XV. DIGITAL SIGNAL PROCESSING

Academic and Research Staff

Prof. A. V. Oppenheim
Prof. J. Allen
Dr. R. M. Mersereau
T. Aoyama

Graduate Students

R. E. Crochiere
D. E. Dudgeon
H. Hersey
M. R. Portnoff
J. M. Tribolet
E. Vincent
V. W. Zue

RESEARCH OBJECTIVES AND SUMMARY OF RESEARCH

The general research aims of this group are directed toward the development and application of techniques for digital signal processing and digital filtering. There has been a clear trend over the past several years toward increased use of digital rather than analog processing of signals, largely because of the inherent flexibility and reliability of digital processing. With the continuing development of integrated circuit technology, faster digital hardware with reduced cost and size is constantly becoming available. Digital signal-processing techniques have found application in a wide variety of areas including speech and picture processing, radar and sonar signal processing, and seismic data analysis. A summary of some of our present research projects and plans follows.

1. Digital Network Theory and Filter Structures

National Science Foundation (Grant GK-31353)

R. E. Crochiere, A. V. Oppenheim

When implementing digital filters on large general-purpose computers, with floating-point arithmetic or long fixed registers, it is generally not necessary to pay close attention to the filter structure that is used. In contrast, when a digital filter is implemented in special-purpose hardware, speed and filter behavior are closely related to the type of structure used to implement the filter. In the past several years, a considerable amount of work has been done by our group and others with regard to developing and analyzing different filter structures. These include, for example, the analysis of digital wave structures and the development of a filter analysis program that permits the analysis of arbitrary structures. Using this program, we have analyzed and compared several structures. In addition, we have developed a theoretical basis for the design of digital filters to optimize the word length.

Many techniques utilized in analyzing and comparing digital filter structures are based on a somewhat more general theory relating to the properties of digital networks or signal flow graphs. Tellegen's theorem for signal flow graphs, for example, leads to an important method for analyzing the sensitivity of filter structures, and this has been utilized in our filter-analysis program. In a more general sense, we would hope that the design of a filter structure with good finite register length characteristics could be incorporated directly into the filter-design problem. As an alternative, perhaps it would be possible to modify a filter structure in the direction of minimizing the coefficient sensitivity. Some indication about how this might be done is suggested by such network theorems as Tellegen's theorem, and it seems clear that as a better understanding of digital network theory develops, the ways in which filter structures can be modified will also become clearer.

At present, in comparing filter structures, the primary emphasis is on their evaluation in terms of coefficient sensitivity and arithmetic round-off effects. In considering the details of digital filter implementation, however, a variety of other issues are at least equally important. For instance, digital ladder structures appear to have good properties with respect to coefficient sensitivity. On the other hand, certain examples of these structures require all serial computation, that is, much of the arithmetic cannot be carried out in parallel. Since the degree of parallelism is generally related to computation speed, in some applications it may be attractive to sacrifice coefficient sensitivity to gain parallelism in the structure. Consequently, in evaluating filter structures, it seems important also to focus on other aspects of the structure such as the inherent parallelism. Issues such as this again relate to the topology of the digital network and fall within a framework that could appropriately be termed digital network theory.

2. Multidimensional Signal Processing

U. S. Navy Office of Naval Research (Contract N00014-67-A-0204-0064)

National Science Foundation (Grant GK-31353)

D. E. Dudgeon, R. M. Mersereau, A. V. Oppenheim

A class of problems that is becoming increasingly important and interesting involves two-dimensional digital processing. During the past year, we have completed a project directed toward the reconstruction of a multidimensional signal from its projections. In this work, a theorem has been determined that states that under specific conditions a reconstruction can theoretically be made from a single projection. We are investigating design techniques for multidimensional digital filters. In image processing, for example, digital filtering is important in such problems as reducing noise, inverse filtering, matched filtering for pattern recognition, contrast enhancement, and dynamic range compression. In contrast with one-dimensional filters, however, the problem of designing two-dimensional digital filters is for the most part unresolved. The difficulties arise from the fact that the system function is a ratio of two-dimensional polynomials, for which factorization theorems do not exist as they do for one-dimensional polynomials. Consequently, two-dimensional system functions cannot simply be characterized in terms of poles and zeros. Thus stability is considerably harder to guarantee. In one dimension, the approximation problem is generally concerned with the design of a squared magnitude function which is then factored. In two dimensions, the factorability of magnitude squared functions cannot be guaranteed. Furthermore, the general theory of approximation by two-dimensional polynomials is considerably less developed than in the one-dimensional case. Thus the general area of two-dimensional digital filtering presents a number of interesting research problems that we propose to pursue.

Our recent efforts have been directed toward developing algorithms that can be used to design magnitude squared functions that approximate a given ideal frequency response. An algorithm has been developed that provides an optimum approximation in the Tchebychev (minimax) sense on a limited pointed set. We shall continue this research and also investigate the problem of implementing these designs recursively. Since no two-dimensional factorization theorem exists which allows straightforward splitting of these magnitude squared functions to obtain recursible systems, we are looking for other techniques that will perform this splitting for us. Ironically, our two-dimensional recursive filter design algorithm has also proved useful for the design of one-dimensional filters. We are exploring the use of this algorithm for one-dimensional filter design.

