DIGITAL SIGNAL PROCESSING:

A System Design Approach

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Preface

Digital signal processing started to gain popularity in the mid- to late 1960s with the development of integrated circuits (ICs). Since this period, the technology has evolved from medium-scale integrated (MSI) circuits to very large-scale integrated (VLSI) circuits and very high speed integrated circuits (VHSIC). This advancement in digital technology has made the implementation of sophisticated algorithms (e.g., the fast Fourier transform) oriented toward performing real-time digital signal processing tasks feasible.

The development of several more efficient algorithms including multirate processing techniques and fast algorithms for filtering coupled with the latest technologies has made complex real-time digital signal processing systems possible. Programmable signal processors capable of performing the complex tasks for high data-rate applications within limited space and stressing environments can be achieved. Examples of applications that can be implemented efficiently using digital signal processing cover a broad spectrum including sonar, radar, seismic, communications, navigation, telephony, speech, image, and audio processing.

The material in this text evolved over several years from working in the field, from teaching digital signal processing courses to practicing electrical and computer science engineers at IBM, and from teaching senior undergraduate and graduate university students at the University of Virginia. Based on this experience, it was evident that a need existed for a text that goes beyond the presentation of the individual signal processing concepts and develops a systems approach showing the interrelationships between the individual processing elements in solving application problems.

The material in this text has been designed for a broad spectrum of technical users including advanced undergraduate and graduate students in engineering and computer science curriculums and practicing engineers and scientists. The text introduces the digital signal processing concepts, develops design and analysis expressions that can be easily implemented on a personal computer, and illustrates the use of the concepts in designing systems. The material is presented from a system engineering perspective. A major feature of the text is the development of a signal processing system design methodology as a systematic approach to solving complex application design problems.

Chapters 1 through 6 present the basic concepts of linear systems, digital filters, *Z*-transform analysis, discrete Fourier transforms (DFT), and fast Fourier transform (FFT) algorithms. The advanced concepts of multirate processing, power spectrum estimation, finite arithmetic analysis, system design, and adaptive processing are presented in Chapters 7 through 11. A summary of each of the chapters is provided in the following paragraphs.

Chapter 1 presents an overview of digital signal processing and discusses key concepts and approaches to performing the overall signal processing system design and implementation. Discrete linear systems are defined in Chapter 2—along with the associated theory of convolution, frequency response, and sampling—which are key to the remaining material presented in the text. The general difference equation for a digital filter is defined, and the relationship of two filter types: infinite impulse response (IIR) and finite impulse response (FIR) are developed.

Chapter 3 defines the *Z*-transform and inverse *Z*-transform pair and the application of the transform pair to the analysis of discrete linear systems. The definition of a discrete linear system transfer function is developed. Basic digital network concepts and filter realizations are presented. Important *Z*-transform properties are defined and the relationship between the *Z*-transform and frequency response is described.

IIR and FIR design techniques are presented in Chapters 4 and 5, respectively. A systematic approach is developed for the design of IIR digital filters using classic analog design functions. This approach is demonstrated by using flow diagrams to determine the poles, zeros, and coefficients of elliptic filters. Examples of lowpass, highpass, bandpass, and bandstop designs are presented. The method has been used to implement the filter designs on a personal computer. IIR design techniques are continuing to evolve. Approaches to direct design of the IIR digital filter are discussed.

FIR filter properties are presented. A systematic approach to designing FIR filters using the Fourier series method is presented with emphasis on the Kaiser window design. The most widely used MINMAX FIR design optimization approach is described. That approach uses the Remez exchange algorithm and was developed by Parks and McClellan. Potential pitfalls in the design are discussed. Again, detailed examples together with computer results illustrate the merit of the systematic approach to the design.

The DFT and the FFT are presented in Chapter 6. The properties of the DFT are presented with emphasis on the spectral leakage and circular shift resulting from the finite length transform. Key characteristics relating to the application of the DFT to spectrum analysis are presented based on the frequency response. The use of weighting functions is described. The need for redundancy processing when averaging successive DFT outputs is defined.

