Sardar Patel Institute of Technology

Bhavan's Campus, Munshi Nagar, Andheri (West), Mumbai-400058-India
(Autonomous Institute Affiliated to University of Mumbai)

Course Code	Course Name	Teaching Scheme (Hrs/week)			Credits Assigned			
		L	T	P	L	T	P	Total
ETC602	Discrete Time Signal Processing	4	-	--	4	-	--	4
		Examination Scheme						
		ISE	MSE	ESE				
		10	30	100 (60% Weightage)				

Pre-requisite Course Codes		ETC 405: Signals and System
After successful completion of the course, student will be able to		
Course Outcomes	CO1	Able to compute various Transform Analysis of Linear Time Invariant System.
	CO2	Ability to apply engineering problem solving strategies to DSP problems.
	CO3	Ability to Design and simulate digital filters.
	CO4	Ability to Design and test signal processing algorithms for various applications.
	CO5	Ability to Recover information from signals.

Module No.	Unit No.	Topics	Ref.	Hrs.
1	**Transform Analysis of Linear Time Invariant System**		1,2	04
	1.1	Review of *Z* transform and its properties, response to sinusoidal and complex exponential signals, steady-state response to periodic input signals, response to aperiodic input signals, relationships between the system function and the frequency response function, computation of the frequency response function		
	1.2	LTI systems as frequency-selective filters like; low pass, high pass, band pass, notch, comb, all-Pass filters, and digital resonators.		
	1.3	Invertibility of LTI systems, minimum-phase, maximum-phase, mixed-phase systems		
2	**The Discrete Fourier Transform and Efficient Computation.**		1,2	12
	2.1	Frequency domain sampling and reconstruction of discrete time signals, discrete Fourier transform (DFT), DFT as a linear transformation, properties of the DFT,relationship of the DFT to other transforms		
	2.2	Fast Fourier Transform: Radix-2 and split-radix fast Fourier transform (FFT) algorithms and their applications		
	2.3	Quantization effects in the computation of the DFT		
3	**Design of Digital filters and Implementation**		1,2	12
	3.1	Design of Infinite Impulse Response (IIR) filters using impulse invariant method and bilinear transformation method, Butterworth and Chebyshev filter approximation.		
	3.2	Concepts of Finite Impulse Response (FIR) filter, symmetric and		

		anti symmetric FIR filter, FIR filter design using window method and frequency sampling method.		
	3.3	Realization structures for IIR and FIR filters using direct form structures, cascade, parallel structures, and lattice, ladder structure (only conceptual understanding)		
4	**Multi rate Signal Processing**		1,2,3	08
	4.1	Decimation by a factor *D*, interpolation by I, sampling rate conversion by a rational factor I/D		
	4.2	Polyphase filter structures, interchange of filers and down samplers/up samplers,sampling rate conversion with cascade integrator comb filters, polyphase structures for decimation and interpolation filters, structures for rational sampling rate conversion.		
	4.3	Multistage implementation of sampling rate conversion.		
	4.4	Sampling rate conversion of band pass signals		
	4.5	Sampling rate conversion by an arbitrary factor – arbitrary re-sampling with polyphase interpolators, narrow band filter structures.		
	4.6	Application of Multirate Signal Processing for design of phase shifters, interfacing of digital systems with different sampling rates, implementation of narrowband low pass filters, sub band coding of speech signals.		
5	**Analysis of Finite Word length effects**		1,2	08
	5.1	Quantization process and errors, quantization of fixed-point numbers, quantization of floating-point numbers, analysis of coefficient quantization effects		
	5.2	A/D Conversion Noise Analysis, Analysis of Arithmetic Round-Off Errors and dynamic range scaling		
6	**Applications of Digital Signal processing:**		2,5	08
	6.1	Dual –Tone multi frequency signal detection, spectral analysis of sinusoidal signals, spectral analysis of non stationary signals, and spectral analysis of random signals		
	6.2	Musical sound processing, digital music synthesis, discrete time analytic signal generation.		
	6.3	Trans-multiplexers, oversampling ADC and DAC and sparse antenna array design		
			Total	**52**

References

1. Alan V. Oppenheim and Ronald Schafer, "Discrete Time Signal Processing", Pearson Education
2. J. Proakis, D. G. Manolakis, and D. Sharma, "Digital Signal Processing: Principles, Algorithms and Applications", Pearson Education.
3. P.P. Vaidyanathan, "Multirate Systems and Filter Banks", Pearson.
4. Robert Schilling and Sandra Harris, "Fundamentals of Digital Signal Processing using MATLAB", Cengage Learning.
5. Sanjit K.Mitra, "Digital Signal Processing", McGrawHill education