Digital Waveform Processing and Recognition

Editor

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PREFACE

There is strong interest and numerous activity in digital signal processing and digital pattern recognition and their applications. Many industrial, academic, and government facilities are concerned with both processing and recognition of waveforms. A need clearly exists to integrate both digital signal processing and digital pattern recognition into a single volume book such that the common, as well as interrelated methodologies in *both* processing and recognition can be developed and applied to various applications.

The present volume is written with these objectives in mind. Chapters 2 and 3 are concerned with the basic principles of digital filtering and spectral analysis along with discrete detection and estimation, as well as some illustrative examples using speech and biomedical waveforms. Chapters 4 and 5 review the statistical and syntactic pattern recognition with application to signal processing. More detailed discussions on applications to speech, geophysics, sonar, and radar are presented in Chapters 6 to 10. Certain aspects of digital system implementation for waveform processing and recognition are considered in Chapter 11. The book should serve a dual purpose of reference and textbook in theory, design, and applications of digital waveform processing and recognition.

It is my pleasure to acknowledge the important contributions to this book by Professors K. S. Fu, D. G. Childers, and Alistair D. C. Holden. In addition, I express my appreciation to the Air Force Office of Scientific Research and the Office of Naval Research for their sponsorship of my research activities in pattern recognition and statistical signal processing.

C. H. Chen

THE EDITOR

Chi-Hau Chen received the Ph.D. degree in electrical engineering from Purdue University, West Lafayette, Ind., in 1965. From 1965 to 1968, he worked with ADCOM, Inc. and AVCO Systems Division, both in the greater Boston area, on various projects in digital communications and statistical data processing. Since 1968, he has been a member of the faculty of Southeastern Massachusetts University, North Dartmouth, where he is a Professor of Electrical Engineering. He teaches graduate courses in Digital Signal Processing, Pattern Recognition, Communication Theory, Speech Sonar and Seismic Signal Processing, Signal Detection Theory, and Time Series Analysis. His recent and current research interests have been in Statistical Pattern Recognition, Seismic Signal Processing and Discrimination, Discrete Orthogonal Transforms, Imagery Processing and Recognition, and Detection and Estimation Theory.

He has published over 100 technical papers in the areas of communications, pattern recognition, and signal processing. He is the author of the book, *Statistical Pattern Recognition*, Hayden Book Co., 1973; and editor of the books *Pattern Recognition and Artificial Intelligence*, Academic Press, Inc., 1976, and *Computer-Aided Seismic Analysis and Discrimination*, Elsevier Scientific Publishing Co., 1978. He was the director of the 1978 NATO Advanced Study Institute on Pattern Recognition and Signal Processing held in Paris, and editor of its Proceedings, published by Sijthoff & Noordhoff International in 1978.

Dr. Chen is a senior member of the Institute of Electrical and Electronic Engineers (IEEE), a member of the American Statistical Association, and Pattern Recognition Society.

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Chapter 1

INTRODUCTION

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I. DESCRIPTION OF DIGITAL WAVEFORMS

The measured data in many applications are a set of waveforms (or a waveform) from which desired information must be extracted. In the biomedical area, for example, the electrocardiogram (EKG) is taken from a patient and interpreted by a physician to determine whether the patient's heart is normal or abnormal. In the case of abnormality, further information may be obtained from the EKG. In practice, waveforms measured in many applications contain far more information than can be fully extracted by human users. Also, the large volume of data will make it very difficult if not impossible for human users to obtain the desired information in a reasonable amount of time. Computer-aided, interactive, and fully automatic techniques have been developed for processing and recognition. It is often necessary to digitize the waveforms before any processing and recognition can be performed by digital computers. The result of processing and recognition will be the output of the computer. For the EKG example, the processed result may be the spectral display, while the recognition result may be an interpretation of the EKG.

In this book the waveforms considered are in digital form. For processing, the input is thus a set of digital sequence or time-series data. For recognition, the input is usually some processed data such as the vector sample consisting of several feature measurements. Both digital systems for processing and recognition are also considered in this book.

