Digital Signal Processing(304182)

Teaching Scheme:
Lectures: 4 Hrs/ Week

Examination Scheme:
In Semester Assessment:
Phase I : 30
End Semester Examination:
Phase II: 70

Course Objectives:

- To introduce students with transforms for analysis of Discrete time signals and systems.
- To understand the digital signal processing, sampling and aliasing
- To use and understand implementation of digital filters.
- To understand concept of sampling rate conversion and DSP processor architecture.

Course Outcomes:

After successfully completing the course students will be able to

- Understand use of different transforms and analyze the discrete time signals and systems.
- Realize the use of LTI filters for filtering different real world signals.
- Capable of calibrating and resolving different frequencies existing in any signal.
- Design and implement multistage sampling rate converter.

Unit I : DSP Preliminaries 6L

Sampling, DT signals, sampling theorem in time domain, sampling of analog signals, recovery of analog signals, and analytical treatment with examples, mapping between analog frequencies to digital frequency, representation of signals as vectors, concept of Basis function and orthogonality. Basic elements of DSP and its requirements, advantages of Digital over Analog signal processing.

Unit II : Discrete Fourier Transform 8L

DTFT, Definition, Frequency domain sampling , DFT, Properties of DFT, circular convolution, linear convolution, Computation of linear convolution using circular convolution, FFT, decimation in time and decimation in frequency using Radix-2 FFT algorithm, Linear filtering using overlap add and overlap save method, Introduction to Discrete Cosine Transform.

Unit III : Z transform 6L

Need for transform, relation between Laplace transform and Z transform, between Fourier transform and Z transform, Properties of ROC and properties of Z transform, Relation between pole locations and time domain behavior, causality and stability considerations for LTI systems, Inverse Z transform, Power series method, partial fraction expansion method, Solution of difference equations.

Unit IV : IIR Filter Design 8L

Concept of analog filter design (required for digital filter design), Design of IIR filters from analog filters, IIR filter design by approximation of derivatives, , IIR filter design by impulse invariance method, Bilinear transformation method, warping effect. Characteristics of Butterworth filters, Chebyshev filters and elliptic filters, Butterworth filter design, IIR filter realization using direct form, cascade form and parallel form, Finite word length effect in IIR filter design

Unit V : FIR Filter Design 6L

Ideal filter requirements, Gibbs phenomenon, windowing techniques, characteristics and comparison of different window functions, Design of linear phase FIR filter using windows and frequency sampling method. FIR filters realization using direct form, cascade form and lattice form, Finite word length effect in FIR filter design

Unit VI : Multirate DSP and Introduction to DSP Processor 6L

Concept of Multirate DSP, Sampling rate conversion by a non-integer factor, Design of two stage sampling rate converter, General Architecture of DSP, Case Study of TMS320C67XX, Introduction to Code composer studio. Application of DSP to Voice Processing, Music processing, Image processing and Radar processing.

Text Books

1. John G. Proakis, Dimitris G. Manolakis, " Digital Signal Processing: Principles, algorithms and applications" Fourth edition, Pearson Prentice Hall.
2. S. Salivahanan, C. Gnanpriya, " Digital Signal processing", McGraw Hill

Reference Books

1. Ifaeachor E.C, , Jervis B. W., " Digital Signal processing : Practical approach", Pearson publication
2. Dr. Shaila Apte, "Digital Signal Processing" Wiley India Publication, second edition
3. K.A. Navas, R. Jayadevan, " Lab Primer through MATLAB", PHI
4. Li Tan, Jean Jiang, " Digital Signal Processing : Fundamentals and applications" Academic press,