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No:- **Date:**

CSXX2806: Digital Signal Processing

L-T-P-Cr: 3-0-0-3

Prerequisites: Discreet mathematics and linear algebra
Objectives: This course concerns with different concepts of Digital Signal Processing and it's need for different real world applications.

Course Outcome: By the end of this course, students will be able to:

Sl. No.	Course Outcome (CO)	Program Outcome (PO)
1	Analyze signals in both time and frequency domains using z-transforms and Discrete Fourier Transforms (DTFT).	PO1, PO2
2	Apply the Fast Fourier Transform (FFT) algorithms (DIT and DIF) to efficiently compute DFTs for various applications like spectral analysis and filtering long data sequences.	PO2, PO4, PO5
3	Design and differentiate between various types of digital filters (FIR, IIR, all-pass, etc.) based on desired frequency response characteristics.	PO1, PO3
4	Implement different FIR filter design techniques like windowing and frequency sampling to achieve specific frequency responses.	PO3, PO5
5	Apply analog filter design methodologies (Butterworth, Chebyshev) and translate them to design IIR digital filters using bilinear transformation.	PO1, PO3
6	Analyze and implement multirate signal processing techniques like decimation, interpolation, and filter banks for efficient signal processing.	PO2, PO3, PO5

UNIT –I: Review of z-transform and DTFT-Review of z-transform and DTFT. **Lectures: 4**

UNIT –II: Discrete Fourier Transform (DFT)-Frequency domain sampling (Sampling of DTFT), DFT and its inverse, zero padding, DFT as a linear transformation (matrix method), properties. Spectrum analysis using DFT. Filtering of long data sequences using DFT: overlap save method, overlap add method. **Lectures: 8**

UNIT –III: Fast Fourier Transform (FFT): Radix-2 FFT algorithms-Decimation-in-time (DIT-FFT) algorithm, Decimation-in-frequency (DIF-FFT) algorithm. Inverse DFT using FFT algorithms. Goertzel algorithm, Chirp-z transform algorithm. **Lectures: 5**

UNIT –IV: Filter Concepts-Frequency response and filter characteristics, phase delay and group delay, zero-phase filter, linear-phase filter, Simple FIR filters, Simple IIR filters, All pass filter, Minimum-phase system, Averaging filter, Comb filter, Digital resonator, Notch filter, Digital sinusoidal oscillator.

Lectures: 5

UNIT –V: FIR Digital Filter-Desirability of linear-phase filters, Frequency response of linear phase FIR filters, Filter specifications: absolute specifications, relative specifications, analog filter specifications. Design techniques: windowing, frequency sampling method, digital Hilbert transformer.

Lectures: 5

UNIT –VI: IIR Digital Filter-Analog filters, Butterworth and Chebyshev approximation. Bilinear transformation method, warping effect. Spectral transformation. Design of low pass ,high pass, band pass and band elimination filter **Lectures: 5**

UNIT –VII: Realizations of Digital Filters-FIR filter structures: direct form, cascade form, linear-phase form, FIR Lattice structure. IIR filter structures: direct form-I, direct form-II, cascade form, parallel form, All pole lattice structure, lattice-ladder (pole-zero) lattice structure. **Lectures: 5**

UNIT –VIII: Multirate Signal Processing-Decimation, Interpolation, The polyphase decomposition, Digital filter banks, Nyquist filters, Two-channel QMF. **Lectures: 5**

Text Book:

1. Digital Signal Processing by Alan V. Oppenheim, Ronald W. Schafer , PHI

Reference Book:

1. S K Mitra, Digital Signal Processing-A Computer Based Approach, Tata McGraw Hill.

2. Digital Signal Processing by John G. Proakis, Dimitris K Manolakis, Pearson.