XVIII. DIGITAL SIGNAL PROCESSING

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1. TWO-DIMENSIONAL DIGITAL FILTER STRUCTURES

David S. K. Chan, James H. McClellan

The objective of this research is to develop insight into two-dimensional digital filter structures by synthesizing new structures and comparing them with existing structures. Such an undertaking is important for efficient implementation of 2-D filters and in the design of special structures that might permit real-time processing or distributed processing.

Contrary to the effect in the one-dimensional case, the signal flow graph has been found inadequate for representing 2-D filter structures. It fails to characterize how data are sequenced through a structure. Information of this kind is important because the order in which computations are performed can seriouly affect factors in a structure such as storage requirements and precedence relations.

We have been developing a more general framework for characterizing filter structures, and our work has resulted in a new representation based on a state-space approach.[1] This representation incorporates in the description of a structure the order in which data are computed, and, in a natural manner, also describes precedence relations between operations, an intrinsic part of any filter implementation. Such precedence relations characterize the inherent limitations of a structure with regard to parallel processing, and hence are important in considering such issues as pipelining and space/time tradeoffs.

This new representation also casts light on the 1-D filter implementation problem, and we are now investigating ways in which it may be applied to analysis and synthesis of 1-D and 2-D digital filter structures.

References

1. D. S. K. Chan, "A Novel Framework for the Description of Realization Structures for 1-D and 2-D Digital Filters," EASCON '76 Record, IEEE Electronics and Aerospace Systems Convention, Washington, D. C. , September 26-29, 1976, pp. 157A-157H.

2. J. H. McClellan and D. S. K. Chan, "A New Structure for 2-D FIR Filters Designed by Transformations," Tenth Annual Asilomar Conference on Circuits, Systems, and Computers, Pacific Grove, California, November 22-24, 1976.

2. RECONSTRUCTION OF VELOCITY STRUCTURES FROM TELESEISMIC FIRST ARRIVAL TIMES

David B. Harris, James H. McClellan

Many geophysical investigations require a knowledge of the spatial variation of the velocity of wave propagation in the earth's crust and upper mantle. Some aspects of earthquake prediction and determination of tectonic features fall into this category. Information about velocity structure on this scale is only available from measurements made at the earth's surface, for example, from measurements of the first arrival times of teleseismic P-waves recorded at an array of seismometers.

The purpose of this project is to determine the feasibility of applying the theory of reconstruction of functions from their projections to the problem of reconstruction of velocity structures from recorded first arrival times. The basis of this idea is that the arrival time of a P-wave at a seismometer is given by a line integral of the reciprocal velocity function along a path terminating at the seismometer. Thus, the first arrival times recorded at an array of seismometers constitute approximately a projection of the inverse velocity function onto the surface. Multiple projections from P-waves at different angles of incidence can be used to reconstruct the Fourier transform of the reciprocal velocity function. This may be inverse transformed to compute the velocity structure.

An implicit assumption of this technique is that the incoming waves have plane-wave structure. The effects of deviations from plane-wave structure on the quality of reconstructions are being investigated by using theoretically computed first arrival times.

3. APPLICATION OF HOMOMORPHIC FILTERING TO SEISMIC DATA PROCESSING

José M. Tribolet, Alan V. Oppenheim

Homomorphic filtering is a nonlinear signal-processing technique that has been applied to various deconvolution problems. The aim of this research is to study its application to the "seismic deconvolution problem." Seismic data can often be modeled on a short-time basis as a convolution of a bandpass wavelet p(n) with a sequence of impulses r(n). Letting w(n) represent a short-time window, we may represent a seismic trace segment x(n) as

$$x(n) = w(n)[p(n) * r(n)].$$

The seismic deconvolution problem is to recover the sequence of impulses $r[n]$ from which the potential determination of the depths of the subsurface reflectors is possible.

Short-time seismic data models exhibit characteristics that have to be accounted for carefully in terms of their effects on homomorphic signal analysis. For example, their bandlimited nature has led to the generalization of homomorphic systems for this class of input signals.

The short-time model has led to an understanding of windowing effects in the cepstral domain, which, when conveniently explored, enable effective homomorphic wavelet estimation by means of low-time cepstral gating. The corresponding estimates may then be used to design optimum lag Wiener spiking filters that ultimately resolve the reflector series $r[n]$. This technique has been tested with good results on synthetic seismic data.

4. ENHANCEMENT OF DEGRADED SPEECH

Jae S. Lim, Alan V. Oppenheim

The objective of this project is to develop speech enhancement techniques to increase the intelligibility and quality of degraded speech when the degradation is caused by addition of random noise. This research began last year and, as a starting point in our research, we considered two existing speech-enhancement techniques, comb filtering and the INTEL (Intelligibility Enhancement by Liftering) system.