II. PROCESSING OF DIGITAL WAVEFORMS

Processing includes practically all operations of the digital waveform by digital computers other than decision making and interpretations. Processing is closely related to recognition because a preliminary processing of waveforms is almost always needed for recognition. An important class of processing operations is the filtering of the data to minimize the effect of instrumentation noise or to constrain the data to certain frequency range. This can be done in both analog and digital form. Digital filtering has gained much acceptance because of its flexibility and high performance. Another important class of processing operations is the spectral analysis of waveforms. In addition to time-domain analysis of the data, frequency-domain analysis is essential in waveform study. Quite often both time-domain and frequency-domain processing are needed to derive certain desired information.

III. RECOGNITION OF DIGITAL WAVEFORMS

The main recognition objective is to classify a pattern or, in this case, the digital waveform into one of several possible categories (classes). Another recognition objective may be to obtain certain descriptions or structures of the data. However, a key problem in recognition is the feature extraction, i.e., the determination of characteristic features that can discriminate data from different classes and correctly identify samples from the same classes. By considering the patterns as statistical in nature, the statistical pattern recognition deals with the statistical description of digital signals, the extraction of mathematical features, decision rules, clustering, and the estimation of parameters and densities. Syntactic pattern recognition, on the other hand, deals with primitive selection and pattern grammar, syntactic classification and error correcting, and parsing and syntactic clustering.

IV. OVERVIEW OF THE BOOK

There is now very strong interest and numerous developments in both the digital signal processing and digital pattern recognition fields and their applications. By considering both fields in the same book, the common, interrelated methodologies in both processing and recognition can be developed and applied to various applications.

The fundamentals of digital signal processing considered in Chapter 2 deal with the specific topics of nonrecursive digital filters and discrete estimation and detection. The latter, which is not discussed in digital signal processing texts, is of fundamental importance to the processing and classification of digital waveforms. Chapter 3 is concerned primarily with the spectral analysis, with illustrative examples on speech and biomedical waveforms. Chapter 4 provides a comprehensive treatment of the statistical theory in pattern recognition. An excellent presentation of the syntactic approach to pattern recognition with application to signal processing is provided in Chapter 5. Chapter 6 examines several important techniques in speech processing. A very detailed presentation on geophysical data in both processing and recognition is given in Chapters 7 and 8. Both teleseismic and intrusion-detection seismic data are examined in detail as illustrative examples. Listings of several important computer programs are also provided. Key aspects of sonar and radar signal processing are examined in Chapter 9 and 10, respectively. Finally in Chapter 11, we consider the digital system implementation problems for both processing and recognition.

BIBLIOGRAPHIC NOTE

In each of the two fields, digital signal processing and pattern recognition, there are now well over 20 books published in the last 15 years. The number of articles published far exceeds 1000 in each field. The following short bibliography is representative of the enormous amount of published literature in both fields.

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Chapter 2

FUNDAMENTALS OF DIGITAL SIGNAL PROCESSING

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I. INTRODUCTION

The fundamentals of digital signal processing encompass a broad spectrum of topics: sampling and reconstruction, transform methods, digital filtering, spectral analysis, adaptive signal processing, discrete estimation and detection, and signal processing hardware and software, etc. These topics are examined on a selective basis in this chapter and in Chapters 3, 7, 8, and 11. The basic principles of digital signal processing are well described in introductory level books[1,2] and more advanced level books.[3,4] These books are recommended to the reader for a complete introduction to the subject. Following an overview of the field of digital signal processing, this chapter will deal with several aspects of the field.

A typical digital signal processor starts with the digitization (i.e., sampling) of a continuous waveform at a sampling rate that is at least twice the highest significant frequency of the waveform. The remaining part of this analog-to-digital conversion is quantization of the discrete data, which, unlike sampling, is an irreversible process. The resulting data are processed digitally by filtering, transforming, and the combination of various techniques so that the desired information can be extracted or enhanced. The output of the digital processor can be in digital form or, in many applications, converted to analog form. Both the processed digital and analog data can be displayed as needed.

