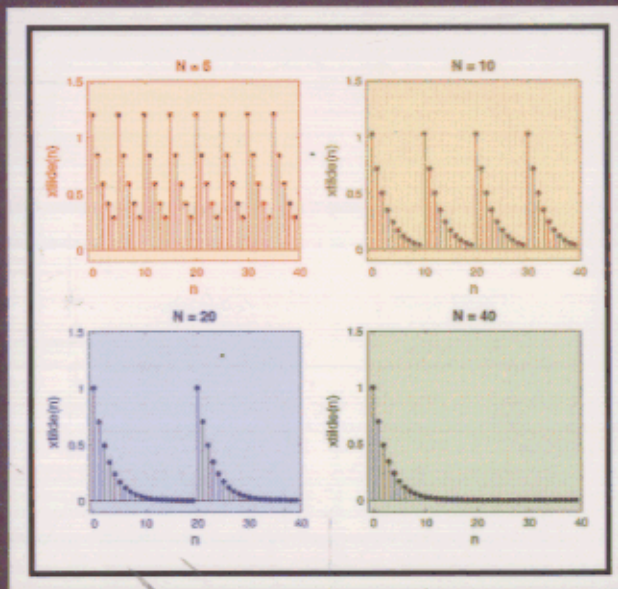


INTERNATIONAL STUDENT EDITION

Second Edition



Digital Signal Processing using MATLAB[®]

Vinay K. Ingle

John G. Proakis

Bookware Companion Series

Not for Sale in the
United States

Contents

PREFACE xi

1 INTRODUCTION 1

Overview of Digital Signal Processing 2
A Few Words about MATLAB® 6

2 DISCRETE-TIME SIGNALS AND SYSTEMS 7

Discrete-time Signals 7
Discrete Systems 20
Convolution 22
Difference Equations 29
Problems 34

3 THE DISCRETE-TIME FOURIER ANALYSIS 40

The Discrete-time Fourier Transform (DTFT)	40
The Properties of the DTFT	47
The Frequency Domain Representation of LTI Systems	53
Sampling and Reconstruction of Analog Signals	60
Problems	74

4 THE z -TRANSFORM 80

The Bilateral z -Transform	80
Important Properties of the z -Transform	84
Inversion of the z -Transform	89
System Representation in the z -Domain	95
Solutions of the Difference Equations	105
Problems	111

5 THE DISCRETE FOURIER TRANSFORM 118

The Discrete Fourier Series	119
Sampling and Reconstruction in the z -Domain	126
The Discrete Fourier Transform	131
Properties of the Discrete Fourier Transform	141
Linear Convolution Using the DFT	155
The Fast Fourier Transform	162
Problems	174

6 DIGITAL FILTER STRUCTURES 186

- Basic Elements 187**
- IIR Filter Structures 187**
- FIR Filter Structures 201**
- Lattice Filter Structures 212**
- Problems 223**

7 FIR FILTER DESIGN 231

- Preliminaries 232**
- Properties of Linear-phase FIR Filters 235**
- Window Design Techniques 250**
- Frequency Sampling Design Techniques 272**
- Optimal Equiripple Design Technique 286**
- Problems 302**

8 IIR FILTER DESIGN 313

- Some Preliminaries 314**
- Characteristics of Prototype Analog Filters 317**
- Analog-to-Digital Filter Transformations 339**
- Lowpass Filter Design Using MATLAB 357**
- Frequency-band Transformations 362**
- Comparison of FIR vs. IIR Filters 375**
- Problems 376**

9 FINITE WORD-LENGTH EFFECTS 386

- Overview 386
- Representation of Numbers 387
- The Process of Quantization and Error Characterizations 402
- Quantization of Filter Coefficients 409
- Analysis of A/D Quantization Noise 422
- Round-off Effects in IIR Digital Filters 435
- Round-off Noise in FIR Filter Realizations 462
- Problems 474

10 SAMPLING RATE CONVERSION 483

- Introduction 484
- Decimation by a Factor D 486
- Interpolation by a Factor I 495
- Sampling Rate Conversion by a Rational Factor I/D 501
- FIR Filter Designs for Sample Rate Conversion 506
- FIR Filter Structures for Sampling Rate Conversion 528
- Problems 538

11 APPLICATIONS IN ADAPTIVE FILTERING 546

- LMS Algorithm for Coefficient Adjustment 548
- System Identification or System Modeling 551
- Suppression of Narrowband Interference in a Wideband Signal 552
- Adaptive Line Enhancement 555
- Adaptive Channel Equalization 555

12 APPLICATIONS IN COMMUNICATIONS 559

Pulse-Code Modulation 559

Differential PCM (DPCM) 563

Adaptive PCM (ADPCM) and DPCM 566

Delta Modulation (DM) 570

Linear Predictive Coding (LPC) of Speech 574

Dual-tone Multifrequency (DTMF) Signals 578

Binary Digital Communications 582

Spread-Spectrum Communications 583

BIBLIOGRAPHY 587

INDEX 589