

COURSE CODE	COURSE NAME	L-T-P-C	YEAR OF INTRODUCTION
EC301	Digital Signal Processing	3-1-0-4	2016
Prerequisite: EC 202 Signals & Systems			
Course objectives: <ol style="list-style-type: none"> 1. To provide an understanding of the principles, algorithms and applications of DSP 2. To study the design techniques for digital filters 3. To give an understanding of Multi-rate Signal Processing and its applications 4. To introduce the architecture of DSP processors 			
Syllabus Discrete Fourier Transform and its Properties, Linear Filtering methods based on the DFT, Frequency analysis of signals using the DFT, Computation of DFT, FFT Algorithms, IDFT computation using Radix-2 FFT Algorithms, Efficient computation of DFT of two real sequences and a 2N-Point real sequence, Design of FIR Filters, Design of linear phase FIR Filters using window methods and frequency sampling method, Design of IIR Digital Filters from Analog Filters, IIR Filter Design, Frequency Transformations, FIR Filter Structures, IIR Filter Structures, Introduction to TMS320C67xx digital signal processor, Multi-rate Digital Signal Processing, Finite word length effects in DSP systems, IIR digital filters, FFT algorithms.			
Expected outcome: The students will understand <ol style="list-style-type: none"> (i) the principle of digital signal processing and applications. (ii) the utilization of DSP to electronics engineering 			
Text Books: <ol style="list-style-type: none"> 1. Oppenheim A. V., Schafer R. W. and Buck J. R., Discrete Time Signal Processing, 3/e, Prentice Hall, 2007. 2. Proakis J. G. and Manolakis D. G., Digital Signal Processing, 4/e, Pearson Education, 2007. 			
References: <ol style="list-style-type: none"> 1. Chassaing, Rulph., DSP applications using C and the TMS320C6x DSK. Vol. 13. John Wiley & Sons, 2003. 2. Ifeachor E.C. and Jervis B. W., Digital Signal Processing: A Practical Approach, 2/e, Pearson Education, 2009. 3. Lyons, Richard G., Understanding Digital Signal Processing, 3/e. Pearson Education India, 2004. 4. Mitra S. K., Digital Signal Processing: A Computer Based Approach, 4/e McGraw Hill (India), 2014. 5. NagoorKani, Digital Signal Processing, 2e, Mc Graw –Hill Education New Delhi, 2013 6. Salivahanan, Digital Signal Processing, 3e, Mc Graw –Hill Education New Delhi, 2014 (Smart book) 7. Singh A., Srinivasan S., Digital Signal Processing: Implementation Using DSP Microprocessors, Cenage Learning, 2012. 			

Course Plan			
Module	Course content	Hours	End Sem. Exam Marks
I	The Discrete Fourier Transform: DFT as a linear transformation, Relationship of the DFT to other transforms, IDFT	2	15
	Properties of DFT and examples Circular convolution	4	
	Linear Filtering methods based on the DFT- linear convolution using circular convolution, overlap save and overlap add methods	3	
	Frequency Analysis of Signals using the DFT	2	
II	Computation of DFT: Radix-2 Decimation in Time and Decimation in Frequency FFT Algorithms	3	15
	IDFT computation using Radix-2 FFT Algorithms	2	
	Efficient computation of DFT of Two Real Sequences and a 2N-Point Real Sequence	2	
FIRST INTERNAL EXAM			
III	Design of FIR Filters- Symmetric and Anti-symmetric FIR Filters	2	15
	Design of linear phase FIR Filters using Window methods (rectangular, Hamming and Hanning) and frequency sampling Method	6	
	Comparison of Design Methods for Linear Phase FIR Filters	1	
IV	Design of IIR Digital Filters from Analog Filters (Butterworth)	4	15
	IIR Filter Design by Impulse Invariance, and Bilinear Transformation	3	
	Frequency Transformations in the Analog and Digital Domain	2	
SECOND INTERNAL EXAM			
V	Block diagram and signal flow graph representations of filters	1	20
	FIR Filter Structures: (Linear structures), Direct Form, Cascade Form and Lattice Structure	3	
	IIR Filter Structures: Direct Form, Transposed Form, Cascade Form and Parallel Form	2	
	Computational Complexity of Digital filter structures	1	
	Computer architecture for signal processing : Introduction to TMS320C67xx digital signal processor	2	
VI	Multi-rate Digital Signal Processing: Decimation and Interpolation (Time domain and Frequency Domain Interpretation without proof)	3	20
	Finite word length effects in DSP systems: Introduction (analysis not required), fixed-point and floating-point DSP arithmetic, ADC quantization noise	2	

	Finite word length effects in IIR digital filters: coefficient quantization errors	2	
	Finite word length effects in FFT algorithms: Round off errors	2	
END SEMESTER EXAM			

Question Paper Pattern (End Sem Exam)

Maximum Marks: 100

Time : 3 hours

The question paper shall consist of three parts. Part A covers modules I and II, Part B covers modules III and IV, and Part C covers modules V and VI. Each part has three questions uniformly covering the two modules and each question can have maximum four subdivisions. In each part, any two questions are to be answered. Mark patterns are as per the syllabus with 40 % for theory and 60% for logical/numerical problems, derivation and proof.

