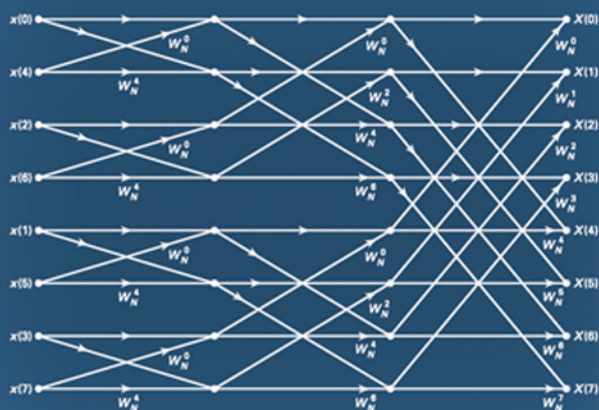


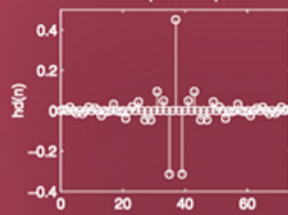
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DIGITAL **SIGNAL** PROCESSING

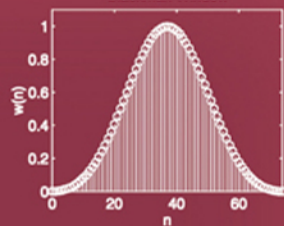
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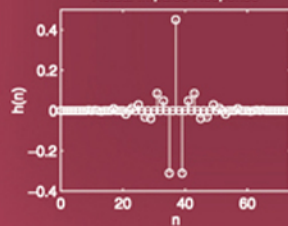
Ideal Impulse Response



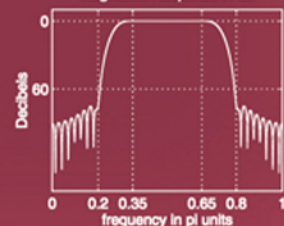
Blackman Window



Actual Impulse Response

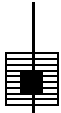


Magnitude Response in dB

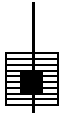


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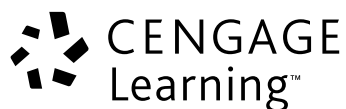


Digital Signal Processing Using MATLAB[®] Third Edition



Digital Signal Processing Using MATLAB[®] Third Edition

Vinay K. Ingle
John G. Proakis
Northeastern University



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Preface

From the beginning of the 1980s we have witnessed a revolution in computer technology and an explosion in user-friendly applications. This revolution is still continuing today with low-cost personal computer systems that rival the performance of expensive workstations. This technological prowess should be brought to bear on the educational process and, in particular, on effective teaching that can result in enhanced learning. This companion book on digital signal processing (DSP) makes a small contribution toward reaching that goal.

The teaching methods in signal processing have changed over the years from the simple “lecture-only” format to a more integrated “lecture-laboratory” environment in which practical hands-on issues are taught using DSP hardware. However, for effective teaching of DSP the lecture component must also make extensive use of computer-based explanations, examples, and exercises. For the past several years, the MATLAB software developed by *The MathWorks, Inc.* has established itself as the de facto standard for numerical computation in the signal-processing community and as a platform of choice for algorithm development. There are several reasons for this development, but the most important reason is that MATLAB is available on practically all-computing platforms. In this book we have made an attempt at integrating MATLAB with traditional topics in DSP so that it can be used to explore difficult topics and solve problems to gain insight. Many problems or design algorithms in DSP require considerable computation. It is for these that MATLAB provides a convenient tool so that many scenarios can be tried with ease. Such an approach can enhance the learning process.

SCOPE OF THE BOOK

This book is primarily intended for use as a supplement in junior- or senior-level undergraduate courses on DSP. Although we assume that the student (or user) is familiar with the fundamentals of MATLAB, we have provided a brief introduction to MATLAB in Chapter 1. Also, this book is not written as a textbook in DSP because of the availability of excellent textbooks. What we have tried to do is to provide enough depth to the material augmented by MATLAB functions and examples so that the presentation is consistent, logical, and enjoyable. Therefore, this book can also be used as a self-study guide by anyone interested in DSP.

ORGANIZATION OF THE BOOK

The first ten chapters of this book discuss traditional material typically covered in an introductory course on DSP. The final two chapters are presented as applications in DSP with emphasis on MATLAB-based projects. The following is a list of chapters and a brief description of their contents.

