SCHOOL CORE

Course Code	Course Title	L	T	P	S	C
22EC104009	**DIGITAL SIGNAL PROCESSING**	2	-	2	4	4

Pre-Requisite Signals and Systems

Anti-Requisite -

Co-Requisite -

COURSE DESCRIPTION: This course provides an analysis on Continuous and discrete signals and sequences; systems**;** DFT and FFT algorithms for the analysis of discrete sequences; design and realization of Digital IIR and FIR filters.

COURSE OUTCOMES: After successful completion of this course, the students will be able to:

CO1 Analyze discrete-time systems using suitable transforms.

CO2 Apply Discrete and Fast Fourier Transforms to analyze the response of linear systems.

CO3 Design IIR and FIR digital filters by applying transformation and windowing Techniques.

CO4 Realize IIR and FIR digital filters using various structures.

CO5 Work independently and in teams to solve problems with effective communication.

CO-PO-PSO Mapping Table:

Course Outcomes	Program Outcomes												Program Specific Outcomes		
	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	PSO1	PSO2	PSO3
CO1	3	3	-	2	1	-	-	-	-	-	-	-	-	3	-
CO2	3	2	-	2	3	-	-	-	-	-	-	-	-	3	-
CO3	3	2	3	2	2	-	-	-	-	-	-	-	-	3	-
CO4	3	-	-	-	-	-	-	-	-	-	-	-	-	3	-
CO5	-	-	-	-	-	-	-	-	3	3	-	-	-	3	-
Course Correlation Mapping	**3**	**3**	**3**	**2**	**3**	**-**	**-**	**-**	**3**	**3**	**-**	**-**	**-**	**3**	**-**

Correlation Levels: 3: High; 2: Medium; 1: Low

COURSE CONTENT

Module 1: FREQUENCY ANALYSIS OF DISCRETE TIME SIGNALS ***(06 Periods)***

Fourier series for DT periodic signal and power density spectrum, the Fourier transform of DT aperiodic signals, convergence of the Fourier transform and energy density spectrum, Solution for difference equations of digital filters using Z-transforms.

Module 2: DISCRETE AND FAST FOURIER TRANSFORMS ***(07 Periods)***

Discrete Fourier Transforms (DFT): Properties of DFT, linear filtering methods based on DFT, frequency analysis of signals using DFT.

Fast Fourier transforms (FFT): Radix-2 Decimation in time (DIT) and Decimation in frequency (DIF) FFT algorithms.

Module 3: IIR FILTER DESIGN ***(06 Periods)***

Design of IIR digital filters from analog filters-IIR filter design by approximation of derivatives, impulse invariance and bilinear transformation. Characteristics of commonly used analog filters, Frequency transformations.

Module 4: FIR FILTER DESIGN ***(06 Periods)***

Symmetric and anti-symmetric FIR filters, Design of linear phase FIR digital filters using windows- Barlett, Blackman, Hamming and Hanning. Frequency sampling technique.

Module 5: REALIZATION OF DISCRETE-TIME SYSTEMS ***(05 Periods)***

Structural realization of IIR Systems-direct, cascade and parallel form structures.

Structural realization of FIR Systems-direct, cascade-form structures and Lattice structures.

Total Periods: 30

EXPERIENTIAL LEARNING

LIST OF EXERCISES:

1. Introduction to Code Composer Studio and Digital Signal Processor.
2. Verify linear convolution of aperiodic sequences using CCS on DSP processors andalso verify using MATLAB.
3. Verify the circular convolution on Periodic sequences using CCS on DSP processors and also verify using MATLAB.
4. Verify N-point DFT & IDFT using CCS on DSP processors and also verify usingMATLAB.
5. Verify N-point FFT algorithm using CCS on DSP processors and also verify usingMATLAB.
6. Find the frequency response of analog Butterworth prototype filters (LP/HP/BP/BR)using MATLAB.

7. Find the frequency response of analog chebyshev prototype filters (LP/HP/BP/BR)using MATLAB.
8. Design FIR filter (LP/HP/BP/BR) using following windowing techniques with MATLAB
    1. Barlett window
    2. Blackman window
9. Design FIR filter (LP/HP/BP/BR) using following windowing technique with MATLAB
    1. Hamming window
    2. Hanning window
10. Design FIR filter (LP/HP/BP/BR) using Frequency sampling technique with MATLAB.
11. Implement IIR Butterworth filter (LP/HP/BP/BR) using bilinear transformationtechniques with MATLAB.
12. Implement IIR Chebyshev filter (LP/HP/BP/BR) using impulse-invariance transformation techniques with MATLAB.

PROJECT BASED LEARNING

1. Voice biometric speaker recognition
2. Identification of Musical Instruments
3. Speaker recognition system based on MFCC
4. Disease detection based on ECG
5. Implementation of 5-Band Audio Equalizer in MATLAB

(Note: It's an indicative one. The course instructor may change the activities and the same shall be reflected in course handout.)

RESOURCES

TEXT BOOKS:

1. J. G. Proakis and D.G. Manolakis, "*Digital Signal Processing: Principles, Algorithms andApplications,*" Prentice Hall, Fourth Edition, 2007.
2. B.Venkataramani, M. Bhaskar, "*Digital Signal Processors – Architecture, Programming andApplications,*" TATA McGraw Hill, Second Edition, 2010.

REFERENCE BOOKS:

1. Alan. V. Oppenheim, Ronald.W. Schafer and John.R. Buck, *"Discrete-Time Signal Processing,"* Pearson Education, Second Edition, 2006.
2. Emmanuel C. Ifeachor& Barrie. W. Jervis, "*Digital Signal Processing*," Pearson Education / Prentice Hall, Second Edition, 2002.

VIDEO LECTURES:

1. https://www.digimat.in/nptel/courses/video/117102060/L01.html
2. https://archive.nptel.ac.in/courses/108/105/108105055/
3. https://www.coursera.org/specializations/digital−signal−processing
4. https://www.coursera.org/learn/dsp1

Web Resources:

1. https://www.tutorialspoint.com/digital_signal_processing/digital_signal_processing_useful_ resources.htm