certain conditions, reconstruction of a sequence from its phase information alone is possible. One such set of conditions, which is not very restrictive in practice, was recently developed by Hayes, Oppenheim, and Lim. Specifically, it has been shown that a finite duration sequence which has no zeros on the unit circle or in conjugate reciprocal pairs is uniquely defined by a finite number of samples of the phase function. In addition, several algorithms were developed to reconstruct a sequence from its phase under such conditions.

Even though the results by Hayes, Oppenheim, and Lim have important theoretical significance, they are limited in practice since the results are based on the assumption that the <u>exact</u> phase is available. In many practical potential application problems, such as blind deconvolution problems or Fourier transform signal coding, the phase available is gradually degraded by measurement noise, quantization noise, etc. Thus, the purpose of this work is to investigate the effect of the phase degradation on the reconstructed sequence and to develop techniques that minimize the phase degradation effect.

7. SIGNAL RECONSTRUCTION FROM PHASE OR MAGNITUDE

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National Science Foundation (Grant ECS80-07102)

Monson H. Hayes, Jae S. Lim, Alan V. Oppenheim

For both continuous-time and discrete-time signals, the magnitude and phase of the Fourier transform are, in general, independent functions, i.e., the signal cannot be recovered from knowledge of either alone. Under certain conditions, however, relationships exist between these components. For example, when the signal is a minimum phase or maximum phase signal both the log magnitude and phase can be obtained from the other through the Hilbert transform. This relationship has been exploited in a variety of ways in many fields including network theory, communications, and signal processing. As a result of some research into the general problem of signal reconstruction from phase or magnitude, several sets of conditions have been specified for which a discrete-time sequence is uniquely specified, to within a scale factor, by the phase of its Fourier transform, without the restriction of minimum or maximum phase. The first set of conditions considers the case in which the phase is specified at all frequencies, whereas the second set considers the case in which the phase is specified only at a discrete set of frequencies. Once these conditions have been established, a dual set of conditions are presented for which a discrete-time sequence is uniquely specified to within a sign and a delay by the magnitude of its Fourier transform, again without the restriction of minimum or maximum phase.

For those sequences satisfying the appropriate set of conditions, three algorithms have been devised for reconstructing a sequence from its phase or samples of its phase. The first algorithm is conceptual in that it provides some insight into the fundamental results of phase or magnitude signal reconstruction. The remaining two algorithms are numerical and provide a practical method of performing phase-only signal reconstruction. The first numerical algorithm is an iterative technique in which the estimate of the desired sequence is improved at each iteration. The second is a closed-form solution for which the desired sequence is found by solving a set of linear equations.

Although stated only in terms of one-dimensional sequences, the results are readily extendible to multidimensional sequences. This extension is accomplished by utilizing a few well-known results on projection and slices. With this extension, a potential application for phase-only reconstruction is the blind deconvolution of images which have been blurred by a symmetric point-spread function.

Further research is expected to address such questions as the noise sensitivity of reconstruction techniques, the convergence of iterative solutions, the existence of more efficient algorithms that may be realistically applied to processing large amounts of data, alternate forms of the multidimensional theorems which do not depend on projection/slice ideas, and the extension of the known results to include additional constraints on the sequence (such as positivity).

8. FORMATION PARAMETER ESTIMATION IN SONIC WELL LOGGING

U.S. Navy — Office of Naval Research (Contract N00014-75-C-0951)
Schlumberger-Doll Research Center Fellowship

Andrew L. Kurkjian, Alan V. Oppenheim

The sonic well logging problem is modeled as an ideal point source in an infinite fluid cylinder surrounded by an infinite elastic solid. The source is located on the borehole axis and the source waveform is assumed to be known. An array of ideal receivers is placed on the axis at a known distance from the source. The borehole radius and parameters of the fluid are also assumed to be known. The overall problem is to estimate the parameters of the solid formation from the received signals.

