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Paulo S. R. Diniz, Eduardo A. B. da Silva and Sergio L. Netto
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Digital Signal Processing

This new, fully revised edition covers all the major topics of digital signal processing (DSP) design and analysis in a single, all-inclusive volume, interweaving theory with real-world examples and design trade-offs.

Building on the success of the original, this edition includes new material on random signal processing, a new chapter on spectral estimation, greatly expanded coverage of filter banks and wavelets, and new material on the solution of difference equations. Additional steps in mathematical derivations make them easier to follow, and an important new feature is the Do-it-Yourself section at the end of each chapter, where readers get hands-on experience of solving practical signal processing problems in a range of MATLAB® experiments.

With 120 worked examples, 20 case studies, and almost 400 homework exercises, the book is essential reading for anyone taking digital signal processing courses. Its unique blend of theory and real-world practical examples also makes it an ideal reference for practitioners.

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System Analysis and Design

Second Edition

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To our families,
our parents,
and our students.

Contents

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Preface

This book originated from a training course for engineers at the research and development center of TELEBRAS, the former Brazilian telecommunications holding. That course was taught by the first author back in 1987, and its main goal was to present efficient digital filter design methods suitable for solving some of their engineering problems. Later on, this original text was used by the first author as the basic reference for the digital filters and digital signal processing courses of the Electrical Engineering Program at COPPE/Federal University of Rio de Janeiro.

For many years, former students asked why the original text was not transformed into a book, as it presented a very distinct view that they considered worth publishing. Among the numerous reasons not to attempt such task, we could mention that there were already a good number of well-written texts on the subject; also, after many years of teaching and researching on this topic, it seemed more interesting to follow other paths than the painful one of writing a book; finally, the original text was written in Portuguese and a mere translation of it into English would be a very tedious task.

In later years, the second and third authors, who had attended the signal processing courses using the original material, were continuously giving new ideas on how to proceed. That was when we decided to go through the task of completing and updating the original text, turning it into a modern textbook. The book then took on its first-edition form, updating the original text, and including a large amount of new material written for other courses taught by the three authors up to 2002.

This second edition barely resembles the original lecture notes for several reasons. The original material was heavily concentrated on filter design and realization, whereas the present version includes a large amount of material on discrete-time systems, discrete transforms, spectral estimation, multirate systems, filter banks, and wavelets.

This book is mainly written for use as a textbook on a digital signal processing course for undergraduate students who have had previous exposure to basic linear systems, or to serve as a textbook on a graduate-level course where the most advanced topics of some chapters are covered. This reflects the structure we have at the Federal University of Rio de Janeiro, as well as at a number of other universities we have contact with. The second edition has a special feature designed for readers to test their learning by hands-on experience through so-called Do-it-yourself sections, with the aid of MATLAB®. A Do-it-yourself section is included in all chapters of the book. The book also includes, at the end of most chapters, a brief section aimed at giving a start to the reader on how to use MATLAB as a tool for the analysis and design of digital signal processing systems. As in the first edition, we decided that having explanations about MATLAB inserted in the main text would in some cases distract the readers, making them lose focus on the subject.

A distinctive feature of this book is to present a wide range of topics in digital signal processing design and analysis in a concise and complete form, while allowing the reader to fully develop practical systems. Although this book is primarily intended as an undergraduate and graduate textbook, its origins on training courses for industry warrant its potential usefulness to engineers working in the development of signal processing systems. In fact, our objective is to equip the readers with the tools that enable them to understand why and how to use digital signal processing systems; to show them how to approximate a desired transfer function characteristic using polynomials and ratios of polynomials; to teach them why an appropriate mapping of a transfer function into a suitable structure is important for practical applications; and to show how to analyze, represent, and explore the trade-off between the time and frequency representations of deterministic and stochastic signals. For all that, each chapter includes a number of examples and end-of-chapter problems to be solved. These are aimed at assimilating the concepts, as well as complementing the text. In particular, the second edition includes many new examples and exercises to be solved.

Chapters 1 and 2 review the basic concepts of discrete-time signal processing and z transforms. Although many readers may be familiar with these subjects, they could benefit from reading these chapters, getting used to the notation and the authors' way of presenting the subject. In Chapter 1 we review the concepts of discrete-time systems, including the representation of discrete-time signals and systems, as well as their time-domain responses. Most important, we present the sampling theorem, which sets the conditions for the discrete-time systems to solve practical problems related to our real continuous-time world. The basic concepts of random signals are also introduced in this chapter, followed by the Do-it-yourself section aiding the reader to test their progress in discrete-time signals and systems. Chapter 2 is concerned with the z and Fourier transforms, which are useful mathematical tools for representation of discrete-time signals and systems. The basic properties of the z and Fourier transforms are discussed, including a stability test in the z transform domain. The chapter also shows how the analysis of random signals can benefit from the z-domain formulation.

