SEMESTER-VI
DEPARTMENT OF ELECTRONIC SCIENCE
Category I

(B.Sc. Honours in Electronics)

DISCIPLINE SPECIFIC CORE COURSE – 16: Digital Signal Processing

CREDIT DISTRIBUTION, ELIGIBILITY AND PRE-REQUISITES OF THE COURSE

Course title & Code	Credits	Credit distribution of the course			Eligibility criteria	Pre-requisite of the course (if any)
		Lecture	Tutorial	Practical/ Practice		
Digital Signal Processing	4	3	-	1	Class XII passed with Physics + Mathematics/Applied Mathematics + Chemistry OR Physics + Mathematics/Applied Mathematics + Computer Science/Informatics Practices	Signals and Systems (DSC 9, Sem III)

Learning Objectives

The Learning Objectives of this course are as follows:
To introduce the techniques of modern digital processing that are fundamental to a wide variety of application areas. Special emphasis is placed on the basic concepts related to discrete-time signals and systems, the analysis of signals in time and frequency using Fourier and Z transform. Introduction to techniques involved in the architecture and design of digital filters.

Learning outcomes

The Learning Outcomes of this course are as follows:

- Grasp fundamentals of discrete time signals, linear time-invariant systems, Z-transform and Fourier transform
- Analyze linear time-invariant systems using Fourier and Z transform

- Understand the Design techniques of Digital FIR and IIR filters using direct methods and methods involving conversion of the analog filter into the digital filter by various transformations.
- Use DFT to perform frequency analysis of signals and application of FFT algorithms.

SYLLABUS OF ELDSC-16 Total Hours- Theory: 45 Hours, Practicals: 30 Hours

UNIT – I (10 Hours)

Discrete Time Sequences and Systems: Introduction to Discrete Time sequences, Properties of DT systems.

Fourier Transform: Fourier Transform, Properties of Fourier Transform, Inverse Fourier Transform, Transfer Function of LSI systems.

UNIT – II (12 Hours)

Z-Transform: Definition, Unilateral Z- transform, Region of Convergence and its properties, Properties of Z-Transform, Initial and final value theorem.

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

UNIT – III (12 Hours)

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 (11 Hours)

Digital Filters: Comparison of Analog and Digital Filters, Types of Digital Filters: FIR and Hanning, Hamming, Blackman, Design of IIR Filters by Approximation of Derivates, Impulse Invariant Method, Bilinear Transformation, Butterworth Filter.

Practical component (if any) – Digital Signal Processing
(Scilab/MATLAB/Python other Mathematical Simulation software)

Learning outcomes

The Learning Outcomes of this course are as follows:

- Simulate, synthesize and process signals using a software tool.

- Apply transform methods for representing signals and systems in the time and frequency domain.
- Simulation and design of FIR and IIR Filters

LIST OF PRACTICALS (Total Practical Hours- 30 Hours)

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

Note: Students shall sincerely work towards completing all the above listed practicals for this course. In any circumstance, the completed number of practicals shall not be less than nine.

Essential/recommended 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.

Suggestive readings

1. S. Salivahanan, Digital Signal Processing, McGraw Hill, 2015.
2. Tarun Kumar Rawat, Digital Signal Processing, Oxford University Press, 2015.
3. Monson Hayes, Digital Signal Processing: Second Edition, Schaum's Outline Series

Note: Examination scheme and mode shall be as prescribed by the Examination Branch, University of Delhi, from time to time.