DIGITAL SIGNAL PROCESSING

B.E., V Semester, Electronics & Communication Engineering / Telecommunication Engineering

[As per Choice Based Credit System (CBCS) scheme]

Subject Code	15EC52	IA Marks	20
Number of Lecture Hours/Week	04	Exam Marks	80
Total Number of Lecture Hours	50 (10 Hours / Module)	Exam Hours	03
CREDITS – 04			

Course objectives: This course will enable students to

- Understand the frequency domain sampling and reconstruction of discrete time signals.
- Study the properties and the development of efficient algorithms for the computation of DFT.
- Realization of FIR and IIR filters in different structural forms.
- Learn the procedures to design of IIR filters from the analog filters using impulse invariance and bilinear transformation.
- Study the different windows used in the design of FIR filters and design appropriate filters based on the specifications.

Modules

Module-1	RBT Level
Discrete Fourier Transforms (DFT): Frequency domain sampling and reconstruction of discrete time signals. DFT as a linear transformation, its relationship with other transforms. Properties of DFT, multiplication of two DFTs- the circular convolution.	L1, L2
Module-2	
Additional DFT properties, use of DFT in linear filtering, overlap-save and overlap-add method. Fast-Fourier-Transform (FFT) algorithms: Direct computation of DFT, need for efficient computation of the DFT (FFT algorithms).	L1, L2, L3
Module-3	
Radix-2 FFT algorithm for the computation of DFT and IDFT–decimation-in-time and decimation-in-frequency algorithms. Goertzel algorithm, and chirp-z transform.	L1, L2, L3
Module-4	
Structure for IIR Systems: Direct form, Cascade form, Parallel form structures. IIR filter design: Characteristics of commonly used analog filter – Butterworth and Chebyshev filters, analog to analog frequency transformations. Design of IIR Filters from analog filter using Butterworth filter: Impulse invariance, Bilinear transformation.	L1, L2, L3
Module-5	
Structure for FIR Systems: Direct form, Linear Phase, Frequency sampling	L1, L2,

structure, Lattice structure. FIR filter design: Introduction to FIR filters, design of FIR filters using - Rectangular, Hamming, Hanning and Bartlett windows.	L3

Course Outcomes: After studying this course, students will be able to:

- Determine response of LTI systems using time domain and DFT techniques.
- Compute DFT of real and complex discrete time signals.
- Computation of DFT using FFT algorithms and linear filtering approach.
- Solve problems on digital filter design and realize using digital computations.

Question paper pattern:

- The question paper will have ten questions
- Each full question consists of 16 marks.
- There will be 2 full questions (with a maximum of three sub questions) from each module.
- Each full question will have sub questions covering all the topics under a module
- The students will have to answer 5 full questions, selecting one full question from each module.

Text Book:

Digital signal processing – Principles Algorithms & Applications, Proakis & Monalakis, Pearson education, 4th Edition, New Delhi, 2007.

Reference Books:

1. Discrete Time Signal Processing, Oppenheim & Schaffer, PHI, 2003.
2. Digital Signal Processing, S. K. Mitra, Tata Mc-Graw Hill, 3rd Edition, 2010.
3. Digital Signal Processing, Lee Tan: Elsevier publications, 2007.

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