Digital Signal Processing (GE)

Credits: Theory-03 **Theory Lectures: 45h**

Course Learning Objectives

The course aims to familiarize students with contemporary digital processing techniques essential across diverse application domains. Key focus areas include fundamental concepts pertaining to discrete-time signals and systems, alongside the analysis of signals in both time and frequency domains utilizing Fourier and Z transforms. Additionally, students will be introduced to the methodologies involved in the architecture and design of digital filters.

Course Learning Outcomes

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

CO1: Illustrate digital signals, systems and their significance
CO2: Master fundamental concepts of discrete-time signals, linear time-invariant systems, and transform techniques such as Z-transform and Fourier transform.
CO3: Analyze linear time-invariant systems proficiently using Fourier and Z-transform methods.
CO4: Understand and apply design techniques for Digital FIR and IIR filters, including direct methods and analog-to-digital filter conversion.
CO5: Utilize DFT for frequency analysis and implement FFT algorithms for efficient computation in signal processing applications.

Prerequisite: Basic knowledge of electronic circuits, signals and their representation, fourier series and Laplace transforms.

L-T-P: 3-0-1

Syllabus Contents

Unit I: **(10 Lectures)**

Discrete Time Sequences and Systems: Applications of Digital Signal Processing, Review of continuous time and discrete time signals and systems**,** Introduction to discrete time sequences, Properties of DT systems. LTI systems and their properties
Fourier Transform: Frequency domain representation of signals and their interpretations, Interchangeable relationships between time and frequency domains Fourier Transform, Properties of Fourier Transform, Inverse Fourier Transform, Transfer Function of LSI systems.

Unit II: **(12 Lectures)**

Z-Transform: Definition, Unilateral Z- transform, Region of Convergence and its properties, Properties of Z-Transform, Initial and final value theorem. Z-transform and its properties.
Inverse Z Transform: Long division, Partial fraction, and Residual methods. Parseval's Theorem and Applications.

System Function: Linear constant coefficient difference equation, Representation and analysis of Discrete Time Systems, Stability, Causality, Realisation of Digital Linear Systems: Block diagram, Signal Flow Graph, structure for IIR and FIR systems form structures, Cascade structures

Unit III: **(12 Lectures)**
Discrete Fourier Transform: DFT assumptions and Inverse DFT, magnitude and phase representation Matrix relations, relationship with Fourier Transform, Linear and circular convolution, properties of DFT, Computation of DFT. FFT Algorithms- Decimation in time FFT. Decimation in frequency FFT, FFT using radix 2 FFT — Butterfly structure, Concept of Gibb’s phenomenon and word length effects.

Unit IV: **(12Lectures)**
Digital Filters: Characteristics of commonly used Analog filters, Comparison of Analog and Digital Filters, Types of Digital Filters: FIR and IIR. FIR Filter realization using Windowing method, Design of IIR Filters by Approximation of Derivates, Impulse Invariant Method, Bilinear Transformation, Analog Butterworth Filter Design, and Frequency transformations.

References/Suggested Readings

1. A.V. Oppenheim and Schafer, Discrete Time Signal Processing, Prentice Hall, 1999.
2. John G. Proakis and D.G. Manolakis, Digital Signal Processing: Principles, Algorithms and Applications, Prentice Hall, 2007.
3. S. Salivahanan, Digital Signal Processing, McGraw Hill, 2015.
4. Tarun Kumar Rawat, Digital Signal Processing, Oxford University Press, 2015.
5. Monson Hayes, Digital Signal Processing: Second Edition, Schaum,s Outline Series
6. Sanjit K Mitra “Digital Signal Processing” TMH

Digital Signal Processing Lab

Course Learning Outcomes of the Lab

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

CO1: Utilize software tools to simulate, synthesize, and manipulate signals, enhancing practical understanding and application of signal processing concepts.

CO2: Employ transform techniques to represent signals and systems effectively in both time and frequency domains, enabling comprehensive analysis and interpretation.

CO3: Engage in simulation and design processes for Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters, fostering hands-on experience and proficiency in digital filter design methodologies.

(Scilab/MATLAB/Python other Mathematical Simulation software)

1. Write a program to generate discrete time Unit Impulse, Unit Step, Unit ramp and Sinusoidal sequences.
2. Write a program for shifting and scaling Discrete time systems.
3. Write a program to find the Fourier Transform of a sequence.
4. Write a program to find the pole-zero plot of a function.
5. Write a program to find a function's Z transform and inverse Z transform.
6. Write a program to find the circular convolution of two sequences.
7. Write a program to find the DFT of a sequence using the direct method.
8. Write a program to find the DFT of a sequence using FFT.
9. Magnitude Response of Low Pass Filter and High Pass Filter.
10. Design FIR Filter using Window Function.
11. Convert Analog Filter to Digital IIR Filter.