The comb-filtering technique capitalizes on the periodic structure of speech waveforms and attempts to eliminate the frequency bands where speech contributes little

energy. By using this elimination process, we hope that more noise than speech will be rejected. In our laboratory, some modification has been made on the adaptive comb-filtering method[1] and the modified system was implemented on the PDP 11/50 computer. To determine the effect on the intelligibility score with the use of this system, nonsense sentences were processed and an intelligibility test is now in progress.

The INTEL system is based on the concept that speech and noise contribute differently to the autocorrelation function. In our laboratory, the original INTEL system[2] was reduced to a simpler form, which resulted in two important advantages: a 40% reduction in computation time, and the conceptual simplicity of a less complex system. The system was implemented on the PDP 11/50 computer. As before, to determine the effect of the system on the intelligibility score, we processed nonsense sentences and are testing their intelligibility.

References

1. R. H. Frazier, "An Adaptive Filtering Approach toward Speech Enhancement," S.M. and E.E. Thesis, Department of Electrical Engineering and Computer Science, M.I.T., June 1975.

2. M. R. Weiss et al., "Study and Development of the INTEL Technique for Improving Speech Intelligibility," Final Report No. NSC-Fi/4023, Nicolet Scientific Corporation, Northvale, New Jersey, December 1974.

5. ENHANCEMENT OF LOWPASS FILTERED SPEECH

Elliot Singer, Alan V. Oppenheim

This project is concerned with enhancing the quality of lowpass filtered speech by reinserting the missing spectral information. When only the low-frequency portion of the signal is present, it is possible that much of the missing high-frequency structure can be determined from an examination of the available spectral energy and thus the original speech can be reconstructed. This is especially true for voiced speech where the steady-state frequencies and amplitudes of the formants are well established.

In a broad sense, an enhancement system for lowpass filtered speech would incorporate algorithms for deducing the missing high-frequency structure and processing schemes for synthesizing the enhanced speech signal. Thus far, in our research we have concentrated on techniques that keep the high-frequency characteristics fixed in time, and hence adaptive filters are not required.

In order to achieve the most natural-sounding speech output, it was necessary to make use of the available signal as directly as possible. This principle has been applied to voice-excited vocoders with considerable success. In this system, the speech

synthesis is performed by extracting a subband of the original speech and processing it to make it a suitable excitation to the synthesizer. Linear prediction techniques may be used to flatten spectrally a lowpass speech signal whose spectrum has been broadened through rectification. Theoretically, this approach should produce a signal whose spectral harmonics are closely related to those of natural speech.

This technique was applied to an all-voiced sentence that had been lowpass filtered to 2 kHz. The resulting processed signal was superior in quality to the unenhanced original but suffered from a good deal of hoarseness. We believe that this hoarseness is attributable in part to the effects of spectral broadening and the manner in which the linear prediction process operates on the speech signal.

6. SPEECH ANALYSIS-SYNTHESIS BASED ON HOMOMORPHIC FILTERING AND CCD TECHNOLOGY

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

A nonreal-time speech analysis-synthesis system based on homomorphic filtering and the real cepstrum has been completed. We are developing a modification of this simulation within the context of Charge Coupled Device (CCD) technology. In order to improve this system, we are incorporating the complex cepstrum into the homomorphic algorithm and studying the effects of phase on the quality of the synthesized speech. One important result is the sensitivity of the phase estimation to the time-domain window duration, shape, and onset. This work should lead to a fixed phase compensation, which can be implemented by CCD technology.

The FFT algorithm will be replaced by a skewed spectral analysis, the sliding chirp-z-transform, which is also well-suited to CCD technology. The sensitivity of this new spectral technique to the nonstationarity of the input speech waveform is being examined.

We are also investigating novel methods of filtering the log spectrum in frequency and the cepstrum in quefrency, which should reduce the raucous quality of vocoder speech.

7. SPEED TRANSFORMATIONS OF SPEECH SIGNALS

Michael R. Portnoff, Alan V. Oppenheim

We have designed and implemented on our PDP-11 computer a speech analysis-synthesis system based on the discrete short-time Fourier transform. This system represents a speech signal by an appropriate set of time-variant parameters and has

the property that when it is time-scaled and used in the synthesis procedure it produces speed-transformed speech. We are now investigating two possible synthesis techniques and beginning our investigation of feature-dependent speed transformations by selectively transforming only the stationary portions of the speech signal.