1. Record Nr. UNINA9910592993003321

Autore	Palani S.
Titolo	Principles of digital signal processing / / S. Palani
Pubbl/distr/stampa	Cham, Switzerland : , : Springer, , [2022] ©2022
ISBN	9783030963224 9783030963217
Edizione	[Second edition.]
Descrizione fisica	1 online resource (689 pages)
Disciplina	621.3822
Soggetti	Signal processing - Digital techniques
Lingua di pubblicazione	Inglese
Formato	Materiale a stampa
Livello bibliografico	Monografia
Nota di bibliografia	Includes bibliographical references and index.
Nota di contenuto	Intro -- Preface to the Second Edition -- Preface to the First Edition -- Contents -- About the Author -- 1 Representation of Discrete Signals and Systems -- 1.1 Introduction -- 1.2 Terminologies Related to Signals and Systems -- 1.2.1 Signal -- 1.2.2 System -- 1.3 Continuous and Discrete Time Signals -- 1.4 Basic Discrete Time Signals -- 1.4.1 The Unit Impulse Sequence -- 1.4.2 The Basic Unit Step Sequence -- 1.4.3 The Basic Unit Ramp Sequence -- 1.4.4 Unit Rectangular Sequence -- 1.4.5 Sinusoidal Sequence -- 1.4.6 Discrete Time Real Exponential Sequence -- 1.5 Basic Operations on Discrete Time Signals -- 1.5.1 Addition of Discrete Time Sequence -- 1.5.2 Multiplication of DT Signals -- 1.5.3 Amplitude Scaling of DT Signal -- 1.5.4 Time Scaling of DT Signal -- 1.5.5 Time Shifting of DT Signal -- 1.5.6 Multiple Transformation -- 1.6 Classification of Discrete Time Signals -- 1.6.1 Periodic and Non-periodic DT Signals -- 1.6.2 Odd and Even DT Signals -- 1.6.3 Energy and Power of DT Signals -- 1.7 Discrete Time System -- 1.8 Properties of Discrete Time System -- 1.8.1 Linear and Nonlinear Systems -- 1.8.2 Time Invariant and Time Varying DT Systems -- 1.8.3 Causal and Non-causal DT Systems -- 1.8.4 Stable and Unstable Systems -- 1.8.5 Static and Dynamic Systems -- 1.8.6 Invertible and Inverse Discrete Time Systems -- 2 Discrete and Fast Fourier Transforms (DFT and FFT) -- 2.1 Introduction -- 2.2 Discrete Fourier Transform (DFT) -- 2.2.1 The Discrete Fourier Transform Pairs

