GUJARAT TECHNOLOGICAL UNIVERSITY

DIGITAL SIGNAL PROCESSING ALGORITHMS
SUBJECT CODE: 2710429
SEMESTER: I

Type of course: Digital signal processing algorithms and applications

Prerequisite: Higher Engineering Mathematics, Fundamental knowledge of signals and systems along with types, Mathematical representation of signals and system modeling in time as well as frequency domain. Transforms especially like Laplacian, Fourier and Z. Difference between basic analysis and synthesis procedure.C programming

Rationale: PG Students of EC Engineering need to possess good understanding of the fundamentals and applications of discrete-time signals and systems, including sampling, convolution, filtering, and discrete Fourier transforms. They are expected to be able to design digital filters, and perform spectral analysis on real signals using the discrete Fourier transform. They will be practiced in sampling, processing and playing back audio and other signals using MATLAB software running on PCs ,this includes both the analysis and synthesis. They will be able to design some applications on DSP kit TMS320C6713 using Code Composer Studio (CCS)

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)		
						ESE	OEP	PA	RP	
3	0	2	5	70	30	20	10	10	10	150

Content:

Sr. No.	Topics	Teaching Hrs.	Module Weightage
1	**Introduction:** Discrete time systems, Analysis of Discrete-time Linear time invariant systems, Frequency analysis of Discrete time signals and Fourier Transform properties, Frequency Response of LTI system, The Z-transforms, Inverse Z-transforms, stability, The Schur-Cohn stability test.	6	14
2	**Implementation of Discrete time systems:** Structures for FIR,IIR Systems, Quantization of Filter Co-efficients, Round-off Effects in Digital Filters	5	12
3	**Discrete Fourier Transform and its applications:** Frequency domain Sampling, Properties of DFT, Circular Convolution, Linear Filtering Methods Based on the DFT, The discrete Cosine Transform, frequency analysis of signals using the DFT	6	14
4	**Fast Fourier Transform:** Efficient computation of DFT (FFT Algorithms-DIT and DIF), Application of FFT Algorithms, The Goertzel Algorithm, The Chirp-z Algorithm, Quantization Effects in the Computation of DFT	6	14

5	**Design of Digital Filters:** Design Techniques of FIR Filters – Windowing methods, frequency sampling method ,Optimum Equiripple Linear Phase Filters, FIR differentiators,Hilbert Transformers;design techniques of IIR filters : Impulse invariance. Bilinear transformation, finite difference, frequency sampling method,Frequency Transformations	8	19
6	**Discrete Wavelet Transform** : Introduction and applications	2	5
7	**Power Spectrum Estimation:** Estimation of spectra from finite duration signals, Periodogram Nonparametric methods - Bartlett, Welch and Blackman-Tukey methods, Parametric methods – ARMA, AR and MA model based spectral estimation	7	17
8	APPLICATION OF DSP	2	5

Reference Books:

1. J.G. Proakis and D.G. Manolakis - Digital Signal Processing: Principles Algorithms and Applications,4th Edition Pearson Education, 2007
2. B.Venkataramani and M.Bhaskar, "Digital Signal Processors – Architecture,Programming and Applications" –TMH ,2003
3. S.K. Mitra - Digital Signal Processing: A computer based approach, TMH, 2001
4. A.V. Oppenheim, R.W. Schafer - Digital Signal Processing, Pearson Education, 2004
5. Thomas J. Cavicchi, Digital Signal Processing,Wiley India
6. Emmanuel Ifeachor,Barry Jervis Digital Signal Processing, 2nd Edition,Pearson Education, 2002
7. Li Tan , Digital Signal Processing fundamentals and applications,Elsevier
8. User guides Texas Instrumentation, Analog Devices, Motorola.
9. C.Sidney Burrus, Fast Fourier Transforms, Connexions, 2009 http://cnx.org/content/col10550/
10. C. Sidney Burrus, Digital Signal Processing and Digital Filter Design,Connexions, 2009 http://cnx.org/content/col10598/

Course Outcome:

1. To Analyze and implement digital signal processing systems in time domain.
2. To Compute the Fourier series and the discrete time Fourier transform (DTFT) of discrete-time
3. signals.
4. To Analyze digital signal processing systems using Z-transform and the DTFT.
5. To Design frequency-selective digital filters.
6. To Design digital filters using windows.
7. To Sample and reconstruct analog signals.
8. To Compute circular convolution and the discrete Fourier transform (DFT) of discrete-time
9. signals.
10. To Analyze and implement digital systems using the DFT and the Fast Fourier Transform (FFT).
11. To Use MATLAB for DSP system analysis and design

List of Experiments:

1. Write a MATLAB program to plot a magnitude response and phase response of a signal x (n) = a^n u (n) .a)using loopb)using MATLAB command for DFT.
2. Write a MATLAB program to find out even and odd component of a function

3. Write a MATLAB program for a convolution of two sequences x (n) and h (n). Prove that convolution in T.D. is equivalent to multiplication in F.D. and vice versa using MATLAB command
4. Study and Perform the interfacing of Code Composer Studio with DSP Processor Kit TMS X 6713
5. Write C code to generate various signals in CCS and display them on CRO through DSP Processor Kit TMS X 6713
6. Write a C program for a convolution of two sequences x (n) and h (n) using CCS and DSP Processor Kit TMS X 6713
7. Write a MATLAB program for a z – transform and inverse of z – transform
8. Write a MATLAB program to find out DFT and IDFT of given function
9. A) Write a MATLAB program to find out DFT of a signal.
   B) Write a MATLAB program to verify the circular symmetry property of DFT
10. Write a MATLAB program to find out the response of the Elliptic IIR low pass filter
11. Write a MATLAB program to find out the response of the Butterworth low pass filter
12. Write a MATLAB program to convert filter from direct form to cascade form

Open Ended Problems:

1. Write ‘C’ program and MATLAB code to generate various sequences.
2. Write ‘C’ program and MATLAB code to plot the magnitude response and phase response of a signal $x(n) = a^n u(n)$.
3. Write a ‘C’ Program to find out even and odd component of a function.
4. Write a ‘C’ program for a convolution of two sequences using ‘For loop’ and ‘While Loop’.
5. Write ‘C’ code to generate various signals in CCS and display them on CRO through DSP Processor Kit TMS 320C6713

Major Equipments : DSP Processor Kit TMS X 6713

List of Open Source Software:

Matlab, CCS

Learning website:www.nptel.ac.in