II. OVERVIEW OF DIGITAL SIGNAL PROCESSING

The field of digital signal processing has grown enormously in the past 15 years to provide firm theoretical background for a number of topics, as mentioned in the previous section. The major subdivisions of the field are digital filtering and spectrum analysis. The field of digital filtering is further divided into nonrecursive or finite impulse response (FIR) filters and recursive or infinite impulse response (IIR) filters. The latter may be considered as discrete counterparts of the continuous linear time invariant system. The field of spectrum analysis is broken into calculation of spectra via the discrete Fourier transform (DFT) and via statistical techniques as in the case of random signals, e.g., quantization noise in a digital system. The fast Fourier transform (FFT), which is a computationally efficient procedure to calculate DFT, and the related area of fast convolution are almost exclusively used in practical spectrum analysis techniques. The remaining aspects of digital signal processing are the important topics of implementation of digital systems and application areas. The applications are treated in detail in this book and the book by Oppenheim.[5] A good understanding of the issues involved in practical implementation of digital systems for signal processing in finite precision software or hardware, is essential to make good use of the theoretical study. The flexibility and greater accuracy offered by digital processing will help motivate the development of new digital components. Eventually, digital signal processing will replace analog processing in most applications, as the digital computer replaces the analog computer.

III. DESIGN OF NONRECURSIVE DIFFERENTIATOR AND HILBERT TRANSFORMER

In this section, we consider a simple technique of designing nonrecursive digital filters, as proposed by Gold and Radar.[6] The technique is flexible enough so that only a minor change in computer program statements is required to change from the differentiator to the Hilbert transformer or vice versa.[7] The effects of various window functions and the size of the window on the frequency response of the resulting filter are also examined.

The transfer functions of an ideal differentiator and an ideal Hilbert transformer are given by

$$F_D\,(j\omega) = j\omega, \qquad \pi<\omega T<\pi \tag{1}$$

and

$$F_{H\,(T)} = \begin{cases} +j & 0<\omega T<\pi \\ -j & -\pi<\omega T<0 \end{cases} \tag{2}$$

respectively. Here, T is the sampling period. The real part of frequency response is zero for each filter. The subscripts D and H refer to the differentiator and the Hilbert transformer, respectively. The impulse response β_n is the Fourier coefficient defined by[6]

$$\beta_n = \frac{1}{2\pi}\int_{\pi}^{\pi} F(\omega T)e^{+j\omega Tn}d(\omega T) \tag{3}$$

For the differentiator

$$\beta_{nD} = \begin{cases} \dfrac{(-1)^n}{n} & , \ n \neq 0 \\ 0 & , \ n = 0 \end{cases} \tag{4}$$

For the Hilbert transformer

$$\beta_{nH} = \begin{cases} -\dfrac{2}{n\pi} & , \ n \neq 0 \text{ and n odd integer} \\ 0 & , \ n = 0 \text{ or n even integer} \end{cases} \tag{5}$$

As is well known, weighting functions (also called window functions) are used to modify and truncate a Fourier series that represents a periodic function. The resulting Fourier representation, which exhibits Gibb's phenomenon, is an approximation to the original function.

Let $\omega(n)$ be the window function and N, a power of 2, be the size of the window. The window functions typically employed are

(1) Rectangular window

$$\omega\,(n) = \begin{cases} 1, & |n|<N \\ 0, & |n|\geqslant N \end{cases} \tag{6}$$

(2) Triangular window

$$\omega\,(n) = \begin{cases} 1 - \left|\dfrac{n}{N}\right| & , \ |n|<N \\ 0 & , \ |n|\geqslant N \end{cases} \tag{7}$$

(3) Parabolic window

$$\omega(n) = \begin{cases} 1 - \left|\frac{n}{N}\right|^2, & |n| < N \\ 0, & |n| \geq N \end{cases} \tag{8}$$

(4) Unraised half-cosine window

$$\omega(n) = \begin{cases} \cos\frac{n\pi}{2N}, & |n| < N \\ 0, & |n| \geq N \end{cases} \tag{9}$$

