Course Number and Name
BEC505 - DIGITAL SIGNAL PROCESSING
Credits and Contact Hours
4 and 60
Course Coordinator's Name
Dr B.Karthik
Text Books and References
TEXTBOOK: 1 .JohnG.Proakis&DimitrisG.Manolakis,"DigitalSignalProcessing–Principles, 2. Algorithms&Applications",FourthEdition,PearsonEducation/Prentice Hall,2007. **REFERENCES:** 1. Sanjit K.Mitra,"Digital Signal Processing–A Computer Based Approach",Tata McGraw Hill, 2. A.V.Oppenheim, R.W. Schafer andJ.R. Buck, "Discrete-Time Signal Processing", 8th Indian Reprint,Pearson,2004. 3. www. ocw.mit.edu
Course Description
• To study about discrete time systems and to learn about FFT algorithms. • To study the design techniques for FIR and IIR digital filters • To study the finite word length effects in signal processing • To study the properties of random signal,Multirate digital signal processing and about QMF filters.

Prerequisites	**Co-requisites**
Signals and Systems	Nil

required, elective, or selected elective (as per Table 5-1)
required
Course Outcomes (COs)
CO1 To apply DFT for the analysis of digital signals & systems
CO2 To design FIR filters
CO3 To design IIR filters
CO4 To characterize finite Word length effect on filters
CO5 To have a deep understanding on basics of digital signal processing which can be applied to communication systems
CO6 To design the Multirate Filters
Student Outcomes (SOs) from Criterion 3 covered by this Course

COs/SOs	a	b	c	d	e	f	g	h	i	j	k
CO1	H			H	H						
CO2	H	H	H	H	H				M		
CO3	H	H	H	H	H				M		
CO4		M		M		H			M		
CO5	M									M	
CO6			H			L					

List of Topics Covered
UNIT I DISCRETE – TIME SIGNALS AND SYSTEMS : 12 Sampling of Analogue signals – aliasing – standard discrete time signals – classification – discrete time systems – Linear time invariant stable casual discrete time systems – classification methods – linear and circular convolution – Overlap add and Save methods-Difference equation representation – DFS, DTFT, DFT – FFT computations using DIT and DIF algorithms. **UNIT II INFINITE IMPULSE RESPONSE DIGITAL FILTERS: 12** Review of design of analogue Butterworth and Chebyshev Filters, Frequency transformation in analogue domain – Design of IIR digital filters using impulse invariance technique – Design of digital filters using bilinear transform – pre warping – Frequency transformation in digital domain – Realization using direct, cascade and parallel forms. **UNIT III FINITE IMPULSE RESPONSE DIGITAL FILTERS: 12** Symmetric and Antisymmetric FIR filters – Linear phase FIR filters – Design using Frequency sampling technique – Window design using Hamming, Hanning and Blackmann Windows – Concept of optimum equiripple approximation – Realisation of FIR filters – Transversal, Linear phase and Polyphase realization structures. **UNIT IV FINITE WORD LENGTH EFFECTS: 12** Quantization noise – derivation for quantization noise power – Fixed point and binary floating point number representations – Comparison – Overflow error – truncation error – coefficient quantization error – limit cycle oscillations- signal scaling – analytical model of sample and hold operations. **UNIT V SPECIAL TOPICS IN DSP: 12** Discrete Random Signals- Mean, Variance, Co-variance and PSD – Periodiogram Computation – Principle of Multi rate DSP – decimation and Interpolation by integer factors – Time and frequency domain descriptions – Single, Multi stage, polyphase structures – QMF filters – Subband Coding