Chapter 3 discusses discrete transforms, with special emphasis given to the discrete Fourier transform (DFT), which is an invaluable tool in the frequency analysis of discrete-time signals. The DFT allows a discrete representation of discrete-time signals in the frequency domain. Since the sequence representation is natural for digital computers, the DFT is a very powerful tool, because it enables us to manipulate frequency-domain information in the same way as we can manipulate the original sequences. The importance of the DFT is further increased by the fact that computationally efficient algorithms, the so-called fast Fourier transforms (FFTs), are available to compute the DFT. This chapter also presents real coefficient transforms, such as cosine and sine transforms, which are widely used in modern audio and video coding, as well as in a number of other applications. A discussion about orthogonality in transforms is also included. This section also includes a discussion on the several forms of representing the signals, in order to aid the reader with the available choices.

Chapter 4 addresses the basic structures for mapping a transfer function into a digital filter. It is also devoted to some basic analysis methods and properties of digital filter structures.

The chapter also introduces some simple and useful building blocks widely utilized in some designs and applications.

Chapter 5 introduces several approximation methods for filters with finite-duration impulse response (FIR), starting with the simpler frequency sampling method and the widely used windows method. This method also provides insight to the windowing strategy used in several signal processing applications. Other approximation methods included are the maximally flat filters and those based on the weighted least-squares (WLS) method. This chapter also presents the Chebyshev approximation based on a multivariable optimization algorithm called the Remez exchange method. This approach leads to linear-phase transfer functions with minimum order given a prescribed set of frequency response specifications. This chapter also discusses the WLS–Chebyshev method which leads to transfer functions where the maximum and the total energy of the approximation error are prescribed. This approximation method is not widely discussed in the open literature but appears to be very useful for a number of applications.

Chapter 6 discusses the approximation procedures for filters with infinite-duration impulse response (IIR). We start with the classical continuous-time transfer-function approximations, namely the Butterworth, Chebyshev, and elliptic approximations, that can generate discrete-time transfer functions by using appropriate transformations. Two transformation methods are then presented: the impulse-invariance and the bilinear transformation methods. The chapter also includes a section on frequency transformations in the discrete-time domain. The simultaneous magnitude and phase approximation of IIR digital filters using optimization techniques is also included, providing a tool to design transfer functions satisfying more general specifications. The chapter closes by addressing the issue of time-domain approximations.

Chapter 7 introduces the basic concepts of classical estimation theory. It starts by describing the nonparametric spectral estimation methods based on a periodogram, followed by the minimum-variance spectral estimator. The chapter continues with a discussion on modeling theory, addressing the rational transfer function models and presenting the Yule–Walker equations. Several parametric spectral estimation methods are also presented, namely: the linear prediction method; the covariance method; the autocorrelation method; the Levinson–Durbin algorithm; and Burg's method. The chapter also discusses the Wiener filter as an extension of the linear prediction method.

Chapter 8 deals with basic principles of discrete-time systems with multiple sampling rates. In this chapter we emphasize the basic properties of multirate systems, thoroughly addressing the decimation and interpolation operations, giving examples of their use for efficient digital filter design. The chapter discusses many key properties of multirate systems, such as inverse operations and noble identities, and introduces some analytical tools, such as polyphase decomposition and the commutator models. In addition, we discuss the concepts of overlapped block filtering, which can be very useful in some fast implementations of digital signal processing building blocks. The chapter also includes some discussion on how decimators and interpolators affect the properties of random signals.

Chapter 9 discusses some properties pertaining to the internal structure of filter banks, followed by the concept and construction of perfect reconstruction filter banks. The chapter also includes some analysis tools and classifications for the filter banks and transmultiplexers. This chapter presents several design techniques for multirate filter banks, including several forms of two-band filter banks, cosine-modulated filter banks, and lapped transforms.

Chapter 10 introduces the concepts of time–frequency analysis and the discrete wavelet transform. It also presents the multiresolution representation of signals through wavelet transforms and discusses the design of wavelet transforms using filter banks. In addition, some design techniques to generate orthogonal (as well as biorthogonal) bases for signal representation are presented. Several properties of wavelets required for their classification, design, and implementation are discussed in this chapter.

