B.Sc. (Honours) Physics

DISCIPLINE SPECIFIC ELECTIVE COURSE – DSE 20: DIGITAL SIGNAL PROCESSING

Course Title and Code	Credits	Credit distribution of the course			Pre-requisite of the course
		Lecture	Tutorial	Practical	
Digital Signal Processing DSE 20	4	2	0	2	

COURSE OBJECTIVES

- This paper describes the discrete-time signals and systems, Fourier Transform Representation of Aperiodic Discrete-Time Signals.
- This paper also highlights the concept of filters and realization of Digital Filters.
- At the end of the syllabus, students will develop an understanding of Discrete and fast Fourier Transform.

LEARNING OUTCOMES

At the end of this course, students will be able to develop following learning outcomes:

- Students will learn basic discrete-time signal and system types, convolution sum, impulse and frequency response concepts for linear time-invariant (LTI) systems.
- The student will be in position to understand use of different transforms and analyze the discrete time signals and systems. They will learn to analyze a digital system using z-transforms and discrete time Fourier transforms, region of convergence concepts, their properties and perform simple transform calculations.
- The student will realize the use of LTI filters for filtering different real world signals. The concept of transfer Function and difference-Equation System will be introduced. Also, they will learn to solve Difference Equations.
- Students will develop an ability to analyze DSP systems like linear-phase, FIR, IIR, All-pass, averaging and notch Filter etc.
- Students will be able to understand the discrete Fourier transform (DFT) and realize its implementation using FFT techniques.
- Students will be able to learn the realization of digital filters, their structures, along with their advantages and disadvantages. They will be able to design and understand different types of digital filters such as finite & infinite impulse response filters for various applications.

SYLLABUS OF DSE 20
THEORY COMPONENT
(Hours: 30)

Unit I **(7 Hours)**

Discrete-Time Signals and Systems: Classification of Signals, Transformations of the Independent Variable, Periodic and Aperiodic Signals, Energy and Power Signals, Even and Odd Signals, Discrete-Time Systems, System Properties. Impulse Response, Convolution Sum; Graphical and Analytical Method, Properties of Convolution (General Idea); Sum Property System Response to Periodic Inputs, Relationship Between LTI System Properties and the Impulse Response.

Unit II **(9 Hours)**

Discrete-Time Fourier Transform: Fourier Transform Representation of Aperiodic Discrete-Time Signals, Periodicity of DTFT, Properties; Linearity; Time Shifting; Frequency Shifting; Differencing in Time Domain; Differentiation in Frequency Domain; Convolution Property. The z-Transform: Bilateral (Two-Sided) z-Transform, Inverse z- Transform, Relationship Between z-Transform and Discrete-Time Fourier Transform, z-plane, Region-of-Convergence; Differentiation in the z-Domain; Power Series Expansion Method (General Idea). Transfer Function and Difference-Equation System.

Unit III **(10 Hours)**

Filter Concepts: Phase Delay and Group delay, Zero-Phase Filter, Linear-Phase Filter, Simple FIR Digital Filters (only qualitative treatment).
Discrete Fourier Transform: Frequency Domain Sampling (Sampling of DTFT), Discrete Fourier Transform (DFT) and its Inverse, DFT as a Linear transformation, Properties; Periodicity; Linearity; Circular Time Shifting; Circular Frequency Shifting; Circular Time Reversal; Multiplication Property; Parseval's Relation (general idea), Linear Convolution Using the DFT (Linear Convolution Using Circular Convolution).

Unit IV **(4 Hours)**

Realization of Digital Filters: FIR Filter structures; Direct-Form; Cascade-Form
Finite Impulse Response Digital Filter: Advantages and Disadvantages of Digital Filters, Types of Digital Filters: FIR Filters.

PRACTICAL COMPONENT: DIGITAL SIGNAL PROCESSING
(Hours: 60)

Sessions on the review of experimental data analysis, sources of error and their estimation in detail, writing of scientific laboratory reports including proper reporting of errors and application to the specific experiments done in the lab.

At least 06 experiments from the following using Scilab/Matlab/Python.