(5) Hanning window

$$\omega(n) = \begin{cases} 0.5 + 0.5\cos\frac{n\pi}{N}, & |n| < N \\ 0, & |n| \geq N \end{cases} \tag{10}$$

(6) Blackman's window

$$\omega(n) = \begin{cases} 0.42 + 0.5\cos\frac{n\pi}{N} + 0.08\frac{2n\pi}{N} & |n| < N \\ 0, & |n| \geq N \end{cases} \tag{11}$$

Computationally, β_n is calculated from Equation 3 by using FFT and then multiplied by the window function. Then, we take the FFT of $\beta_n\omega_n$ to obtain the desired frequency response of the nonrecursive digital filter. The imaginary parts, $\mathrm{Im}[F(j\omega)]$, of the frequency response of the differentiator and the Hilbert transformer are shown in Figures 1 and 2, respectively. $\mathrm{Im}[F_D(j\omega)]$ is continuous at $\omega T = 0$ but possesses a jump of $2\pi/T$ at $\omega T = \pi$. $\mathrm{Im}[F_H(j\omega)]$ exhibits a discontinuity at $\omega T = 0$ as well as $\omega T = \pi$. The positive n parts of the impulse response β_n of the filters are shown in Figures 3 and 4 with a total number of points $N_t = 256$. For the rectangular window and N = 16, the frequency responses of the truncated impulse response are shown in Figures 5 and 6, respectively, for the differentiator and the Hilbert transformer. For N = 8, the corresponding results are shown in Figures 7 and 8. The number of ripples is doubled from N = 8 to N = 16 but N = 16 provides a much better approximation to the ideal frequency response. The frequency responses corresponding to the triangular window are shown in Figures 9 and 10 for N = 6 and in Figures 11 and 12 for N = 8. It is noted that the ripples are considerably reduced in going from rectangular to triangular windows. Again, N = 16 provides much better frequency responses compared to N = 8. By changing the triangular to other windows, only slight improvement in frequency response is obtained (see Figures 13 to 16 and 17 to 20. The Blackman's window provides the most smooth frequency response in comparison with the parabolic, half-cosine, and Hanning windows, but it has the largest transition bandwidth. It may be concluded that each window function has its own advantages and disadvantages. The choice of a particular window depends on the requirements of the filter used.

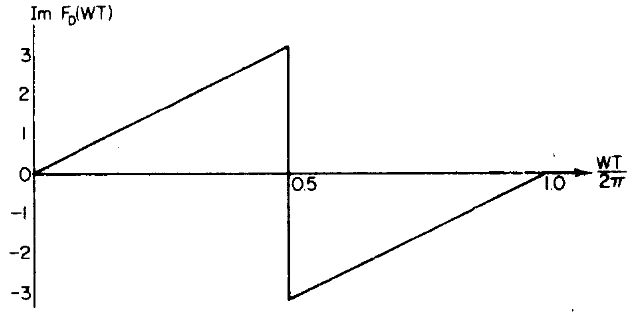

FIGURE 1. Ideal frequency response $F_D(WT)$, imaginary part.

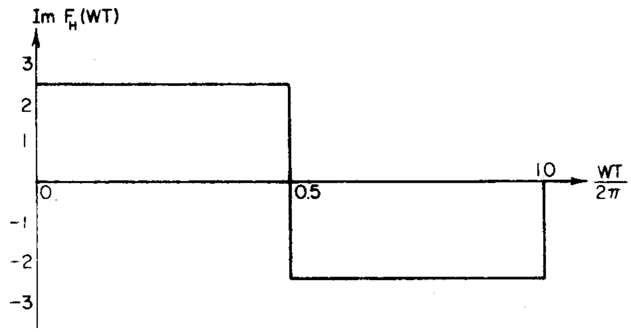

FIGURE 2. Ideal frequency response, $F_H(WT)$ for Hilbert transformer, imaginary part.