Chapter 11 provides a brief introduction to the binary number representations most widely used in the implementation of digital signal processing systems. The chapter also explains how the basic elements utilized in these systems work and discusses a particular, and yet instructive, type of implementation based on distributed arithmetic. Chapter 11 also includes the models that account for quantization effects in digital filters. We discuss several approaches to analyze and deal with the effects of representing signals and filter coefficients with finite wordlength. In particular, we study the effects of quantization noise in products, signal scaling that limits the internal signal dynamic range, coefficient quantization in the designed transfer function, and the nonlinear oscillations which may occur in recursive realizations. These analyses are used to indicate the filter realizations that lead to practical finite-precision implementations of digital filters.

In Chapter 12 we present some techniques to reduce the computational complexity of FIR filters with demanding specifications or specialized requirements. The first structure discussed is the lattice form, which finds application in a number of areas, including the design of filter banks. Several useful implementation forms of FIR filters, such as polyphase, frequency-domain, recursive running sum, and modified-sinc forms, are presented to be employed as building blocks in several design methods. In particular, we introduce the prefilter and interpolation methods which are mainly useful in designing narrowband lowpass and highpass filters. In addition, we present the frequency-response masking approach, for designing filters with narrow transition bands satisfying more general specifications, and the quadrature method, for narrow bandpass and bandstop filters.

Chapter 13 presents a number of efficient realizations for IIR filters. For these filters, a number of realizations considered efficient from the finite-precision effects point of view are presented and their salient features are discussed in detail. These realizations will equip the reader with a number of choices for the design of good IIR filters. Several families of structures are considered in this chapter, namely: parallel and cascade designs using direct-form second-order sections; parallel and cascade designs using section-optimal and limit-cycle-free state-space sections; lattice filters; and several forms of wave digital filters. In addition, this chapter includes a discussion on doubly complementary filters and their use in the implementation of quadrature mirror filter banks.

This book contains enough material for an undergraduate course on digital signal processing and a first-year graduate course. There are many alternative ways to compose these courses; in the following we describe some recommendations that have been employed successfully in signal processing courses.

- An undergraduate course in discrete-time systems or digital signal processing at junior level. This should include most parts of Chapters 1, 2, 3, 4, and the nonparametric methods of Chapter 7. It could also include the noniterative approximation methods of Chapters 5 and 6, namely the frequency sampling and window methods described in Chapter 5, the analog-based approximation methods, and also the continuous-time to discrete-time transformation methods for IIR filtering of Chapter 6.
- An undergraduate course in digital signal processing at senior level. This should briefly review parts of Chapters 1 and 2 and cover Chapters 3, 4, and 7. It could also include the noniterative approximation methods of Chapters 5 and 6, namely the frequency sampling and window methods described in Chapter 5, the analog-based approximation methods, and also the continuous-time to discrete-time transformation methods for IIR filtering of Chapter 6. Chapters 8 and 11 could complement the course.
- An undergraduate course in digital filtering at senior level or first-year graduate. This should cover Chapter 4 and the iterative approximation methods of Chapters 5 and 6. The course could also cover selected topics from Chapters 11, 12, and 13. At the instructor's discretion, the course could also include selected parts of Chapter 8.
- As a graduate course textbook on multirate systems, filter banks and wavelets. The course could cover Chapters 8, 9, and 10, as well as the lattice form in Chapter 12 and doubly complementary filters in Chapter 13.

Obviously, there are several other choices for courses based on the material of this book which will depend on the course length and the judicious choice of the instructor.

This book would never be written if people with a wide vision of how an academic environment should be were not around. In fact, we were fortunate to have Professors L. P. Calôba and E. H. Watanabe as colleagues and advisors. The staff of COPPE, in particular Ms Michelle A. Nogueira and Ms F. J. Ribeiro, supported us in all possible ways to make this book a reality. Also, the first author's early students J. C. Cabezas, R. G. Lins, and J. A. B. Pereira (in memoriam) wrote, with him, a computer package that generated several of the examples of the first edition of this book. The engineers of CPqD helped us to correct the early version of this text. In particular, we would like to thank the engineer J. Sampaio for his complete trust in this work. We benefited from working in an environment with a large signal-processing group where our colleagues always helped us in various ways. Among them, we should mention Professors L. W. P. Biscainho, M. L. R. de Campos, G. V. Mendonça, A. C. M. de Queiroz, F. G. V. de Resende Jr, J. M. de Seixas, and the entire staff of the Signal Processing Lab (`www.lps.ufrj.br`). Professor Biscainho superbly translated the first edition of this book to our mother tongue; he is indeed our inspirational fourth author. We would like to thank our colleagues at the Federal University of Rio de Janeiro, in particular at the Department of Electronics and Computer Engineering of the Polytechnic School of Engineering, the undergraduate studies department, and at the Electrical Engineering Program of COPPE, the graduate studies department, for their constant support during the preparation of this book.

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