DIGITAL SIGNAL PROCESSING

VI Semester: ECE								
Course Code	Category	Hours / Week			Credits	Maximum Marks		
AECC33	Core	L	T	P	C	CIA	SEE	Total
		3	1	-	4	30	70	100
Contact Classes: 45	Tutorial Classes: 15	Practical Classes: Nil				Total Classes: 60		
Prerequisites: Signals and Systems								

I. COURSE OVERVIEW:

This course provides the design of discrete-time systems and analytical tools to analyze the discrete signals and systems. It focuses on the classification of discrete-time signals and systems, linear time invariant systems, discrete fourier transform, fast fourier transform algorithms, digital filter design and multi rate signal processing. Digital signal processing applications are used in speech processing, image processing, audio and video data compression, communication systems.

II. COURSE OBJECTIVES:

The students will try to learn:

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

III. COURSE OUTCOMES:

After successful completion of the course, students should be able to:

CO 1	**Illustrate** the concept of discrete time signals and systems for analyzing the response of LTI system in time domain and frequencydomain.	Understand
CO 2	**Construct** the Decimation-in-time fast fourier transform and decimation-in-frequency fast Fourier transform for reducing computational complexity of DFT.	Apply
CO 3	**Implement** the digital filters and their realization structures using various transformation technique	Apply
CO 4	**Analyze** the performance characteristics of digital filters to meetexpected system specifications using MATLAB.	Analyze
CO 5	**Interpret** the efficient implementation of sample rate conversion of digital signals to interface the digital systems with different samplingrates.	Understand
CO 6	**Identify** the errors in analog to digital conversion for tolerating finiteword length effects.	Apply

IV. COURSE SYLLABUS:

MODULE – I: REVIEW OF DISCRETE TIME SIGNALS AND SYSTEMS: (09)

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 (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: (09)
Analog filters: Butterworth filters; Chebyshev type-1 and 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 (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 and IIR.

MODULE – V: APPLICATIONS OF DSP: (09)
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 and 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.

V. 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.

VI. REFERENCE BOOKS:

1. Li tan, “Digital Signal Processing: Fundamentals and Applications”, Elsevier Science and 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.

VII. WEB REFERENCES:

1. www.edufind.com
2. https://www.tutorialspoint.com/digital_signal_processing/index.htm

VIII. E-TEXT BOOKS:

1. https://users.dimi.uniud.it/~antonio.dangelo/MMS/materials/Guide_to_Digital_Signal_Process.pdf