Chapter 1, Introduction: This chapter introduces readers to the discipline of signal processing and presents several applications of digital signal processing, including musical sound processing, echo generation, echo removal, and digital reverberation. A brief introduction to MATLAB is also provided.

Chapter 2, Discrete-time Signals and Systems: This chapter provides a brief review of discrete-time signals and systems in the time domain. Appropriate use of MATLAB functions is demonstrated.

Chapter 3, The Discrete-time Fourier Analysis: This chapter discusses discrete-time signal and system representation in the frequency domain. Sampling and reconstruction of analog signals are also presented.

Chapter 4, The z -Transform: This chapter provides signal and system description in the complex frequency domain. MATLAB techniques are introduced to analyze z -transforms and to compute inverse z -transforms. Solutions of difference equations using the z -transform and MATLAB are provided.

Chapter 5, The Discrete Fourier Transform: This chapter is devoted to the computation of the Fourier transform and to its efficient

implementation. The discrete Fourier series is used to introduce the discrete Fourier transform, and several of its properties are demonstrated using MATLAB. Topics such as fast convolution and fast Fourier transform are thoroughly discussed.

Chapter 6, Implementation of Discrete-Time Filters: This chapter discusses several structures for the implementation of digital filters. Several useful MATLAB functions are developed for the determination and implementation of these structures. Lattice and ladder filters are also introduced and discussed. In addition to considering various filter structures, we also treat quantization effects when finite-precision arithmetic is used in the implementation of IIR and FIR filters.

Chapter 7, FIR Filter Design: This chapter and the next introduce the important topic of digital filter design. Three important design techniques for FIR filters—namely, window design, frequency sampling design, and the equiripple filter design—are discussed. Several design examples are provided using MATLAB.

Chapter 8, IIR Filter Design: Included in this chapter are techniques used in IIR filter design. The chapter begins with the treatment of some basic filter types, namely, digital resonators, notch filters, comb filters, all-pass filters, and digital sinusoidal oscillators. This is followed by a brief description of the characteristics of three widely used analog filters. Transformations are described for converting these prototype analog filters into different frequency-selective digital filters. The chapter concludes with several IIR filter designs using MATLAB.

Chapter 9, Sampling Rate Conversion: This chapter treats the important problem of sampling rate conversion in digital signal processing. Topics treated include decimation and interpolation by integer factors, sampling rate conversion by rational factor, and filter structures for sampling rate conversion.

Chapter 10, Round-off Effects in Digital Filters: The focus of this chapter is on the effects of finite-precision arithmetic to the filtering aspects in signal processing. Quantization noise introduced in analog-to-digital conversion is characterized statistically and the quantization effects in finite precision multiplication and additions are also modeled statistically. The effects of these errors in the filter output are characterized as correlated errors, called limit cycles and as uncorrelated errors, called round-off noise.

Chapter 11, Applications in Adaptive Filtering: This chapter is the first of two chapters on projects using MATLAB. Included is an introduction to the theory and implementation of adaptive FIR filters with projects in system identification, interference suppression, narrowband frequency enhancement, and adaptive equalization.

Chapter 12, Applications in Communications: This chapter focuses on several projects dealing with waveform representation and coding and with digital communications. Included is a description of pulse-code modulation (PCM), differential PCM (DPCM) and adaptive DPCM (ADPCM), delta modulation (DM) and adaptive DM (ADM), linear predictive coding (LPC), generation and detection of dual-tone multifrequency (DTMF) signals, and a description of signal detection applications in binary communications and spread-spectrum communications.

ABOUT THE SOFTWARE

The book is an outgrowth of our teaching of a MATLAB-based undergraduate DSP course over several years. Most of the MATLAB functions discussed in this book were developed in this course. These functions are collected in the book toolbox called DSPUM and are available online on the book's companion website. Many examples in the book contain MATLAB scripts. Similarly, MATLAB plots were created using scripts. All these scripts are made available at the companion website for the benefit of students and instructors. Students should study these scripts to gain insight into MATLAB procedures. We will appreciate any comments, corrections, or compact coding of these functions and scripts. Solutions to problems and the associated script files will be made available to instructors adopting the book through the companion website. To access the book's companion website and all additional course materials, please visit www.cengagebrain.com. At the CengageBrain.com home page, search for the ISBN of your title (from the back cover of your book) using the search box at the top of the page. This will take you to the product page where these resources can be found.

Further information about MATLAB and related publications may be obtained from

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