3/4 B.Tech - SIXTH SEMESTER

EC6T4 **Digital Signal Processing** **Credits: 4**

Lecture : 4 periods/week **Internal assessment: 30 marks**
Tutorial: 1 period /week **Semester end examination: 70 marks**

Course Objectives:

- To introduce the concepts and techniques associated with discrete time signals and systems.
- To develop the representation of discrete-time signals in the frequency domain using Discrete Fourier transform (DFT) and FFT.
- To learn the basic forms of FIR and IIR filters, and how to design filters with desired frequency responses.
- To apply the concepts of DSP in various applications.

Learning Outcomes:

At the end of this course, the students will be able to

- Understand the analytical tools such as Fourier transforms, Discrete Fourier transforms, Fast Fourier Transforms and Z-Transforms required for digital signal processing.
- Get familiarized with various structures of IIR and FIR systems.
- Design and realize various digital filters for digital signal processing.
- Understand the applications of DSP in speech processing and spectrum analysis.

UNIT-I

Introduction: Introduction to Digital signal processing, Discrete time Signals, Discrete time systems, Linear Time-Invariant Systems, Analysis of LTI systems, Causality and Stability, Linear Constant Coefficient Difference equations.

UNIT-II

Applications of Z – Transforms: System Function H(z) of Digital Systems, Stability Analysis, Structure and Realization of IIR systems, Structure and Realization of FIR systems.

UNIT-III

Discrete Fourier Transform (DFT): Introduction to DFT, Properties of DFT, Circular Convolution, Linear convolution of sequences using DFT, Computation of DFT, Relation between Z-transform and DFT.

UNIT-IV

Fast Fourier Transforms (FFT): Introduction**,** Radix-2 Decimation-In-Time (DIT) and Decimation-In-Frequency (DIF) FFT Algorithms, Inverse FFT.

UNIT-V

IIR Digital Filter Design Techniques: Design of IIR Filters from analog filters, Analog Filters Approximations (Butterworth and Chebyshev Approximations), Frequency Transformations, General Considerations in Digital Filter Design, Bilinear Transformation Method, Step and Impulse Invariance Techniques.

UNIT-VI

Design of FIR Filters: Characteristics of FIR Digital Filters, frequency response of FIR filters, Design of FIR Digital Filters using Fourier Series Method, Window Function Techniques, Frequency Sampling Method, Comparison of IIR and FIR Filters.

UNIT-VII

Multirate Digital Signal Processing : Decimation, Interpolation, sampling rate conversion, Implementation of sampling rate conversion.

UNIT-VIII

Applications: Applications of FFT in Spectrum Analysis and Filtering, Application of DSP in Speech Processing.

Learning Resources

Text Books:

1. Digital Signal Processing : Principles, Algorithms and Applications , John G Proakis & D. G. Manolakis, Pearson Education, 4th Edition, 2007.
2. Discrete time Signal Processing, Alan V. Oppenheim and Ronald W. Schafer, Prentice Hall Signal Processing, 3rd Edition, 2009

References:

1. Fundamentals of Digital Signal Processing - Lonnie C Ludeman, John Wiley & Sons, 2003
2. Digital Signal Processing “A – Computer Based Approach” - Sanjit K Mitra, Tata Mc Graw Hill 2nd Edition, 2003
3. Theory and Application of Digital Signal Processing - Lawrence R Rabiner & Bernard Gold, Prentice Hall.