DIGITAL SIGNAL PROCESSING, TODAY AND TOMORROW

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1. Introduction

There has been a clear trend over the past several years toward the increased use of digital rather than analog processing of signals, in a large part because of the inherent flexibility and reliability of digital processing. In many cases, speed and cost make the use of all-digital equipment impractical for real-time processing, but even in these cases the design of processing algorithms through digital simulation on a general purpose computer is useful and important. Faster digital hardware with lowering cost and size is constantly becoming available, because of the continuing development of integrated circuit technology. Thus, signal processing using digital hardware will continue to play an increasingly important role.

The applications for digital processing techniques can be drawn from a variety of areas. Speech waveform processing for automatic speech recognition, bandwidth compression, linguistic studies, etc., often requires the use of sophisticated techniques that are most easily implemented digitally. Component speed and costs now make radar signal processing using digital techniques practical and advantagous. Similarly, fast, general-purpose signal processing computers permit the implementation of digital techniques for processing two-dimensional analog signals, arising, for example, in seismology, and in the enhancement of photographs, medical x-rays, and electron micrographs.

Along with a growing list of applications, of course, there is a growing list of problems to be solved. Some of these relate to theoretical considerations such as finite word-length effects, while others are directed toward such hardware issues as architectural structures for digital signal processing computers. In this paper, several applications of, and problems in, digital signal processing are reviewed.

2. Applications of Digital Signal Processing

One of the early applications of digital signal processing was in the area of speech processing for speech compression, speech synthesis and speech and speaker recognition. The basis for many speech compression systems is a separation of the spectral envelope, which characterizes the vocal-tract impulse response, and the spectral fine structure which characterizes the vocal-tract excitation.[1] Much of the current work on digital filter design techniques grew out of a desire to simulate analog filter banks digitally to determine appropriate specifications for the filters used to achieve this separation.[2,3] Since that time, the disclosure of the fast Fourier transform algorithm for computing the discrete Fourier transform has demonstrated some advantages in speech compression and has focussed attention on the possibility of all-digital speech compression systems.[4-6]

For speech synthesis the vocal-tract impulse response is approximated as the response of a set of resonant circuits whose resonances coincide with the resonances of the vocal cavity. The vocal cavity is a distributed system, and consequently has an infinite number of resonances. Thus speech synthesizers built using analog elements have included a higher pole correction network which is introduced to raise the lower part of the spectrum in a manner approximating the effect in that band which is due to the higher resonances in the vocal cavity. In contrast, a digital resonator behaves as an infinite number of resonances because the spectrum is periodic. In other words, a digital system is inherently better than an analog filter in modeling a distributed system. In particular, it has been shown that in speech synthesis using digital resonators rather than analog resonators no higher pole correction needs to be introduced.[7]

Another class of applications that appears suited, at present, to digital processing techniques is signal processing for radar systems directed at obtaining both range and velocity information arising, for example, in weather radar and ground-mapping radar systems.[8] In systems of this kind a radar return to be processed consists in the return from a number of transmitted pulses. The time delays between transmission and reception of the pulses provides range information and the frequency shifts of the pulse carrier provides velocity information. A set of pulses comprising a return are demodulated and characterized in terms of amplitude and phase shift of the carrier signal. A spectral analysis of this set of numbers then provides Doppler information. Typically, such an analysis is carried out in each of the range gates provided for in the radar system. This means that the returns in each range gate must be spectrum-analyzed and velocity and range information then determined from the resulting spectrogram. An analog implementation of this kind of system requires analog delay, since all pulses must be available for the processing, and in addition a separate filter bank must be used for each range gate. In contrast, for digital processing the delay is easily obtained in terms of digital memory and one filter bank can be easily multiplexed to accommodate all of the range gates. An efficient alternative to the implementation of a digital filter bank is the use of the fast Fourier transform for implementing the spectral analysis. The issues then in the case of radar signal processing for the trade-off between the use of analog or digital hardware focus primarily around the question of storage and multiplexing.

An area of applications which appears to have considerable potential for the future is that of multidimensional signal processing. For two-dimensional image enhancement the use of digital processing techniques appears to be considerably more flexible than optical techniques, although processing times on present available signal processing systems generally do not permit interactive image enhancement.[9] One of the real difficulties in multidimensional signal processing, however, is not so much in the tools for implementation but rather the lack of a deep understanding of the theoretical basis and a lack of adequate signal processing techniques.[10] Three-dimensional signal processing arises, for example, in reconstruction of three-dimensional objects from two-dimensional projections.

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This problem has arisen in the use of the electron microscope for studying molecular structures.[11] Because of the large depth of focus of the electron microscope, the two-dimensional image is a projection of the three-dimensional structure. By obtaining a number of projections for different angular orientations of the structure, it is possible to reconstruct slices through the object. The theoretical basis for the reconstruction is that the two-dimensional transform of the projection is a slice through the three-dimensional transform of the original structure. If enough such slices are obtained, the original structure can be reconstructed. As would be expected, the procedure is extremely difficult and slow to implement computationally. Furthermore, the theoretical basis derives from a consideration of continuous multidimensional processing, and a corresponding theory based on the use of the discrete Fourier transform is still undeveloped.

3. Some Problems in Digital Signal Processing

Along with a growing number of applications there is a long list of theoretical and practical problems that need to be faced. In digital filter design and implementation a continuing problem is the effect of finite register length both for the arithmetic and for the filter coefficients.[12-14] A related problem is that of choosing digital network structures that are least sensitive to coefficient inaccuracies, and of designing filters in a manner that takes account of the finite word-length constraint. Because of the present and future potential for real-time digital filtering hardware, these issues include number representations and modularity as introduced by the hardware considerations.[15]

One of the areas in digital signal processing that requires constructive theoretical thinking is in multidimensional digital signal processing. General design techniques for two-dimensional recursive and nonrecursive filters are not well developed. Nonrecursive two-dimensional filters generally are implemented using the fast Fourier transform algorithm and the filter characteristics are specified in the frequency domain. Such an implementation is restricted to nonrecursive filters. The primary drawback with an implementation of this type is the computation time required for the two-dimensional transform and inverse transform. Consequently, it is often desirable to implement the filtering directly in the spatial domain. The design of one-dimensional filters can be carried over to two-dimensional filtering if the filter is restricted to be separable. Separable filters are generally not satisfactory, however, because they tend to have poor characteristics in directions other than along the axes on which they are separable. At present, no effective design procedures are available for recursive, nonseparable two-dimensional filters. Furthermore, it is not known how to include a stability constraint in such procedures.[16]

In many cases, multidimensional discrete signals are thought of in terms of a rectangular coordinate system. For spectral analysis in particular, a discrete multidimensional signal on a rectangular grid has the advantage that the fast Fourier transform algorithm can be applied to computing the multidimensional transform. Situations often arise, however, in which data are obtained in a different coordinate system. For example, in the three-dimensional reconstruction problem described above, the three-dimensional transform obtained by computing the two-dimensional transform of the projections is in cylindrical coordinates. Since the reconstruction involves inverse transforming, the FFT algorithm can only be used if the data in cylindrical coordinates are interpolated onto a rectangular grid, a time consuming computation. Another example arises in the scanning of pictures using a spiral scan, in which case the data are essentially on a polar grid. Perhaps algorithms like the FFT algorithm can be devised for handling data in such coordinate systems.

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