16EC308 DIGITAL SIGNAL PROCESSING

Hours Per Week :

L	T	P	C
3	-	2	4

Course Description and Objectives:

This course covers the analysis and representation of discrete-time signal systems and design digital filters.The course objective is to make student understand digital systems and design digital filters including discrete-time convolution, difference equations, the z-transform, and the discrete-time Fourier transform.

Course Outcomes:

Upon successful completion of this course, students should be able to:

CO1: Understand the generations and basic concepts of Digital Signal Processor architecture.

CO2: Understand the basics of discrete time signals and systems.

CO3: Apply the concepts of transform techniques in realizing discrete time signals.

CO4: Analyze various transform properties for discrete time signals.

CO5: Design of analog and digital Filters for a given specification.

CO6: Verify various transform techniques and filters.

SKILLS:

- ✓ *Simulate the response of the system from impulse or step stimulus.*
- ✓ *Identify the accelerating methods for processing through DFT & FFT.*
- ✓ *Implement FFT and Inverse FFT.*
- ✓ *Identify the type and order of the filter for the given application.*
- ✓ *Design FIR/ IIR filters for removing unwanted frequencies in the signal.*
- ✓ *Remove the power hum in electronic systems through notch filter implementation.*
- ✓ *Remove the echo in the audio system using DSP processor.*
- ✓ *Analyze the stability of the designed filter.*

ACTIVITIES:

- o *Find the response of the given LTI system using impulse input.*
- o *Find the response of the given LTI system using step input.*
- o *Compute the DFT of a given system.*
- o *Compare the 8/ 16 point FFT of the given system with DFT.*
- o *Test the stability of given third order filter.*
- o *Remove the surrounding motor sound in the MIC signal output.*

UNIT - 1 **L-9**

INTRODUCTION TO DISCRETE TIME SIGNALS AND SYSTEMS AND Z-DOMAIN ANALYSIS: Review of signals and systems, Linear shift invariant systems, Stability and causality, Linear constant coefficient difference equations, Impulse response, Step response, Response to arbitrary inputs, Frequency domain representation of discrete time signals and systems, Z-Transform and properties, Analysis of linear time invariant systems using Z-domain.

UNIT - 2 **L-9**

DFT AND FFT: Discrete fourier representation of periodic sequences (DTFT), Properties, Frequency response, Discrete fourier transform, Properties of DFT, Linear convolution of sequences using DFT, Computation of DFT, Fast fourier transforms (FFT) - Radix-2 decimation in time and decimation in frequency FFT algorithms, Inverse FFT, Radix-4 FFT.

UNIT - 3 **L-9**

FIR FILTER DESIGN AND REALIZATION: FIR system function, Characteristics of FIR digital filters, Frequency response, Design of FIR digital filters using window techniques, Frequency sampling technique, Structures of FIR - Direct form structure, Cascade form structure, Linear phase structure, Signal flow graphs and transposed structures.

UNIT - 4 **L-9**

IIR FILTER DESIGN AND REALIZATION: IIR system function, Analog filter approximations, Butter worth and Chebyshev, Design of IIR digital filters from analog filters, Analog-to-Digital transformations, Structures of IIR - Direct form I and II, Cascade form, Parallel form, Signal flow graphs and transposed Structures, Comparison of IIR & FIR filters.

UNIT - 5 **L-9**

DIGITAL SIGNAL PROCESSORS: Introduction, DSP processor memory architecture, Pipelining, Overview of TMS320 family DSP processor, First generation to sixth generation, ADSP processor, Selection of digital signal processors.

LABORATORY EXPERIMENTS

LIST OF EXPERIMENTS Total hours-30

1. To verify linear convolution and correlation.
2. To find and sketch impulse and step response.
3. To find the FFT of given 1-D signal and plot.
4. To verify circular convolution.
5. FIR filter design using different window techniques.
6. IIR filter design using analog approximations.
7. Spectrum analysis using DFT.

TEXT BOOKS:

1. John G. Proakis and Dimitris G.Manolakis, “Digital Signal Processing – Principles, Algorithms and Applications”, 4th edition, Pearson Education/Prentice Hall, 2007.
2. Avtar Singh and S. Srinivasan, “Digital Signal Processing – Implementations using DSP Microprocessors with Examples from TMS320C54xx”, Cengage Learning India Private limited, 2012.

REFERENCE BOOKS:

1. A.V.Oppenheim, R.W. Schafer and J.R. Buck, “Discrete-Time Signal Processing”, 3rd edition, Pearson, 2009.
2. Sanjit K. Mitra, “Digital Signal Processing - A Computer Based Approach”, 4th edition, Tata McGraw Hill, 2010.
3. Salivahanan, “Digital Signal Processing”, 3rd edition, McGraw Hill, 2015.
4. Emmanuel C. Ifeachor and Barrie.W.Jervis, “Digital Signal Processing”, 2nd edition, Pearson Education/Prentice Hall, 2002.
5. Andreas Antoniou, “Digital Signal Processing”, 1st edition, Tata McGraw Hill, 2006.