ADVANCED DIGITAL SIGNAL PROCESSING

Course Code: 13EC2102 **L P C**
4 0 3

Course Outcomes:

At the end of the course the student will be able to

CO1: Comprehend the DFTs and FFTs.

CO2: Design and Analyze the digital filters

CO3: Acquire the basics of multi rate digital signal processing

CO4: Analyze the power spectrum estimation

CO5: Comprehend the Finite word length effects in Fixed point DSP Systems

UNIT-I
DISCRETE AND FAST FOURIER TRANSFORMS:
Properties of DFT, Linear Filtering methods based on the DFT, Overlap-save, Overlap -Add methods, frequency analysis of signals, Radix-2 FFT and Split- Radix FFT algorithms The Goertzel and Chirp Z transform algorithms.

UNIT-II
DESIGN OF IIR AND FIR FILTERS:
Design of IIR filters using Butterworth &Chebyshev approximations, frequency transformation techniques, structures for IIR systems – cascade, parallel, lattice & lattice-ladder structures, Fourier series method, Windowing techniques, design of digital filters based on least – squares method, pade approximations, least squares design, wiener filter methods, structures for FIR systems –cascade, parallel, lattice & lattice-ladder structures.

UNIT-III
MULTI RATE SIGNAL PROCESSING:
Decimation by a factor D, Interpolation by a factor I, Sampling rate conversion by a rational factor I/D, Filter design & Implementation for sampling rate conversion, filter bands, sub band coding, polyphase filters.

UNIT-IV

POWER SPECTRAL ESTIMATION:

Estimation of spectra from finite duration observation of signals, Non-parametric methods: Bartlett, Welch &Blackman&Tukey methods. Relation between auto correlation & model parameters, Yule-Waker& Burg Methods, MA & ARMA models for power spectrum estimation.

UNIT-V

ANALYSIS OF FINITE WORD LENGTH EFFECTS IN FIXED-POINT DSP SYSTEMS:

Fixed, Floating Point Arithmetic – ADC quantization noise & signal quality – Finite word length effect in IIR digital Filters – Finite word-length effects in FFT algorithms.

TEXTBOOKS:

[1] J.G.Proakis&D.G.Manolokis, "*Digital Signal Processing – Principles, Algorithms Applications*", PHI.

[2] Alan V Oppenheim & Ronald W Schaffer, "*Discrete Time signal processing*", PHI.

REFERENCE BOOKS:

[1] S. M .Kay, "*Modern spectral Estimation techniques*", PHI, 1997.Emmanuel C. Ifeacher Barrie. W. Jervis, "DSP – A Practical Approach", Pearson Education.