Course Code	Course Name	Teaching Scheme			Credits Assigned			
		Theory	Practical	Tutorial	Theory	TW/Practical	Tutorial	Total
ETC602	Discrete Time Signal Processing	04	--	--	04	--	--	04

Course Code	Course Name	Examination Scheme							
		Theory Marks				Term Work	Practical	Oral	Total
		Internal assessment			End Sem. Exam				
		Test 1	Test 2	Ave. Of Test 1 and Test 2					
ETC602	Discrete Time Signal Processing	20	20	20	80	-	-	-	100

Course Prerequisite: ETC 405: Signals and System

Course Objectives:

- To develop a thorough understanding of the central elements of discrete time signal processing theory and the ability to apply this theory to real-world signal processing applications.
- Use z-transforms and discrete time Fourier transforms to analyze a digital system.
- Understand the discrete Fourier transform (DFT), its applications and its implementation by FFT techniques.
- Design and understand finite & infinite impulse response filters for various applications.
- The course is a prerequisite course for further studying of other multimedia related courses, such as speech processing, image processing, audio and video data compression, pattern recognition, communication systems and so forth.

Course Outcomes: Student will able to

- Formulate engineering problems in terms of DSP tasks
- Apply engineering problem solving strategies to DSP problems
- Design and test signal processing algorithms for various applications
- Recover information from signals
- Design and simulate digital filters

Module No.		Topics	Hrs.
1		**Transform Analysis of Linear Time Invariant System**	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.**	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**	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**	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**	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:**	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

Recommended Books:

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

Internal Assessment (IA):

Two tests must be conducted which should cover at least 80% of syllabus. The average marks of two tests should be considered as final IA marks

End Semester Examination:

1. Question paper will comprise of 6 questions, each of 20 marks.
2. Total 4 questions need to be solved.
3. Question No.1 will be compulsory and based on entire syllabus wherein sub questions for 2 to 5 marks will be asked.
4. Remaining questions will be selected from all the modules