DIGITAL SIGNAL PROCESSING

III B. Tech- II Semester | **L T P C**
Course Code: A3EC28 | **4 1 - 4**

	L	T	P	C
III B. Tech- II Semester				
Course Code: A3EC28	4	1	-	4

Course Overview:

This course covers the concepts and techniques of modern digital signal processing which are fundamental to all the signal/speech/image processing, applications. The course starts with a detailed overview of discrete-time signals and systems, representation of the systems by means of differential equations, and their analysis using Fourier and z-transforms. The sampling theory of continuous-time signals is explained next, followed by exploring the transform-based analysis of linear time-invariant (LTI) systems and their structures. Subsequently, the notion of discrete Fourier transform is introduced, followed by an overview of fast algorithms for its computation. The methods for spectral analysis of discrete-time signals are discussed next, principal methods for design of FIR and IIR filters, followed by multi-rate signal processing and finite word length effects. While this course deals largely with the theory of DSP, we will use a powerful software package, MATLAB, to look at applications of this theory, particularly Fourier analysis and digital filter design.

Course Objectives:

1. To provide background and fundamental material for the analysis and processing of digital signals.
2. To familiarize the relationships between continuous-time and discrete-time signals and systems.
3. To study fundamentals of time, frequency and z-plane analysis and to discuss the inter-relationships of these analytic method.
4. To study the designs and structures of digital (IIR and FIR) filters from analysis to synthesis for a given specifications.
5. To introduce a few real-world signal processing applications.

Course Outcomes:

- Memorize the different types of signals and systems
- Understand the significance of various digital filter structure
- Solve DFT using various FFT algorithms
- Apply the knowledge of multi-rate signal processing in the real time applications
- Design a digital filter using various techniques

SYLLABUS

UNIT–I
INTRODUCTION TO DIGITAL SIGNAL PROCESSING: Discrete time signals & systems, linear shift invariant systems, stability and causality, Discrete time systems described by difference equations, Frequency domain representation of discrete time signals and systems.

UNIT–II
FOURIER SERIES AND FOURIER TRANSFORMS: Discrete Fourier series representation of periodic sequences, Properties of discrete Fourier series, Discrete Fourier transforms: frequency domain sampling, , linear convolution of sequences using DFT, Computation of DFT, Relationship of

DFT to other transforms, Properties of DFT, Fast Fourier transforms (FFT) - Radix-2 FFT algorithm, Radix-4 FFT algorithms, Inverse FFT.

UNIT–III

RANSFORMS: Review of Z-transforms, Properties of Z-transform, Rational Z-transforms, Inversion of Z- transforms, stability and causality.

REALIZATION OF DIGITAL FILTERS: Structures for FIR systems: Direct form structure, Cascade form structures, Structures for IIR systems: Direct form structures, Signal flow graphs and transposed structures, cascade form structures, Parallel form structures.

UNIT–IV

DESIGN OF FIR DIGITAL FILTERS: Symmetric and antisymmeric FIR filters, Design of linear phase FIR Digital Filters using Windows, Design of linear phase FIR Digital Filters by Frequency Sampling method.

DESIGN OF IIR DIGITAL FILTERS: IIR filter design by Approximation of Derivatives, IIR filter design by impulse invariance, IIR filter design by bilinear transformation, Characteristics of commonly used analog filters (Butter worth and Chebyshev), Frequency transformations, comparison of IIR & FIR filters.

UNIT– V

MULTIRATE DIGITAL SIGNAL PROCESSING: Decimation by a factor D, interpolation by a factor I, sampling rate conversion by a rational factor I/D, Filter Design & Implementation for sampling rate conversion, Multi stage Implementation of sampling rate conversion.

TEXT BOOKS:

1. John G. Proakis, Dimitris G. Manolakis (2007), *Digital Signal Processing, Principles, Algorithms, and Applications*,Pearson Education / PHI, India.

2. A.V. Oppenheim, R. W. Schaffer (2009), *Discrete Time Signal Processing*, Prentice Hall of India, New Delhi.

REFERENCE BOOKS:

1. Andreas Antoniou (2006), *Digital Signal Processing*, Tata McGraw Hill, NewDelhi.

2. M. H. Hayes (2007), *Schaums Outlines of Digital Signal Processing,* Tata McGraw Hill, India.