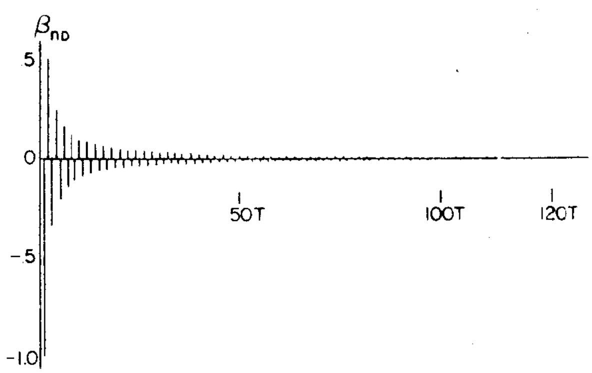

FIGURE 3. Ideal β_{nd}, positive n part. N. = 256

Although only the differentiator and the Hilbert transformer are discussed in this section, the method of approach can be used just as easily for other nonrecursive filter designs.

IV. DIGITAL CROSS-CORRELATOR

Digital cross-correlation and matched filtering operations are frequently used in digital communications, sonar and radar signal processing, etc. Effectiveness of these

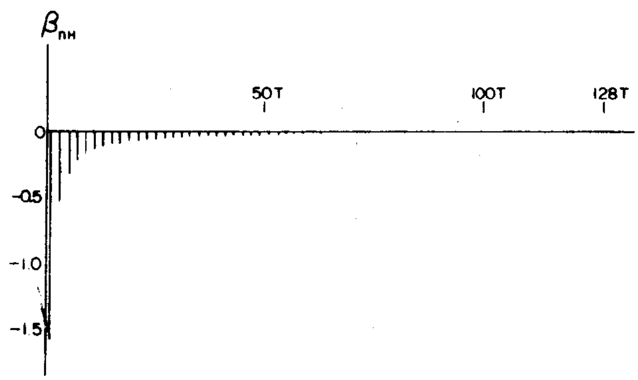

FIGURE 4. Ideal β_{nH}, positive n part, N. = 256.

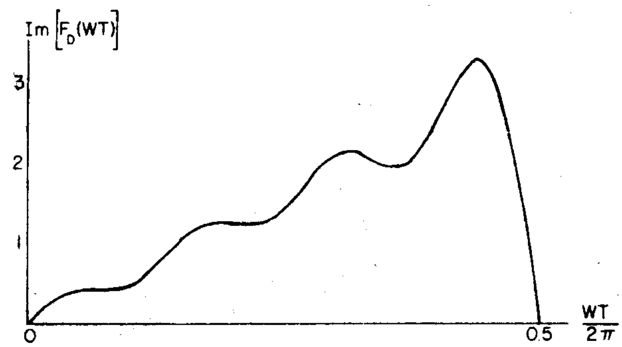

FIGURE 5. F_D(WT), rectangular window, N = 16.

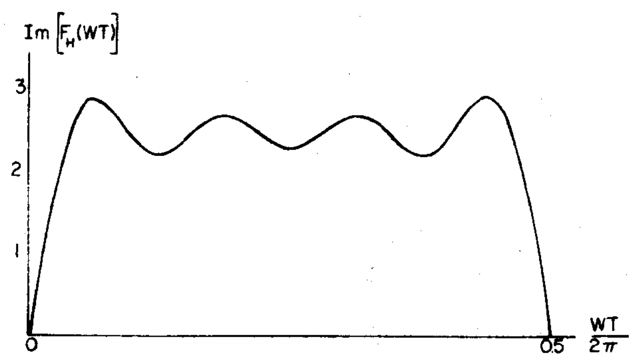

FIGURE 6. F_H(WT), rectangular window, N = 16.

operations depends highly on the noise and quantization. Figure 21 is a block diagram of a digital correlator. The input waveform x(t) = s(t) + n(t) + i(t) consists of a signal s(t), random noise n(t), and interference i(t). The reference waveform r(t) can be deterministic. These waveforms are sampled once every T seconds and then quantized, usually with a symmetric quantizer. The output variables a_i, which is random,