PEI102: DIGITAL SIGNAL PROCESSING TECHNIQUES

L	T	P	Cr
3	1	2	4.5

Course objective: To understand the concepts, classifications and properties of discrete time signals and systems, to understand frequency domain analysis, awareness about filter designing and structure.

Discrete Time Signals and Systems: Introduction, Discrete time signals as array of values, Standard discrete time signals, Classification of discrete time signals, Discrete time systems and their classifications, Linear Time Invariant (LTI) Systems, Difference Equations: Finite Impulse Response (FIR) systems, Infinite Impulse Response (IIR) systems, Non-recursive Systems and Recursive Systems and representation of discrete time systems via difference equations, Correlation: Cross-correlation and Auto-correlation and their properties, Analog to Digital (A/D) Conversion: Sampling, Frequency Relationships, Aliasing, Quantization, Encoding, Sampling Theorem and Anti Aliasing Filter.

The z-Transforms: Introduction, z-transform, Properties of z-transform, Inverse z-transform, System function and Pole-zero plots from z-transform, Causality and Stability in terms of z-transform, Bilateral z-transform, Computation of z-transform

Discrete Fourier Transform (DFT) and Fast Fourier Transform (FFT): Discrete Fourier Transform and its Properties, Efficient Computation of DFT using FFT algorithms: Direct computation of the DFT, Divide and Conquer Approach, Radix-2 and Radix-4 FFT algorithms, Linear Filtering Approach to Computation of DFT.

Digital Filter Structure: Describing Equation of digital filter, Structures for FIR Systems: Direct Form Structure, Cascade Form Structure, Frequency Sampling Structure and Lattice Structure, Structure for IIR Systems: Direct Form Structures (Form-I and Form-II), Cascade Form Structure, Parallel Form Structure and Lattice Structure, Representation of Structures using Signal Flow Graph.

Design of Digital Filters: Characteristics of Practical Frequency Selective Filters, Design of FIR Filters using Windows: Rectangular, Bartlett, Hanning, Hamming and Blackman, Design of IIR Filters from Analog Filters, Frequency Transformations.

Multirate Digital Signal Processing: Introduction, Decimation by factor D, multistage implementation of sampling rate conversion, sampling rate conversion of bandpass filters.

Optimum Filters: Introduction, Forward and backward predictions, AR lattice and ARMA lattice ladder filters, Wiener filters for filtering and prediction.

Laboratory work: Calculation of Z transform, Calculation of Fourier transform, Design of FIR and IIR filters, Multirate signal processing, Design of optimum filters, realization of prediction.

Minor Project:

1. Case studies related to use of DSP application in instrumentation.

Application of DSP algorithms to condition signal from sensors.

Course learning outcome (CLO): After the completion of the course the students will be able to

1. Identify various type of discrete signal.

Recognise various types of systems.

Analyse frequency domain response of systems.

Design various type of filter.

Implement filter structures.

Recommended Books:

1. *Proakis, J.G. and Manolakis, D.K., Digital Signal Processing, Prentice Hall of India (2006 4th ed).*
2. *Rabiner, Lawrence R. and Gold, B., Theory and Applications of Digital Signal Processing, Prentice Hall of India (2000).*

Evaluation Scheme:

S.No.	Evaluation Elements	Weightage (%)
1.	MST	25
	EST	35
	Sessionals (May include Assignments/ Projects/ Tutorials/ Quizes/ Lab Evaluations)	40