DIGITAL SIGNAL PROCESSING

VI Semester: ECE								
Course Code	**Category**	**Hours / Week**			**Credits**	**Maximum Marks**		
AECB23	**CORE**	**L**	**T**	**P**	**C**	**CIA**	**SEE**	**Total**
		3	-	-	3	30	70	100
Contact Classes: 45	**Tutorial Classes: Nil**	**Practical Classes: Nil**				**Total Classes: 45**		

COURSE OBJECTIVES:
Students will try to learn:

I	The representation, classification and analysis of discrete time signals and systems in time and frequency domain.
II	The design and realization structures of finite and infinite impulse response filters and multi rate filters.
III	The implementation of digital filter algorithms using MATLAB tool.

COURSE OUTCOMES:
After successful completion of the course, students will be able to:

CO 1	**Illustrate**the concept of discrete time signals and systems for analysing the response of LTI system in time domain and frequency domain.
CO 2	**Compute**the convolution sum using overlap add method and overlap save methodfor filtering long duration sequences efficiently.
CO 3	**Make use** of direct form I, direct form II, cascade, lattice and parallel form structures to realize the digital filters in terms of block diagram
CO 4	**Apply** Discrete Time Fourier Transform ,Discrte Fourier Transform and its properties for spectral analysis of discrete signals .
CO 5	**Construct** the Decimation-In-Time Fast Fourier Transform and Decimation-In-Frequency Fast Fourier Transformfor reducing computational complexity of DFT.
CO 6	**Model the** Infinite impulse response and finite impulse response filtersusing various transformation techniques.
CO 7	**Analyze** the performance characteristics of digital filters to meet expected system specifications using MATLAB.
CO 8	**Interpret** the efficient implementation of sample rate conversion of digital signals to interface the digital systems with different sampling rates.
CO 9	**Develop** the polyphase filters and phase shifters using the concept of multi rate signal processing.
CO 10	**Summarize** the finite word length effects for implementing of digital filters.
CO11	**Implement** standard digital signal processing algorithms alone or as a member of a small group to meet design specifications.

MODULE-I	REVIEW OF DISCRETE TIME SIGNALS AND SYSTEMS:	Classes: 10
Discrete time signal definition; Signal classification; Elementary signals; Transformation of elementary signals; Concept of digital frequency; Discrete time system definition; System classification; Linear time invariant (LTI) system; Properties of the LTI system; Time domain analysis of discrete time systems; Impulse response; The convolution sum; Methods of evaluating the convolution sum; Filtering using overlap-save and overlap-add method; Realization of digital filters: Concept of IIR and FIR filters; Realization structures for IIR and FIR filters using direct form-I and direct form-II, cascade, lattice and parallel.		
MODULE -II	DISCRETE FOURIER TRANSFORM AND EFFICIENT COMPUTATION:	Classes: 09
Introduction to discrete time Fourier transform (DTFT); Discrete Fourier transform (DFT) definition; Properties of DFT; Linear and circular convolution using DFT; Fast-Fourier-transform (FFT): Direct computation of DFT; Need for efficient computation of the DFT (FFT algorithms); Radix-2 FFT algorithm for the computation of DFT and IDFT using decimation-in-time and decimation-in-frequency algorithms; General Radix-N FFT.		
MODULE-III	STRUCUTRE OF IIR FILTERS:	Classes: 09
Analog filters: Butterworth filters; Chebyshev type-1 & type-2 filters; Analog transformation of prototype LPF to HPF/BPF/BSF. Transformation of analog filters into equivalent digital filters using impulse invariant method and bilinear transform method; Matlab programs of IIR filters.		
MODULE -IV	SYMMETRIC AND ANTISYMMETRIC FIR FILTERS	Classes: 09
Design of linear phase FIR filters windowing and frequency sampling methods; Equiripple linear phase FIR filters; Parks-McClellan algorithm and remez algorithm; Least-mean-square error filter design; Design of FIR differentiators; Matlab programs of FIR filters; Comparison of FIR & IIR.		
MODULE -V	APPLICATIONS OF DSP:	Classes: 10
Multirate signal processing; Decimation; Interpolation; Polyphase structures for decimation and interpolation filters; Structures for rational sampling rate conversion; Applications of multirate signal processing for design of phase shifters, interfacing of digital systems with different sampling rates, sub band coding of speech signals. Analysis of finite word length effects: Representation of numbers; ADC quantization noise, coefficient quantization error, product quantization error, truncation & rounding errors; Limit cycle due to product round-off error; Round-off noise power; Limit cycle oscillations due to overflow in digital filters; Principle of scaling; Dead band effects.		

Text Books:

1. John G. Proakis, Dimitris G. Manolakis, Digital signal processing, Principles, Algorithms and Applications, Prentice Hall, 4th Edition, 2007
2. Sanjit K Mitra, Digital signal processing, A computer base approach, McGraw-Hill Higher Education, 4th Edition, 2011.
3. Emmanuel C, Ifeacher, Barrie. W. Jervis, DSP-A Practical Approach, Pearson Education, 2nd Edition, 2002.
4. A.V. Oppenheim, R.W. Schaffer, Discrete Time Signal Processing, PHI, 2nd Edition, 2006.

Reference Books:

1. Li tan, Digital signal processing: fundamentals and applications, Elsevier Science &. Technology Books, 2nd Edition, 2008.
2. Robert J.schilling, Sandra. L.harris, Fundamentals of Digital signal processing using Matlab, Thomson Engineering, 2nd Edition, 2005.
3. Salivahanan, Vallavaraj, Gnanapriya, Digital signal processing‖, McGraw-Hill Higher Education, 2nd Edition, 2009.