3/4 B.Tech. SECOND SEMESTER

EE6T6 **DIGITAL SIGNAL PROCESSING** **Credits: 3**

Lecture: 3 periods/week **Internal assessment: 30 marks**

Tutorial: 1 period /week **Semester end examination: 70 marks**

Objectives:

This course is designed to give students the required knowledge for DFT, FFT, Z Transforms and its computation and understand the design techniques for digital filters

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)

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, 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. Introduction to wavelets.

Unit VIII Applications

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

Learning Resources

Text books:

1. Proakis, J. & D. G. Manolakis. (2007), "Digital Signal Processing : Principles, Algorithms and Applications" , 4th Edition, Pearson Education
2. Alan V. Oppenheim and Ronald W. Schafer: Digital Signal Processing, PHI.

Reference Books:

1. Sanjit K. Mitra, Digital Signal Processing "A – Computer Based Approach", Tata Mc Graw Hill 2nd Edition, 2003.

2. Raddar and Rabiner, Application of Digital Signal Processing
3. Lonnie C Ludeman, "Fundamentals of Digital Signal Processing", John Wiley & Sons, 2003