DIGITAL SIGNAL PROCESSING
ENEX 413

Lecture : 3 **Year : IV**
Tutorial : 1 **Part : I**
Practical : 1.5

Course Objectives

The objective of this course is to introduce students to the fundamental concepts and techniques of Digital Signal Processing (DSP). The course covers signal representation, Fourier analysis, Z-transform, digital filter design, and realization of digital systems. Students will learn important DSP algorithms such as convolution and Fast Fourier Transform (FFT), along with practical applications in telecommunications, audio processing, image processing, and embedded systems.

1 Introduction to Digital Signal Processing (3 hours)

1.1 Introduction to Digital Signal Processing
1.2 Components of DSP Systems
1.3 Basic Elements of DSP
1.4 Applications of DSP
1.5 Advantages of DSP

2 Signal Processing and Linear Time-Invariant Systems (7 hours)

2.1 Fourier Analysis of Discrete-Time Signals
2.2 Discrete Fourier Transform (DFT)
2.3 Fast Fourier Transform (FFT)
2.4 Linear Time-Invariant (LTI) Systems
2.5 LTI Systems in Digital Signal Processing
2.6 Importance of Convolution in Digital Signal Processing

3 Z-Transform (5 hours)

3.1 Introduction to Z-Transform
3.2 Definition of Z-Transform
3.3 Advantages of Z-Transform
3.4 Region of Convergence (ROC)
3.5 Properties of ROC
3.6 Properties of Z-Transform
3.7 Inverse Z-Transform
3.8 Transient and Steady-State Response
3.9 Applications of Z-Transform

4 Frequency Response of Digital Systems (6 hours)

4.1 Introduction

4.2 Frequency Response of Rational System Function

4.3 Frequency Response of Single Pole and Zero

4.4 Frequency Response of Multiple Poles and Zeros

5 Digital Filter Design (10 hours)

5.1 Introduction to Filter Design

5.2 Basics of Analog Filters

5.3 Advantages and Disadvantages of Digital Filters

5.4 Magnitude Characteristics of Physically Realizable Filters

5.5 IIR Filter Design: Impulse Invariance Method, Bilinear Transformation Method and comparison

5.6 Analog Filter Approximations:

- 5.61 Butterworth Approximation, Chebyshev Approximation and Elliptic Filters
- 5.62 Spectral Transformations in Digital Domain
- 5.63 Comparison of Analog and Digital Low-Pass Filters
- 5.64 Digital Low-Pass Filters

5.7 FIR Filters

- 5.71 Types of Digital Filters
- 5.72 Phase Response of FIR Filters

5.8 Windowing Techniques

- 5.81 Window Functions
- 5.82 Types of Windows
- 5.83 Attenuation and Transition Width of Windows
- 5.84 Gibbs Phenomenon
- 5.85 Design of Linear Phase FIR Filters using Window Functions
- 5.86 Frequency Sampling Method for FIR Filter Design
- 5.87 Computer-Aided Design of Equiripple Linear Phase FIR Filters
- 5.88 Practical Applications of Digital Filter Design

6 Realization of Digital Filters (10 hours)

6.1 FIR and IIR Systems

6.2 Characteristics and Comparison of FIR and IIR Systems

6.3 Recursive and Non-Recursive Systems

6.4 Filter Realization Structures

- 6.41 Direct Form Realization of FIR Filters
- 6.42 Cascade Realization

6.43 Parallel Form Realization
6.44 Lattice Structure of FIR Filters
6.45 Lattice Structure for All-Pole Systems
6.46 Lattice-Ladder Structures
6.5 Finite Word Length Effects
6.51 Number Representation
6.52 Quantization Error
6.53 Finite Word Length Effects in Digital Systems
6.54 Round-Off Error
6.55 Truncation Error
6.56 Errors due to Rounding and Truncation
6.57 Limit Cycle Oscillations in Recursive Systems
6.58 Dead Band Effects

7 DSP Processors and Architectures (4 Hours)

7.1 Introduction to DSP Processors
7.2 Classification of DSP Chips
7.3 DSP Processor Architecture
7.4 Pipelining Concepts in DSP Processors
7.5 Performance Enhancement Techniques in DSP Chips
7.6 Selection Criteria for DSP Processors

Tutorial (15 hours)

1. DSP Fundamentals and Architecture Overview
   a) DSP vs. ASP: Compare Digital Signal Processing with Analog Signal Processing in terms of precision, flexibility, power consumption, and hardware cost.
   b) Processor Architectures: Compare Von-Neumann vs. Harvard and Super Harvard (SHARC) architectures for DSP chips. Explain how MAC (Multiply-Accumulate) units and pipelining improve processing speed.
   c) DSP Hardware Selection**:** Select appropriate DSP processor criteria (fixed-point vs. floating-point, memory bandwidth, word size) for designing a real-time speech processing system vs. a mobile base station receiver.
2. LTI Analysis, DFT and FFT
   a) Test the system for Linearity, Time-Invariance, Causality, and Stability.
   b) Computing the linear convolution using both analytical and tabular methods
   c) Numerical Problem related to Circular Convolution
   d) Computation of 4 point and 8-point DFT and drawing complete butterfly flow graph.
3. Z-Transform Applications and Frequency Response
   a) Determine the Z-transform and Region of Convergence (ROC)
   b) Finding inverse z transform using Partial Fraction Expansion

c) Plotting the poles and zeros on the z-plane and evaluating its magnitude and phase response for digital systems

4. Digital IIR and FIR Filter Design
   a) Design of a digital low-pass filter using Butterworth and Chebyshev approximation technique
   b) Design an ideal low-pass FIR filter using rectangular, Hamming, Bartlett, Hanning, Blackman and Kaiser Window.
5. Filter Realization Structures & Finite Word Length Effects
   a) Draw the lattice and Lattice-Ladder structure for a digital filter
   b) Truncation versus rounding errors in 16-bit fixed-point representations and calculate the dead-band.

Practical **(22.5 hours)**

1. Computation of Energy and Power of Signals
2. Discrete-Time Convolution, DFT and FFT
3. Z-Transform and Inverse Z-Transform
4. Frequency Response Analysis of Digital Systems
5. Design of Infinite Impulse Response (IIR) Filters
6. Design of Finite Impulse Response (FIR) Filters

Evaluation Scheme:

There will be questions covering all the chapters in the syllabus. The evaluation scheme for the question will be as indicated in the table below:

Chapters	Hours	Marks
1 and 2	10	12
3	5	8
4	6	8
5	10	16
6	10	12
7	4	4
Total	**45**	**60**

References

1. Gautam, G. (2017). Signal analysis. Shree Chandeshwori Publication.
2. Gautam, G. (2022). Digital signal processing (Latest ed.). Shree Chandeshwori Publication.
3. Oppenheim, A. V., & Schafer, R. W. (2009). Discrete-time signal processing (3rd ed.). Pearson Education.
4. Oppenheim, A. V., Willsky, A. S., & Nawab, S. H. (1997). Signals and systems (Latest ed.). Pearson Education.
5. Proakis, J. G., & Manolakis, D. G. (2007). Digital signal processing (Latest ed.). Pearson Education.