Another approach to two-dimensional systems that we are exploring is that of mapping the two-dimensional problem to a one-dimensional problem, performing the signal processing in one dimension, and then mapping the output back to two dimensions. Such mappings exist, and are invertible. This approach is potentially rewarding because of the wide variety of available one-dimensional signal-processing techniques. Such an approach to processing two-dimensional signals is ideally suited to the processing of drum-scanned images and televisionlike signals, which are already somewhat one-dimensional in character.

3. Applications of Digital Signal Processing to Speech

U. S. Navy Office of Naval Research (Contract N00014-67-A-0204-0064)

A. V. Oppenheim, M. R. Portnoff, J. M. Tribolet, V. W. Zue

In this aspect of our research, we are interested in the applications of digital signal-processing techniques to speech processing and parameter extraction. One project is directed toward studying in detail and summarizing in quantitative terms the acoustic properties of various speech sounds within and around stressed syllables. Thus far, our effort has been directed primarily toward those consonants preceding stressed vowels. The objective of the study is at least twofold. First, we aim to provide quantitative data of these speech sounds for immediate applications in such research areas as computer recognition of speech and speech synthesis by rule. Second, through a controlled study of this kind we hope to arrive at a better understanding of the encoding and decoding processes in the speech communication chain.

Most of our effort has been directed toward the development of a good acoustic analysis system. A speech analysis system centered around linear prediction analysis has been implemented on a computer. This system has a highly interactive facility for online data display and playback.

At present, linear prediction analysis is limited to extracting the poles of the vocal-tract transfer function. Another aspect of our research is to investigate the extension of this set of techniques for the determination of zeros, as well as poles. One such possibility, which we are now investigating, is based on applying linear prediction to the cepstrum, a technique that we have referred to as cepstral prediction.

A third project, directed toward the application of digital signal processing techniques to speech, concerns speed transformations. There are many applications in which it is of interest to modify speech in such a way that its speed is changed while retaining an intelligible, natural quality. In learning a second language, for example, it is common practice to have exercises available at several speeds, depending on the level of the student. An eventual objective in learning a second language is comprehension at natural speaking speeds. Preliminary training, however, is often carried out at slower speeds. At present, it is necessary to record the exercises at each of several speeds. It would appear to be of considerable advantage if the speed transformation could be implemented automatically and continuously under the control of the student. Another class of applications arises in the consideration of reading machines. It is commonly understood that it is possible to read text faster than narrating it. In many instances, however, it is more convenient to receive information aurally. Other researchers have demonstrated that speed transformations of speech can be implemented by factors of two or three without destroying the intelligibility. In the other direction, it is sometimes useful to reduce the speed of speech so that otherwise unintelligible speech becomes intelligible.

One approach is the use of a vocoder which carries out the speed transformation by time-scaling the synthesis parameters. In principle, this is a reasonable approach, which results in intelligible speech. The speech quality is degraded, however, because of the degradation inherent in the vocoder. While this degradation is tolerable in some applications, in many, such as second-language learning, it is not. For this approach to be successful, an extremely high-quality speech analysis-synthesis system must be developed. The specifications for the system constitute a departure from the traditional point of view, in that efficient coding and bit-rate reduction is not a consideration. One aspect of our research is to develop such a system.

Even with a high-quality analysis-synthesis system, some important considerations remain. To a first approximation, speech transformation can be implemented by time-scaling the synthesis parameters. In speaking at different speeds, however, we do not simply change the speed of motion of the articulators, but, in fact, employ a set of rules. For example, for vowels it is reasonable to assume that a simple time-scaling is applied. For plosives, on the other hand, the duration of the burst will remain approximately constant. As a further consideration, coarticulation effects vary with speed and we often employ contractions to effect a speed change. Many of these rules, at present, are not well understood and will comprise one aspect of the research.

4. Digital Signal Processing Computer (The Black Box)

U. S. Navy Office of Naval Research (Contract N00014-67-A-0204-0064)

J. Allen

A small fast processor called the "Black Box" is being built primarily for use as a peripheral processor for speech synthesis with a DEC PDP-9 host computer. With the exception of the host computer I/O interface, the detailed logic design of the Black Box digital signal processing computer is complete.

Following completion of construction, and hardware debugging, a set of test programs will be written. We also plan to write a Black Box simulator on the PDP-9 computer to check the detailed logical functioning of the machine. This is particularly important for the Black Box multiplier, which is very complex, and must be checked with all possible bit patterns. During this period, we shall also write application programs. The first example will be a vocal-tract model, but an FFT will also be written, and software for several functions, such as vector scalar and display generation, will be completed. We expect that a large number of users will find the Black Box useful for their applications.