DIGITAL SIGNAL PROCESSING AND APPLICATION

ENEX 304

Lecture	**: 3**	**Year : III**
Tutorial	**: 1**	**Part : I**
Practical	**: 3/2**	

Course Objectives:

The objective of this course is to impart fundamental knowledge and techniques for processing discrete-time signals and systems, including analog-to-digital and digital-to-analog conversion and the relationship between continuous-time and discrete-time signals. It emphasizes the analysis, design, and implementation of Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) digital filters with desired frequency responses. By the end of the course, students will be able to apply the Discrete Fourier transform (DFT) using Fast Fourier transform (FFT) algorithms for practical signal processing applications in electrical engineering.

1 Discrete Time Signals and Systems (6 hours)

1.1 Basic elements of digital signal processing
1.2 Need of digital signal processing over analog signal processing
1.3 Sampling of continuous time signal, spectral properties of sampled signal
1.4 Discrete time signal, basic signal types
1.5 Transformation of independent variable
1.6 Energy signal and power signal
1.7 Periodicity of discrete time signal
1.8 Discrete time Fourier transform and properties
1.9 Discrete time system properties
1.10 Linear time invariant (LTI) system, properties of LTI system

2 Z-transform (3 hours)

2.1 Definition, Convergence of Z-transform and region of convergence
2.2 Properties of Z-transform (Linearity, time shift, multiplication by exponential sequence, differentiation, time reversal, convolution, multiplication)
2.3 Inverse z-transform by long division and partial fraction expansion

3 Analysis of LTI System in Frequency Domain (5 hours)

3.1 Frequency response of LTI system, response to complex exponential
3.2 Linear constant co-efficient difference equation and corresponding system function
3.3 Relationship of frequency response to pole-zero of system
3.4 Linear phase of LTI system and its relationship to causality

4 Discrete Filter Structures (6 hours)

4.1 FIR filter, structures for FIR filter (Direct form, cascade, lattice)
4.2 IIR filter, structures for IIR filter (Direct form I, direct form II, cascade, lattice, lattice ladder)
4.3 Limit cycles

5 FIR Filter Design (9 hours)

5.1 Filter design by window method, commonly used windows (Rectangular window, Bartlett window, Hanning window, Hamming window, Blackman window)
5.2 Filter design by Kaiser window
5.3 Filter design using optimum approximation, Remez exchange algorithm
5.4 Types of FIR filters (Type-1, Type-2, Type-3 and Type-4)

6 IIR Filter Design (9 hours)

6.1 Filter design by impulse invariance method
6.2 Filter design using bilinear transformation
6.3 Design of digital low pass Butterworth filter
6.4 Frequency transformation of lowpass IIR filters (Transformation of lowpass digital filter prototype to high-pass, band-pass and band-stop filters)

7 Discrete Fourier Transform (5 hours)

7.1 Discrete Fourier transform (DFT) representation, properties of DFT (Linearity, time shift, frequency shift, conjugation and conjugate symmetry, duality, convolution, multiplication), circular convolution
7.2 Fast Fourier transform (FFT) algorithm (Decimation in time algorithm, decimation in frequency algorithm)
7.3 Computational complexity of FFT algorithm

8 Applications of Digital Signal Processing (2 hours)

8.1 DSP application in power system monitoring and diagnostics (Fault detection, power quality analysis)
8.2 DSP application in protection and control systems (Smart grid management)
8.3 DSP application in reliability and maintenance

Tutorial (15 hours)

1. Problems related to the conversion of analog signal to discrete time systems
2. Problems related to identify whether the given signal is periodic or non-periodic signal
3. Numerical problems to determine whether the given system is linear or non-linear, time invariant or time varying, causal or non-causal, stable or unstable

4. Problems related with linear convolution
5. Numerical problems related to z-transform and inverse z-transform
6. Solutions of difference equations
7. Solve the numerical problems related to frequency response of the LTI system
8. Draw the direct form-I, direct form-II, cascaded and lattice structure for FIR and IIR filters
9. Design of FIR filters using different windowing techniques (Rectangular window, Bartlett window, Hanning window, Hamming window, Blackman window, Kaiser window)
10. Design of Low pass IIR filters using impulse invariance method and bilinear transformation
11. Problems related to the frequency transformation of lowpass IIR filters to high pass, band pass and band stop filters (Using transformation equations)
12. Numerical problems depicting examples of DIT-FFT and DIF-FFT

Practical **(22.5 hours)**

1. Introduction to Continuous Time and Discrete Time Signals
2. Linear Convolution
3. Linear Time Invariant System (LTI), Cascade of second order systems
4. Design of IIR filter (Impulse Invariance and Bilinear Transformation)
5. Design of FIR filter using window method. Comparison of FIR filter for different windowing method
6. Introduction to FFT

Final Exam

The questions will cover all the chapters in the syllabus. The evaluation scheme will be as indicated in the table below:

Chapter	Hours	Marks distribution*
1	6	8
2	3	4
3	5	6
4	6	8
5	9	12
6	9	12
7, 8	7	10
Total	**45**	**60**

* There may be minor deviation in marks distribution.

References

1. Oppenheim, A.V., Schafer, R.W., Buck, J.R. (2009). Discrete-time signal processing. Pearson Education.
2. Proakis, J.G., Manolakis, D.G. (2021). Digital signal processing: Principles, algorithms, and applications. Pearson.

3. Ludeman, L. (2009). Fundamentals of digital signal processing. John Wiley & Sons.
4. Tan, L., Jiang, J. (2025). Digital signal processing: Fundamentals, applications, and deep learning. Academic Press.
5. Schilling, R.J., Harris, S.L. (2004). Fundamentals of digital signal processing using MATLAB. CL Engineering.
6. ElAli, T.S. (2011). Discrete systems and digital signal processing with MATLAB. CRC Press.
7. Kani, A.N. (2021). Digital signal processing. CBS Publishers & Distributors.