DIGITAL SIGNAL PROCESSING

VI Semester: ECE								
Course Code	**Category**	**Hours / Week**			**Credits**	**Maximum Marks**		
AEC012	**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**		

OBJECTIVES:

I Provide background and fundamental material for the analysis and processing of digital signals and to familiarize the relationships between continuous-time and discrete-time signals and systems.
II Study fundamentals of time, frequency and z-plane analysis and to discuss the inter-relationships of these analytic method and to study the designs and structures of digital (IIR and FIR) filters from analysis to synthesis for a given specifications.
III Introduce a few real-world signal processing applications.
IV Acquaint in FFT algorithm, multi-rate signal processing techniques and finite word length effects.

COURSE OUTCOMES:

CO1 Interpret, represent and process discrete/digital signals and systems
CO2 Understanding of time domain and frequency domain analysis of discrete time signals and systems.
CO3 Understand DFT for the analysis of digital signals & systems
CO4 Demonstrate and analyze DSP systems like FIR and IIR Filter
CO5 Understand multi rate signal processing of signals through systems.

COURSE LEARNING OUTCOMES (CLOs):

1. Understand how digital to analog (D/A) and analog to digital (A/D) converters operate on a signal and be able to model these operations mathematically.
2. Define simple non-periodic discrete-time sequences such as the impulse and unit step, and perform time shifting and time-reversal operations on such sequences.
3. Given the difference equation of a discrete-time system to demonstrate linearity, time-invariance, causality and stability, and hence show whether or not a given system belongs to the important class of causal, LTI (linear time-invariant) systems.
4. Given the impulse response of a causal LTI system, show whether or not the system is bounded-input/bounded-output (BIBO) stable.
5. Perform time, frequency and Z-transform analysis on signals.
6. From a linear difference equation of a causal LTI system, draw the Direct Form I and Direct Form II filter realizations.
7. Knowing the poles and zeros of a transfer function, make a rough sketch of the gain response.
8. Define the Discrete Fourier Transform (DFT) and the inverse DFT (IDFT) of length N.
9. Understand the inter-relationship between DFT and various transforms.
10. Understand the significance of various filter structures and effects of round-off errors.
11. Understand the fast computation of DFT and appreciate the FFT Processing.
12. Design of infinite impulse response (IIR) filters for a given specification.
13. Design of finite impulse response (FIR) filters for a given specification.
14. Compare the characteristics of IIR and FIR filters.
15. Understand the tradeoffs between normal and multi rate DSP techniques and finite length word effects.
16. Understand the signal interpolation and decimation, and explain their operation
17. Explain the cause of limit cycles in the implementation of IIR filters.

UNIT-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.		
UNIT -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.		
UNIT -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.		
UNIT -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.		
UNIT -V	APPLICATIONS OF DSP	Classes: 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 & 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.