DIGITAL SIGNAL PROCESSING

(Core Subject)

Course Code:	**10B11EC512**	**Semester:**	**5th Semester, B. Tech (ECE)**
Credits:	**4**	**Contact Hours:**	**L-3, T-1,P-0**

Pre-requisites: Signal and System and Mathematics

Course Objectives:

The primary objective of this course is to provide a thorough understanding and working knowledge of design, implementation and analysis DSP systems.

Course Outcomes

Upon successful completion of this course the students will have developed following skills/abilities:

1. Interpret, represent and process discrete/digital signals and sytems
2. Thorough understanding of frequency domain analysis of discrete time signals.
3. Ability to design & analyze DSP systems like FIR and IIR Filter etc.
4. Practical implementation issues such as computational complexity, hardware resource limitations as well as cost of DSP systems or DSP Processors.
5. Understanding of spectral analysis of the signals

Course Contents :

Unit	Topics	Text book	Lectures
1.	**Introduction** DSP Applications, Concepts of Frequency and Filtering, Commonly used signals in DSP, characterization of LTI systems.	[2] & [1]	2
2.	**Review of Z-transform** a) Z-transform, Concepts of zeros and poles of a system, region of convergence (ROC) of z- transform[2L], b) Inverse z-transform and Properties of Z- transform[1L]	[1] & [2]	3
3.	**Frequency Domain Representation of Signals** a) Concept of spectrum [1L] b) Sampling theorem; decimation and interpolation of discrete signals. [2L] c) Frequency representation of discrete time signals: i. Discrete time Fourier transform (DTFT) [2L] ii. Discrete Fourier transform (DFT) [2L] iii. Fast Fourier transform (DIT and DIF)[2L] iv. Concepts of circular shift and convolution, [1L]	[1], &[2]	11

	b) Filtering of long data sequence[1L]		
4.	**Linear Time Invariant (LTI) Systems in Transform Domain** a) Concept of filtering – revisited, lowpass, bandpass and highpass filters [1L] b) Transfer function and the frequency response of a system [1L] c) Types of transfer functions i. FIR filters, ideal filters, linear phase filters, zero locations of linear phase FIR filters, [2L] ii. IIR filters, pole and zero locations of IIR filters, all pass filters, comb filters, stability issues for IIR filters [2L]	[1]	6
5.	**Filter Structures** a) IIR system- direct cascade and parallel form; Transposed form [2L] b) FIR system – direct and cascade form, and structure for linear phase FIR systems [2L]	[1]	4
6.	**Filter Design Techniques** a) Digital filter specifications, selection of filter type, and filter order [2L] b) FIR filter design using windowing Techniques[3L] c) FIR filter design using frequency sampling method[2L] d) IIR filter design using Impulse Invariance [1L] e) IIR filter design using bilinear transformation [1L] f) Spectral transformations for designing a filter with new characteristics based on a previously designed filter [1L] g) Finite precision: Quantization and round-off error, Finite word length effects in digital filter[2L]	[1],[2] &[4]	12
7.	**Random Signal Analysis & Spectral Estimation** a) Autocorrelation and cross correlation with examples [2L] b) power spectral density and Spectral estimation [2L]	[1] & [4]	4
8	Introduction to Digital Signal Processors		1
	Total Lectures		**43**

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Evaluation Scheme

1. Test 1 : 15 marks
2. Test 2 : 25 marks
3. Test 3 : 35 marks
4. **Internal Assessment** : 25 marks
    1. 10 Marks : Class performance, Tutorials & Assignments
    2. 10 Marks : Quizzes
    3. 5 marks : Attendance

Text Books

1. Oppenheim, Alan V., Ronald W. Schafer, and John R. Buck. Discrete-time signal processing, 2nd edition, Pearson Education.
2. Mitra, Sanjit Kumar, and Yonghong Kuo. Digital signal processing: a computer- based approach, 2nd edition, Tata McGraw-Hill.
3. Mitra, Sanjit Kumar, and Yonghong Kuo. Digital signal processing, 3rd edition, Tata McGraw-Hill.

REFERENCE BOOKS

1. Proakis, John G. Digital signal processing: principles algorithms and applications. Pearson Education India.
2. Hayes, Monson H. Digital signal processing Tata McGraw-Hill edition 2004