The influence of the formation parameters on the sonic waveforms is not well understood even though the borehole model is exceedingly simple. It is known that the signals are greatly influenced by such uninteresting parameters as the borehole radius, the parameters of the fluid, and the source waveform. The problem of extracting the interesting information from these signals while rejecting the uninteresting is of central concern in this effort. In particular, the compressional and shear head waves should be more sensitive to formation parameters than the modes. This is because the head waves actually traverse a patch within the formation while modes represent energy trapped within the borehole. With the view that the modes are uninteresting arrivals in the sonic waveforms, we would like to design processing which will remove the model energy while leaving the head waves undisturbed. Such a processor would effectively remove the multiple energy from the waveforms, thus leaving only the first compressional and shear head waves and a Stoneley interface wave. There is evidence that the separation of head waves and modes is possible. The design of such a separation processor is the central focus of this work.

9. WAVENUMBER SPECTRAL ESTIMATION

National Science Foundation (Grant ECS79-15226)
Hertz Foundation Fellowship

Stephen W. Lang, James H. McClellan

Wavenumber spectral estimation is a problem of some importance in radar and sonar applications. An array of sensors is used in conjunction with a suitable signal processing algorithm to provide an estimate of incident power as a function of wavevector. This estimate constitutes the radar's "picture" of the world.

The application of modern spectral estimation algorithms to this array processing problem is currently of great interest. These new algorithms generally use parametric models of the spectrum which are tailored to specific applications. Unfortunately, many of these algorithms are formulated for evenly sampled, one-dimensional time series, and their application to the multidimensional, unevenly sampled array problem often proves difficult.

Research is being done on the theoretical foundation of an important class of modern spectral estimation algorithms, the correlation extension techniques such as the maximum entropy method and Pisarenko's Method, in this array processing context. In addition, practical algorithms for the computation of certain estimates and design techniques for array configurations are being developed.

10. TIME DELAY ESTIMATION OF MULTIPLE ARRIVALS

Yen-ta Li

Time delay estimation is an important problem in seismic signal processing. Work has been done on time delay estimation of multiple arrivals without noise. For example, one unknown wavelet arrives at each of two receivers an arbitrary number of times. Each arrival has an arbitrary attenuation factor; the two received time series are otherwise noise free. The problem is to determine the relative arrival times and relative attenuations at the two receivers. The conventional time delay estimation techniques, such as those based on the crosscorrelation of the two received signals, do not perform well in the severely overlapped multiple arrival problem. The actual relative arrival time and attenuation information is seen to be "smeared."

A new method, which uses only the phase of the cross power spectrum, has been found to produce the correct relative arrival times and attenuations with no "smearing." An example of this method applied to synthetic data is provided. At present, we are working on the time delay estimation of multiple arrivals in the noisy case.

11. TWO-DIMENSIONAL MAXIMUM ENTROPY POWER SPECTRUM ESTIMATION

U.S. Navy — Office of Naval Research (Contract N00014-75-C-0951)
Government of Pakistan Scholarship

Naveed A. Malik, Jae S. Lim, Alan V. Oppenheim

The applications of 2-D Power Spectrum Estimation (PSE) are widespread and cover such diverse areas as seismic signal processing, image restoration, radar, sonar, and radio astronomy.

As part of this continuing research, various 2-D PSE techniques such as conventional periodogram estimates, autoregressive signal modelling, etc. have been considered and studied. The major result has been the development of a new algorithm[1] for solving the highly nonlinear 2-D maximum entropy PSE problem.

The initial results obtained via the interactive algorithm are extremely encouraging and the algorithm promises to open up several hitherto unexplored areas of research. The algorithm is completely general, and therefore applicable to signals of any dimensionality. The properties of the techniques are to be studied and thoroughly characterized.

References

1. J.S. Lim and N.A. Malik, "A New Algorithm for 2-D Maximum Entropy PSE," Lincoln Laboratory Technical Note No. 1980-37, August 7, 1980.