1. Write a program to generate and plot the following sequences: (a) Unit sample sequence $\delta(n)$, (b) unit step sequence *u(n)*, (c) ramp sequence *r(n)*, (d) real valued exponential sequence $x(n) = (0.8)^n u(n)$ for $0 \leq n \leq 50$.
2. Write a program to compute the convolution sum of a rectangle signal (or gate function) with itself for N = 5

$$\mathrm{x(n)} = \mathrm{rect}\left(\frac{\mathrm{n}}{2\mathrm{N}}\right) = \prod \quad \left(\frac{\mathrm{n}}{2\mathrm{N}}\right) = \{1 \quad -\mathrm{N} \leq \mathrm{n} \leq \mathrm{N}\ 0 \qquad \text{otherwise}$$

3. An LTI system is specified by the difference equation: y(n)=0.8y(n-1)+x(n)
   (a) Determine $\mathrm{H}(\mathrm{e}^{\mathrm{iw}})$
   (b) Calculate and plot the steady state response y(n) to $\mathrm{x(n)} = \cos\cos\,(0.5\pi\mathrm{n})\ \mathrm{u(n)}$
4. Given a casual system y(n)=0.9y(n-1)+x(n)
   (a) Find H(z) and sketch its pole-zero plot
   (b) Plot the frequency response $\left|H(e^{jw})\right|$ and $\angle H(e^{jw})$
5. Design a digital filter to eliminate the lower frequency sinusoid of *x(t)=sin7t+sin200t*. The sampling frequency is 500 Hz. Plot its pole zero diagram, magnitude response, input and output of the filter.
6. Let *x(n)* be a 4-point sequence:
   $x(n) = \{1,1,1,1\} = \{1 \;\; 0 \leq n \leq 3\ 0 \;\; otherwise$

   Compute the DTFT $X(e^{jw})$ and plot its magnitude
   (a) Compute and plot the 4 point DFT of *x(n)*
   (b) Compute and plot the 8 point DFT of *x(n)* (by appending 4 zeros)
   (c) Compute and plot the 16 point DFT of *x(n)* (by appending 12 zeros)
7. Let *x(n)* and *h(n)* be the two 4-point sequences,
   x(n)={1,2,2,1} *h(n)={1,-1,-1,1}*

   Write a program to compute their linear convolution using circular convolution.
8. Using a rectangular window, design a FIR low-pass filter with a pass-band gain of unity, cut off frequency of 1000 Hz and working at a sampling frequency of 5 KHz. Take the length of the impulse response as 17.

9. Design an FIR filter to meet the following specifications:

   Passband edge F_p=2 KHz
   Stopband edge F_s=5 KHz
   Passband attenuation A_p=2 dB
   Stopband attenuation A_s=42 dB
   Sampling frequency F_{sf}=20 KHz

10. The frequency response of a linear phase digital differentiator is given by

$$\mathrm{H_d}\left(\mathrm{e}^{\mathrm{jw}}\right) = \mathrm{jwe}^{-\mathrm{j}\tau\mathrm{w}}\ |\mathrm{w}| \leq \pi$$

   Using a Hamming window of length M = 21, design a digital FIR differentiator. Plot the amplitude response

REFERENCES

Essential Readings for the Theory Component

1. Digital Signal Processing, Tarun Kumar Rawat, 2015, Oxford University Press, India
2. Digital Signal Processing, S. K. Mitra, McGraw Hill, India.
3. Principles of Signal Processing and Linear Systems, B.P. Lathi, 2009, 1st Edn. Oxford University Press.
1. Fundamentals of signals and systems, P.D. Cha and J.I. Molinder, 2007, Cambridge University Press.
2. Digital Signal Processing Principles Algorithm & Applications, J.G. Proakis and D.G. Manolakis, 2007, 4th Edn., Prentice Hall.

Additional Readings for the Theory Component

1. Digital Signal Processing, A. Anand Kumar, 2nd Edition, 2016, PHI learning Private Limited.
2. Digital Signal Processing, Paulo S.R. Diniz, Eduardo A.B. da Silva, Sergio L. Netto, 2nd Edition, 2017, Cambridge University Press.

References for the Practical Component

1. A Guide to MATLAB, B.R. Hunt, R.L. Lipsman, J.M. Rosenberg, 2014, 3rd Edn., Cambridge University Press.
2. Fundamentals of Digital Signal processing using MATLAB, R.J. Schilling and S.L. Harris, 2005, Cengage Learning.
3. Getting started with MATLAB, Rudra Pratap, 2010, Oxford University Press.