-- 2.2.2 Four-Point, Six-Point and Eight-Point Twiddle Factors -- 2.2.3 Zero Padding -- 2.3 Relationship of the DFT to Other Transforms -- 2.3.1 Relationship to the Fourier Series Coefficients of a Periodic Sequence -- 2.3.2 Relationship to the Fourier Transform of an Aperiodic Sequence -- 2.3.3 Relationship to the z-Transform -- 2.4 Properties of DFT -- 2.4.1 Periodicity.
2.4.2 Linearity -- 2.4.3 Circular Shift and Circular Symmetric of a Sequence -- 2.4.4 Symmetry Properties of the DFT -- 2.4.5 Multiplication of Two DFTs and Circular Convolution -- 2.4.6 Time Reversal of a Sequence -- 2.4.7 Circular Time Shift of a Sequence -- 2.4.8 Circular Frequency Shift -- 2.4.9 Complex-Conjugate Properties -- 2.4.10 Circular Correlation -- 2.4.11 Multiplication of Two Sequences -- 2.4.12 Parseval's Theorem -- 2.5 Circular Convolution -- 2.5.1 Method of Performing Circular Convolution -- 2.5.2 Performing Linear Convolution Using DFT -- 2.6 Fast Fourier Transform (FFT) -- 2.6.1 Radix-2 FFT Algorithm -- 2.6.2 Radix-4 FFT Algorithms -- 2.6.3 Computation of IDFT through FFT -- 2.6.4 Use of the FFT Algorithm in Linear Filtering and Correlation -- 2.7 In-Plane Computation -- 3 Design of IIR Digital Filters -- 3.1 Introduction -- 3.1.1 Advantages -- 3.1.2 Disadvantages -- 3.2 IIR and FIR Filters -- 3.3 Basic Features of IIR Filters -- 3.4 Performance Specifications -- 3.5 Impulse Invariance Transform Method -- 3.5.1 Relation Between Analog and Digital Filter Poles -- 3.5.2 Relation Between Analog and Digital Frequency -- 3.6 Bilinear Transformation Method -- 3.6.1 Relation Between Analog and Digital Filter Poles -- 3.6.2 Relation Between Analog and Digital Frequency -- 3.6.3 Effect of Warping on the Magnitude Response -- 3.6.4 Effect of Warping on the Phase Response -- 3.7 Specifications of the Lowpass Filter -- 3.8 Design of Lowpass Digital Butterworth Filter -- 3.8.1 Analog Butterworth Filter -- 3.8.2 Frequency Response of Butterworth Filter -- 3.8.3 Properties of Butterworth Filters -- 3.8.4 Design Procedure for Lowpass Digital Butterworth Filters -- 3.9 Design of Lowpass Digital Chebyshev Filter -- 3.9.1 Analog Chebyshev Filter -- 3.9.2 Determination of the Order of the Chebyshev Filter.
3.9.3 Unnormalized Chebyshev Lowpass Filter Transfer Function -- 3.9.4 Frequency Response of Chebyshev Filter -- 3.9.5 Properties of Chebyshev Filter (Type I) -- 3.9.6 Design Procedures for Lowpass Digital Chebyshev IIR Filter -- 3.10 Frequency Transformation -- 3.10.1 Analog Frequency Transformation -- 3.10.2 Digital Frequency Transformation -- 3.11 IIR Filter Design by Approximation of Derivatives -- 3.12 Frequency Response from Transfer Function H(z) -- 3.13 Structure Realization of IIR System -- 3.13.1 Direct Form-I Structure -- 3.13.2 Direct Form-II Structure -- 3.13.3 Cascade Form Realization -- 3.13.4 Parallel Form Realization -- 3.13.5 Transposed Direct Form Realization -- 3.13.6 Transposition Theorem and Transposed Structure -- 3.13.7 Lattice Structure of IIR System -- 3.13.8 Conversion from Direct Form to Lattice Structure -- 3.13.9 Lattice-Ladder Structure -- 4 Finite Impulse Response (FIR) Filter Design -- 4.1 Introduction -- 4.1.1 LTI System as Frequency Selective Filters -- 4.2 Characteristic of Practical Frequency Selective Filters -- 4.3 Structures for Realization of the FIR Filter -- 4.3.1 Direct Form Realization -- 4.3.2 Cascade Form Realization -- 4.3.3 Linear Phase Realization -- 4.3.4 Lattice Structure of an FIR Filter -- 4.4 FIR Filters -- 4.4.1 Characteristics of FIR Filters with Linear Phase -- 4.4.2 Frequency Response of Linear Phase FIR Filter -- 4.5 Design Techniques for Linear Phase FIR Filters -- 4.5.1 Fourier Series Method of FIR Filter Design -- 4.5.2 Window Method -- 4.5.3 Frequency Sampling Method -- 5 Finite Word Length Effects -- 5.1 Introduction -- 5.2 Representation of Numbers in Digital System -- 5.2.1 Fixed Point

Representation -- 5.2.2 Floating Point Representation -- 5.3 Methods of Quantization -- 5.3.1 Truncation -- 5.3.2 Rounding -- 5.4 Quantization of Input Data by Analog to Digital Converter. 5.4.1 Output Noise Power Due to the Quantization Error Signal -- 5.5 Quantization of Filter Coefficients -- 5.6 Product Quantization Error -- 5.7 Limit Cycles in Recursive System -- 5.7.1 Zero-Input Limit Cycles -- 5.7.2 Overflow Limit Cycle Oscillation -- 5.8 Scaling to Prevent Overflow -- 6 Multi-rate Digital Signal Processing -- 6.1 Introduction -- 6.2 Advantages and Applications of Multi-rate Signal Processing -- 6.3 Downsampling (Decimator) -- 6.4 Upsampling (Interpolator) -- 6.5 Sampling Rate Conversion by Non-integer Factors Represented by Rational Number -- 6.6 Characteristics of Filter and Downsampler -- 6.7 Linearity and Time Invariancy of Decimator and Interpolator -- 6.7.1 Linearity of Decimator -- 6.7.2 Linearity of an Interpolator -- 6.7.3 Time Invariancy of a Decimator -- 6.7.4 Time Invariancy of an Interpolator -- 6.8 Spectrum of Downsampled Signal -- 6.9 Effect of Aliasing in Downsampling -- 6.10 Spectrum of Upsampling Signal -- 6.10.1 Anti-imaging Filter -- 6.11 Efficient Transversal Structure for Decimator -- 6.12 Efficient Transversal Structure for Interpolator -- 6.13 Identities -- 6.14 Polyphase Filter Structure of a Decimator -- 6.14.1 The Polyphase Decomposition -- 6.14.2 Polyphase Structure of a Decimator Using z-Transform -- 6.14.3 Polyphase Structure of an Interpolator -- 6.14.4 Polyphase Structure of an Interpolator Using z-Transform -- 6.15 Polyphase Decomposition of IIR Transfer Function -- 6.16 Cascading of Upsampler and Downsampler -- 6.17 Multi-stage Rating of Sampling Rate Conversion -- 6.18 Implementation of Narrow Band Lowpass Filter -- 6.19 Adaptive Filters -- 6.19.1 Concepts of Adaptive Filtering -- 6.19.2 Adaptive Noise Canceller -- 6.19.3 Main Components of the Adaptive Filter -- 6.19.4 Adaptive Algorithms -- Index.