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 (Contract N00014-77-C-0196)

Douglas R. Mook, Alan V. Oppenheim, George V. Frisk
[G.V. Frisk is with the Woods Hole Oceanographic Institution.]

The determination of the reflection coefficient of the ocean floor 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.

13. OPTIMAL SIGNAL ENHANCEMENT AND PARAMETER ESTIMATION IN THE PRESENCE OF NOISE

Bruce R. Musicus, Jae S. Lim, Alan V. Oppenheim

Until recently, the problem of enhancing signals such as speech or images which are degraded by added noise, has been treated by various ad hoc and only moderately successful means. If a valid mathematical model for the signal and the noise is known, perhaps with a few unknown parameters, then we have shown that there are at least three different maximum likelihood approaches to optimally estimating the signal and parameters. Unfortunately, for most signal models of interest, all three approaches require solving a complicated and time-consuming nonlinear optimization problem.

Recent work by Musicus and Lim has shown that for certain signal models these problems can be solved exactly by indirect, iterative techniques. If the signal can be modeled as stationary and infinitely long, with no other a priori structure, the "optimal" maximum likelihood estimation technique can be shown to correspond to a "correlator-subtractor" scheme. When pole-zero signal models are assumed, it can be shown that iterating between a linear filtering step and a linear pole-zero parameter fitting step converges to the correct answer. Adaptive recursive versions of the algorithms have also been proposed, in which the filtering and parameter fitting steps adapt to the data as it arrives. Theoretical analysis and performance testing of these algorithms is now in progress.

14. NOISE REDUCTION IN SPEECH BY SPECTRAL SUBTRACTION

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

Cory Myers, Alan V. Oppenheim, Jae S. Lim

The processing of a signal corrupted by additive noise in order to remove the effect of the noise is an important step in many signal processing tasks. In particular, noise reduction is an important step in many speech processing applications, including speech coding and transmission and speaker and speech recognition.

One reasonably successful approach to the problem of noise reduction has been the use of spectral subtraction (also referred to as correlation subtraction). It is the purpose of this research to examine extensions of the correlator subtractor method for noisy speech signals. The areas which are to be examined include the use of multiple pass processing, various windowing strategies, and methods of averaging several different processed versions of the same signal. These methods have been shown by Nawab et al., to be successful in the reduction of additive noise in image processing. Further extensions of the correlator subtractor method may also be examined.

15. SIGNAL NOISE REDUCTION

U.S. Navy — Office of Naval Research (Contract N00014-75-C-0951)
U.S. Air Force (Contract F19628-81-C-0002)

Syed H. Nawab, Alan V. Oppenheim, Jae S. Lim

This research is concerned with the problem of additive white noise reduction in signals generally, and images in particular. We are particularly interested in signals that are "deterministic" in the sense that filtering procedures, based on the assumption that the signal is a sample of a stationary stochastic process, are inadequate. Image noise reduction problems often fall into such a category.

We approach the problem with the view of developing a mathematical framework for dealing with the filtering of the above-mentioned signals. It is hoped that this leads to adaptive time-varying filtering procedures that have a firm mathematical basis in signal theory.

16. IMPORTANCE OF PHASE IN SPEECH ENHANCEMENT

Hughes Aircraft Company Fellowship

David L. Wang, Jae S. Lim

Conventional speech enhancement systems based on direct estimation of the short-time spectral amplitude are well known. Various enhancement systems based on such methods have been proposed. The general assumption in these systems is that short-time phase is relatively unimportant. Thus, the phase of the original speech signal is approximated by the phase of the noisy speech signal. Recent results suggest that phase distortion due to additive noise contributes to a reduction in intelligibility and seems to indicate that improvement in phase estimation could improve the performance of speech enhancement systems using conventional spectral subtraction techniques. This thesis will focus on human perception of phase distortion, and detailed measurements of the improvement of quality and intelligibility of speech signals using high-resolution phase estimation will be performed.