The FFT decimation-in-time and decimation-in-frequency radix-2 algorithms are developed showing the efficiency gained in computational requirements over the DFT. The FFT development stresses the definition of a basic computational unit that is repeatedly executed to perform the FFT. An efficient algorithm for the implementation of an N-point FFT using a radix-4 computational unit is derived and illustrated. BASIC subroutines for the computational unit are presented.

Multirate digital signal processing techniques are presented in Chapter 7. The key concepts of interpolation and decimation multirate processes are defined. Multirate techniques offer significant computational savings in the design of systems; desired bandwidth is small compared to the input bandwidth. Methods of optimizing the design for minimum computational requirements are presented. Filter design approaches for multirate systems are described.

Chapter 8 presents the basic concepts of discrete linear system responses to random signal inputs and power spectrum estimation of the random processes. The autocorrelation and cross-correlation functions are presented and their relationship to the power spectral density and cross-power spectral density of the random process are defined. These functions provide the basis for interpreting linear system performance to random processes. Estimation of the power spectral density by averaging modified periodograms are described. Finally, a discussion of the basic concepts of detection theory is presented with application to a narrowband detection system.

Chapter 9 covers the effects of finite wordlength implementations on the IIR, FIR, and FFT processing algorithms previously presented. The effects of the finite length filter and FFT coefficients are presented. Finite length arithmetic effects for implementation of recursive structures in cascade are presented in detail with respect to system implementation and performance. Fixed-point implementations are emphasized. Floating-point implementations are discussed. The approach presented is easily programmed on a personal computer.

Chapter 10 develops a methodology for performing a signal processing system design. The methodology is used to solve a spectral analysis application. Key signal processor architectural factors are addressed. A thorough analysis of the processor resource requirements is discussed. Results are presented in personal computer spreadsheet form to illustrate the resource analysis process. An acceptable design is achieved, and its performance is estimated. The performance calculations can be programmed and used to vary parameters such as probability of detection, probability of false alarm, bandwidth, and weighting functions to parametrically determine the performance versus system design choices.

In Chapter 11, digital filters that adapt to a changing environment are discussed. The stochastic Wiener filtering and deterministic least-squares problems are set up, and the similarity between them is pointed out. First, the block-processing approach is taken in the solution of these two problems. Second, a fading memory (recursive) approach is taken. Finally, an application of these algorithms to the adaptive beamforming (ABF) problem is considered.

The goal in writing this textbook was to provide a clear description of the application of digital signal processing concepts in solving system engineering signal processing application problems. A methodology was provided as an aid in performing the complex design and analysis processes required to determine an efficient implementation. We hope that we have achieved our goal and that you will find the material useful in your signal processing related work.

For Instructors

The material provides the basis for a two-semester course. Chapters 1 through 6 make up an undergraduate senior level or first-year graduate level digital signal processing course. They may be presented in order or the instructor may prefer to present the DFT portion of Chapter 6 prior to Chapters 4 and 5. Chapters 7 through 11 offer a variety of options for an advanced course and for defining independent course study topics.

A substantial set of problems are provided at the end of each chapter. Some of these involve derivations and exercises to the text material; while others are concerned with computer solution of problems encountered in real applications. In many cases the problems extend the concepts already developed in the text. These problems provide the instructor with an extra degree of freedom to use different approaches in presenting the material.

A comprehensive instructors manual is available that includes solutions to the problems, and suggested additional exercises and projects which enhance the concepts developed in the text. We feel that the usefulness of the book is directly proportional to a problem set which motivates the reader to further investigate the technical concepts. As a result the instructor is provided with a comprehensive treatment of solutions to all problems, where in many cases computer programs or computer generated spreadsheets are used to describe the approach. Finally, subject matter is included in the instructors manual as a guide in presenting the material.

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