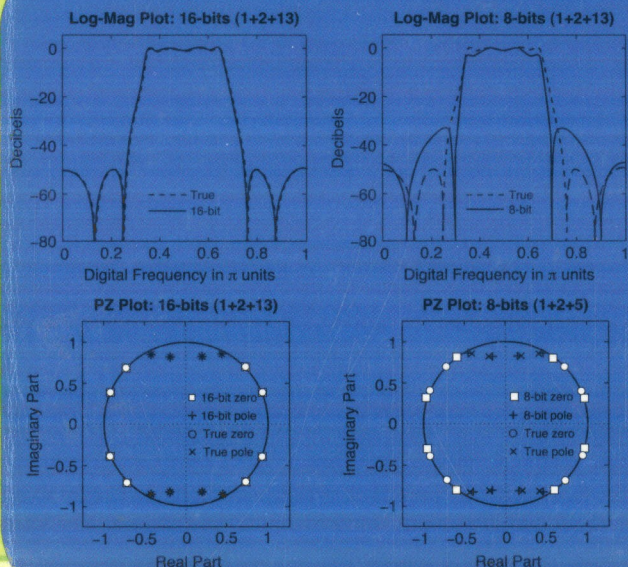
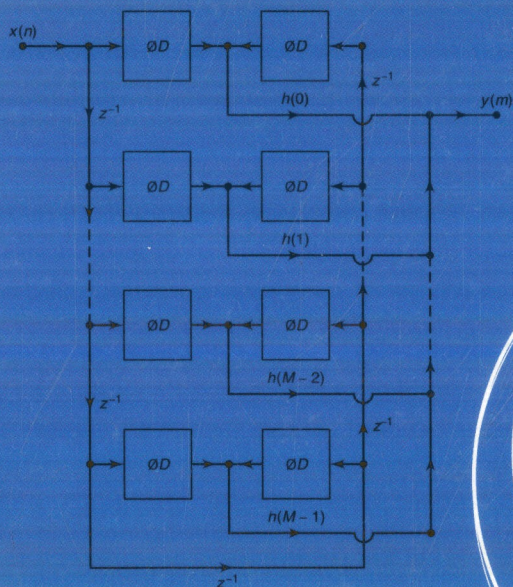


3rd edition



Essentials of Digital Signal Processing using MATLAB®

Vinay K. Ingle
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International
Edition

Contents

PREFACE xi

1 INTRODUCTION 1

- 1.1 Overview of Digital Signal Processing 2**
- 1.2 A Brief Introduction to MATLAB 5**
- 1.3 Applications of Digital Signal Processing 17**
- 1.4 Brief Overview of the Book 20**

2 DISCRETE-TIME SIGNALS AND SYSTEMS 22

- 2.1 Discrete-time Signals 22**
- 2.2 Discrete Systems 36**
- 2.3 Convolution 40**
- 2.4 Difference Equations 47**
- 2.5 Problems 53**

3 THE DISCRETE-TIME FOURIER ANALYSIS 59

- 3.1 The Discrete-time Fourier Transform (DTFT) 59**
- 3.2 The Properties of the DTFT 67**
- 3.3 The Frequency Domain Representation of LTI Systems 74**
- 3.4 Sampling and Reconstruction of Analog Signals 80**
- 3.5 Problems 97**

4 THE z -TRANSFORM 103

- 4.1 The Bilateral z -Transform 103**
- 4.2 Important Properties of the z -Transform 107**
- 4.3 Inversion of the z -Transform 112**
- 4.4 System Representation in the z -Domain 118**
- 4.5 Solutions of the Difference Equations 128**
- 4.6 Problems 134**

5 THE DISCRETE FOURIER TRANSFORM 141

- 5.1 The Discrete Fourier Series 142**
- 5.2 Sampling and Reconstruction in the z -Domain 149**
- 5.3 The Discrete Fourier Transform 154**
- 5.4 Properties of the Discrete Fourier Transform 165**
- 5.5 Linear Convolution Using the DFT 180**
- 5.6 The Fast Fourier Transform 187**
- 5.7 Problems 200**

6 IMPLEMENTATION OF DISCRETE-TIME FILTERS 212

- 6.1 Basic Elements 213**
- 6.2 IIR Filter Structures 214**
- 6.3 FIR Filter Structures 228**
- 6.4 Lattice Filter Structures 239**
- 6.5 Overview of Finite-Precision Numerical Effects 250**
- 6.6 Representation of Numbers 251**
- 6.7 The Process of Quantization and Error
Characterizations 266**
- 6.8 Quantization of Filter Coefficients 273**
- 6.9 Problems 288**

7 FIR FILTER DESIGN 303

- 7.1 Preliminaries 304**
- 7.2 Properties of Linear-phase FIR Filters 307**
- 7.3 Window Design Techniques 322**
- 7.4 Frequency Sampling Design Techniques 344**
- 7.5 Optimal Equiripple Design Technique 358**
- 7.6 Problems 375**

8 IIR FILTER DESIGN 386

- 8.1 Some Preliminaries 387**
- 8.2 Some Special Filter Types 390**

- 8.3 Characteristics of Prototype Analog Filters 400
- 8.4 Analog-to-Digital Filter Transformations 423
- 8.5 Lowpass Filter Design Using MATLAB 443
- 8.6 Frequency-band Transformations 448
- 8.7 Problems 461

9 SAMPLING RATE CONVERSION 474

- 9.1 Introduction 475
- 9.2 Decimation by a Factor D 477
- 9.3 Interpolation by a Factor I 486
- 9.4 Sampling Rate Conversion by a Rational Factor I/D 493
- 9.5 FIR Filter Designs for Sampling Rate Conversion 498
- 9.6 FIR Filter Structures for Sampling Rate Conversion 520
- 9.7 Problems 530

10 ROUND-OFF EFFECTS IN DIGITAL FILTERS 538

- 10.1 Analysis of A/D Quantization Noise 538
- 10.2 Round-off Effects in IIR Digital Filters 550
- 10.3 Round-off Effects in FIR Digital Filters 578
- 10.4 Problems 590

11 APPLICATIONS IN ADAPTIVE FILTERING 594

- 11.1 LMS Algorithm for Coefficient Adjustment 596
- 11.2 System Identification or System Modeling 599

11.3 Suppression of Narrowband Interference
in a Wideband Signal 600

11.4 Adaptive Line Enhancement 603

11.5 Adaptive Channel Equalization 603

12 APPLICATIONS IN COMMUNICATIONS 607

12.1 Pulse-Code Modulation 607

12.2 Differential PCM (DPCM) 611

12.3 Adaptive PCM and DPCM (ADPCM) 614

12.4 Delta Modulation (DM) 618

12.5 Linear Predictive Coding (LPC) of Speech 622

12.6 Dual-tone Multifrequency (DTMF) Signals 626

12.7 Binary Digital Communications 630

12.8 Spread-Spectrum Communications 632

BIBLIOGRAPHY 635

INDEX 637