COURSE CODE: **1151EE114**	COURSE TITLE: **DISCRETE TIME SIGNAL PROCESSING**	L	T	P	C
		3	**0**	**0**	**3**

COURSE CATEGORY:

Program Core

PREAMBLE :

Digital Signal Processing provides an introduction to the basic concepts of signal processing methods and to acquire knowledge of analysis of systems using various transformation techniques. It provides students to realize about different filter structure and also to develop algorithm for signal processing.

PREREQUISITE COURSES:

Signals and Systems

RELATED COURSES:

Advanced Digital Signal Processing

COURSE EDUCATIONAL OBJECTIVES :

The objectives of the course are to make the students,

- Learn discrete Fourier transform and its properties
- Study the characteristics of IIR to design the IIR filter
- Design FIR Filter to filter the undesired signals.
- Understand Finite word length effects & DSP Processor.
- Study the concept of Multirate Signal processing & its Applications

COURSE OUTCOMES :

Upon the successful completion of the course, students will be able to:

CO Nos.	Course Outcomes	Knowledge Level (Based on revised Bloom's Taxonomy)
CO1	Apply Discrete Fourier Transform & Fast Fourier Transform for the given signals.	K3
CO2	Design the Digital Infinite Impulse Response Filters (IIR) from given Specifications	K3
CO3	Design the Digital Infinite Impulse Response Filters (FIR) from given Specifications	K3
CO4	Analyze the finite word length effect on filters.	K3
	Apply the basic signal processing concepts in DSP Processor	K3
CO5	Explain the basics of Multirate Signal Processing concepts & its Applications.	K2

CORRELATION OF COs AND POs

COs	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12
CO1	H	M	L		H							
CO2	H	M	L		H							M
CO3	H	L	L		H							M
CO4	L	M	L		M							

CO5	M	L										M

COURSE CONTENT:

UNIT I	**DISCRETE FOURIER TRANSFORMS**	**9**

Introduction & Properties of DFT – Linear & Circular Convolution Methods, FFT algorithms – Radix-2 FFT algorithms – Decimation in Time – Decimation in Frequency algorithms –Use of FFT algorithms in Linear Filtering and correlation

UNIT II	**IIR FILTER DESIGN**	**9**

Structures of IIR – Analog filter design – Discrete time IIR filter from analog filter – IIR filter design by using Impulse Invariance, Bilinear transformation, IIR Filter structures. Approximation of derivatives – (LPF, HPF, BPF, BRF) filter design using frequency translation.

UNIT III	**FIR FILTER DESIGN**	**9**

Structures of FIR – Linear phase FIR filter - Filter design using windowing techniques (Rectangular Window, Hamming Window, Hanning Window), Frequency sampling techniques, FIR Filter structures.

UNIT IV	**FINITE WORDLENGTH EFFECTS & DSP PROCESSOR**	**9**

Finite word length effects: Quantization- Truncation and Rounding errors - Quantization noise – coefficient quantization error – Product quantization error - Overflow error – limit cycle oscillations, scaling. Introduction to DSP architecture – Harvard architecture - Dedicated MAC unit - Multiple ALUs, Advanced addressing modes, Pipelining, Overview of instruction set of TMS320C5X and C54X.

UNIT V	**MULTIRATE SIGNAL PROCESSING & APPLICATIONS**	**9**

Multirate signal processing: Decimation, Interpolation, Sampling rate conversion by a rational factor –Application-Sub band coding, Musical Sound Processing, Digital Audio sampling rate conversion, Oversampling A/D &D/A.

TOTAL: 45 PERIODS

TEXT BOOKS:

1. John G. Proakis & Dimitris G.Manolakis, “Digital Signal Processing – Principles, Algorithms & Applications”, 4th edition, Pearson Education / Prentice Hall, 2007.
2. B. Venkataramani, M. Bhaskar, “Digital Signal Processors: Architecture, Programming and Applications”, 2nd edition, Tata McGraw-Hill Education, 2002.

REFERENCE BOOKS:

1. S.Salivahanan, A.Vallavaraj, C Gnanapriya, “Discrete Signal Processing”, Tata McGraw-hill Publication, 2002.
2. Emmanuel C..Ifeachor, & Barrie.W.Jervis, “Digital Signal Processing”, 2nd edition, Pearson Education / Prentice Hall, 2002.
3. Sanjit K. Mitra, “Digital Signal Processing – A Computer Based Approach”, Tata Mc Graw Hill, 2007.
4. A.V.Oppenheim, R.W. Schafer and J.R. Buck, “Discrete-Time Signal Processing”, 8th Indian Reprint, Pearson, 2004.
5. Andreas Antoniou, “Digital Signal Processing”, Tata Mc Graw Hill, 2006.