PCEC4304 DIGITAL SIGNAL PROCESSING (3-0-0)

Module – I (10 hours)

The Z-Transform and Its Application to the Analysis of LTI Systems:

The Z-Transform: The Direct Z-Transform, The Inverse Z-Transform; Properties of the Z-Transform; Inversion of the Z-Transforms: The Inversion of the Z-Transform by Power Series Expansion, The Inversion of the Z-Transform by Partial-Fraction Expansion; Analysis of Linear Time-Invariant Systems in the z-Domain: Response of Systems with rational System Functions, Transient and Steady-State Responses, Causality and Stability, Pole-Zero Cancellations.

Selected portions from Chapter 3 (3.1.1, 3.1.2, 3.2, 3.4.2, 3.4.3, 3.5.1, 3.5.2, 3.5.3, 3.5.4) of Textbook – I

The Discrete Fourier Transform: Its Properties and Applications

Frequency Domain Sampling: Frequency-Domain Sampling and Reconstruction of Discrete-Time Signals, The Discrete Fourier Transform, The DFT as a Linear Transformation, Relationship of the DFT to other Transforms; Properties of the DFT: Periodicity, Linearity, and Symmetry Properties, Multiplication of Two DFTs and Circular Convolution, Additional DFT Properties; Linear Filtering Methods Based on the DFT: Use of the DFT in Linear Filtering, Filtering of Long Data Sequences; Frequency Analysis of Signals using the DFT; The Discrete Cosine Transform: Forward DCT, Inverse DCT, DCT as an Orthogonal Transform.

Chapter - 7 of Textbook - 1.

Module – II (10 hours)

Implementation of Discrete-Time Systems:

Structure for the Realization of Discrete-Time Systems, Structure for FIR Systems: Direct-Form Structure, Cascade-Form Structures, Frequency-Sampling Structures; Structure for IIR Systems: Direct-Form Structures, Signal Flow Graphs and Transposed Structures, Cascade-Form Structures, Parallel-Form Structures.

Selected portions from Chapter 9 (9.1, 9.2.1, 9.2.2, 9.2.3, 9.3.1, 9.3.2, 9.3.3, 9.3.4) of Textbook – I

Design of Digital Filters:

General Considerations: Causality and Its Implications, Characteristics of Practical Frequency-Selective Filters; Design of FIR Filters: Symmetric and Antisymmetric FIR Filters, Design of Linear-Phase FIR Filters by using Windows, Design of Linear-Phase FIR Filters by the Frequency-Sampling Method; Design of IIR Filters from Analog Filters: IIR Filter Design by Impulse Invariance, IIR Filter Design by the Bilinear Transformation.

Selected portions from Chapter 10 (10.1.1, 10.1.2, 10.2.1, 10.2.2, 10.2.3, 10.2.4, 10.3.2, 10.3.3) of Textbook – I

Module- III (15 hours)

Efficient Computation of the DFT: Fast Fourier Transform Algorithms

Efficient Computation of the DFT: FFT Algorithms: Direct Computation of the DFT, Radix-2 FFT Algorithms: Decimation-In-Time (DIT), Decimation-In-Time (DIF); Applications of FFT Algorithms: Efficient Computation of the DFT of two Real Sequences, Efficient Computation of the DFT a 2N-Point Real Sequence, Use of the FFT Algorithm in Linear Filtering and Correlation.

Selected portions from Chapter 8 (8.1.1, 8.1.3, 8.2.1, 8.2.2, 8.2.3) of Textbook – I

Adaptive Filters:

Application of Adaptive Filters: System Identification or System Modeling, Adaptive Channel Equalization, Adaptive Line Enhancer, Adaptive Noise Cancelling; Adaptive Direct-Form FIR Filters-The LMS Algorithm: Minimum Mean Square Error Criterion, The LMS Algorithm.

Selected portions from chapter 13 (13.1.1, 13.1.2, 13.1.5, 13.1.6, 13.2.1, 13.2.2) of Text book –I

Text Books

 Digital Signal Processing – Principles, Algorithms and Applications by J. G. Proakis and D. G. Manolakis, 4th Edition, Pearson.

Reference Book:

1. Digital Signal Processing – S.K. Mitra, TMH