GUJARAT TECHNOLOGICAL UNIVERSITY

SUBJECT NAME: Digital Signal Processing
SUBJECT CODE: 2171003
B.E. 7th SEMESTER

Type of course:
Compulsory

Prerequisite:
- Higher Engineering Mathematics, Different Transforms (Fourier, Laplace, Z-transforms)
- Signals and systems

Rationale:
The purpose of this course is to provide an understanding of Digital Signal Processing. Topics include: Introduction to digital signal processing and application, discrete time signals and systems; Analysis of LTI systems; Structures of discrete time systems; Filter designing techniques; DFT and FFT; Architecture of DSP Processors, and Multi-rate Signal Processing and applications.

Teaching and Examination Scheme:

Teaching Scheme			Credits	Examination Marks						Total Marks
L	T	P	C	Theory Marks			Practical Marks			
				ESE (E)	PA (M)		PA (V)		PA (I)	
					PA	ALA	ESE	OEP		
4	0	2	6	70	20	10	20	10	20	150

L- Lectures; T- Tutorial/Teacher Guided Student Activity; P- Practical; C- Credit; ESE- End Semester Examination; PA- Progressive Assessment; OEP-Open Ended problem; AL-Active learning;

Contents:

Sr. No.	Contents	Total Hrs	% Weight age
1	**Introduction to DSP:** Overview: Signals, systems and signal processing, classification of signals, elements of digital signal processing system, concept of frequency in continuous and discrete time signals, Periodic Sampling, Frequency domain representation of sampling, Reconstructions of band limited signals from its samples	3	05
2	**Discrete-Time Signals and Systems (Frequency Domain analysis):** Z-transform & Inverse z-transform, Linear convolution and its properties, Linear Constant Coefficient Difference equations, Frequency domain representation of Discrete-Time Signals & Systems, Representation of sequences by discrete time Fourier Transform, (DTFT), Properties of discrete time Fourier Transform, and correlation of signals, Fourier Transform Theorems.	5	10
3	**Analysis of Linear Time Invariant System:** Analysis of LTI systems in time domain and stability considerations.	8	15

	Frequency response of LTI system, System functions for systems with linear constant-coefficient Difference equations, Freq. response of rational system functions relationship between magnitude & phase, All pass systems, inverse systems, Minimum/Maximum phase systems, systems with linear phase.		
4	**Structures for Discrete Time Systems:** Block Diagram and signal flow diagram representations of Linear Constant-Coefficient Difference equations, Basic Structures of IIR Systems, lattice and lattice-ladder structures, Transposed forms, Direct and cascade form Structures for FIR Systems, Linear Phase FIR structure, Effects of Co-efficient quantization.	7	15
5	**Filter Design Techniques:** Design of Discrete-Time IIR filters from Continuous-Time filters Approximation by derivatives, Impulse invariance and Bilinear Transformation methods; Design of FIR filters by windowing techniques.	9	15
6	**Discrete-Fourier Transform & Fast Fourier Transform:** Representation of Periodic sequences: The discrete Fourier Series and its Properties Fourier Transform of Periodic Signals, Sampling the Fourier Transform, The Discrete-Fourier Transform, Properties of DFT, Linear Convolution using DFT. FFT-Efficient Computation of DFT, Goertzel Algorithm, radix2 Decimation-in-Time and Decimation-in-Frequency FFT Algorithms.	9	15
7	**Advance DSP Techniques:** **Multirate Signal Processing:** Decimation, Interpolation, Sampling rate conversion by rational factor **Adaptive filters:** Introduction**,** Basic principles of Forward Linear Predictive filter and applications such as system identification, echo cancellation, equalization of channels, and beam forming using block diagram representation study only.	7	15
8	**Architecture of DSP Processors & applications:** Harward architecture, pipelining, Multiplier-accumulator (MAC) hardware, architectures of fixed and floating point (TMSC6000) DSP processors. Applications	4	10
	Total	52	100

BookS:

1. "Digital Signal Processing: Principles, Algorithm & Application", 4th edition, Proakis, Manolakis, Pearson
2. "Discrete Time Signal Processing":Oppeheim, Schafer, Buck Pearson education publication, 2nd Edition, 2003.
3. Digital Signal Processing fundamentals and Applications,Li Tan , Jean Jiang, Academic Press,2nd edition,2013
4. Digital Signal Processing – A computer based Approach, S.K.Mitra, Tata McGraw Hill,3rd edition,2006

5. Fundamentals of digital Signal Processing –Lonnie c.Ludeman, Wiley
6. Digital Signal processing-A Practical Approach,second edition, Emmanuel I. feacher, and BarrieW..Jervis, Pearson Education
7. Digital Signal Processing, S.Salivahanan, A.Vallavaraj, C.Gnapriya TMH
8. Digital Signal Processors, Architecture, programming and applications by B. Venkatramani, M Bhaskar, Mc-Graw Hill

Suggested specification table with Marks (Theory):

Distribution of Theory Marks					
R Level	U Level	A Level	N Level	E Level	C Level
5	**15**	**15**	**15**	**10**	**10**

Legends: R: Remembrance; U: Understanding; A: Application, N: Analyze and E: Evaluate C: Create and above Levels (Revised Bloom's Taxonomy)

Note:
This specification table shall be treated as a general guideline for students and teachers. The actual distribution of marks in the question paper may vary slightly from above table.

Course Outcome:

By the end of this course, the student will be able to:

1. Formulate engineering problems in terms of DSP tasks
2. Analyse digital and analog signals and systems
3. Analyze discrete time signals in frequency domain
4. Design digital filters
5. Change sampling rate of the signal
6. Conceptualize the need of adaptive filters in communication applications.
7. Understand the key Architectural features of Digital Signal Processor
8. Apply digital signal processing algorithms to various areas

Suggested List of Suggested Experiments:

Sr. No.	**Experiment Name**
1	Write a program for Direct form – I, II form realization of the given IIR system function.
2	Write a program to plot pole-zero of a given FIR filter.
3	(A) Create Blackman Harris, Hamming and Gaussian window and plot them in the

	same filter design tool. (B) Design an FIR filter with side lobe attenuation of 40 dB using Kaiser Window of 200 points.
4	(A) Design low pass butter worth digital filter with given specification using impulse invariance method. (B) Design a high pass elliptical filter with given specification using impulse invariance method. (C) Design a band pass chebychev-2 filter with given specification using impulse invariance method.
5	Design a second-order digital bandpass Butterworth filter with the following specifications: fu= 2.6 kHz,fL = 2.4 kHz , fs = 8000 Hz. Plot the magnitude and phase response.
6	Write a program to demonstrate the time shifting and frequency shifting property of DTFT.
7	Write a program to perform circular convolution of two sequences using DFT.
8	Write a program to up sample the sinusoidal sequence by an integer factor.
9	Write a program to down sample the sinusoidal sequence by an integer factor.
10	Write a program to convert the sampling by non integer factor of a sinusoidal sequence.

Design based Problems (DP)/Open Ended Problem:

Apply Digital Signal Processing technique to any one specific area like Speech processing, Image processing, Audio processing, Bio-Medical Instrumentation, Encoding of signals, Signal Compression etc. Develop a program for the same using MATLAB/SciLab of equivalent software.

C. List of Software: MATLAB/Code Composer Studio

Learning website: www.nptel,http://ocw.mit.edu, https://cnx.org/content

ACTIVE LEARNING ASSIGNMENTS: Preparation of power-point slides, which include videos, animations, pictures, graphics for better understanding theory and practical work – The faculty will allocate chapters/ parts of chapters to groups of students so that the entire syllabus to be covered. The power-point slides should be put up on the web-site of the College/ Institute,

along with the names of the students of the group, the name of the faculty, Department and College on the first slide. The best three works should submit to GTU.