DIGITAL SIGNAL PROCESSING

Paper Code: ETIT-415 | **L** 3 | **T/P** 0 | **C** 3
Paper: Digital Signal Processing

	L	T/P	C
Paper Code: ETIT-415	**L**	**T/P**	**C**
Paper: Digital Signal Processing	**3**	**0**	**3**

INSTRUCTIONS TO PAPER SETTERS: **MAXIMUM MARKS: 75**

1. Question No. 1 should be compulsory and cover the entire syllabus. This question should have objective or short answer type questions. It should be of 25 marks.
2. Apart from Question No. 1, rest of the paper shall consist of four units as per the syllabus. Every unit should have two questions. However, student may be asked to attempt only 1 question from each unit. Each question should be of 12.5 marks.

Objectives: *The aim of this course is to provide in depth knowledge of various digital signal processing techniques and design of digital filters, learn the concept of DFT FFT algorithms, and design of digital filters using different approximations, DSP processor and architecture. The prerequisites of this subject are basic knowledge of signal and systems.*

UNIT–I :
Frequency Domain Sampling: The Discrete Fourier Transform, Properties of the DFT, Linear filtering methods based of the DFT.
Efficient computation of the DFT: Principal Of FFT, Fast Fourier Transform Algorithms, Applications of FFT Algorithms, A linear filtering approach to computation of the DFT.
Application of DFT, Design of Notch filter

[T2,T1][No. of Hours: 11]

UNIT–II:
Design & Structure of IIR filters from analog filters: Impulse Invariance; Bilinear transformation and its use in design of Butterworth and Chebyshev IIR Filters; Frequency transformation in Digital Domain, Direct, Cascade, Parallel & transposed structure
Design & structure of FIR filters: Symmetric and anti-symmetric FIR filters; Design of Linear Phase FIR filters using windows, Frequency Sampling Method of FIR design, Direct, Cascade, Frequency Sampling, transposed structure

[T1,T2] [No. of Hours: 11]

UNIT–III:
Implementation of Discrete Time Systems:
Lattice structures, Lattice and Lattice-Ladder Structures, Schur - Cohn stability Test for IIR filters; Discrete Hilbert Transform.
Linear predictive Coding:
Lattice filter design, Levension Darwin Technique, Schur Algorithm

[T1,T2] [No. of Hours: 10]

UNIT–IV:
Quantization Errors in Digital Signal Processing: Representation of numbers, Quantization of filter coefficients, Round-off Effects in digital filters.
Multirate Digital Signal Processing: Decimation, Interpolation, Sampling rate conversion by a rational factor; Frequency domain characterization of Interpolator and Decimator; Polyphase decomposition.

[T1, T2][No. of Hours: 10]

Text Books:

[T1] Oppenheim & Schafer, Digital Signal Processing, PHI-latest edition.
[T2] Proakis and Manolakis, Digital Signal Processing, PHI Publication

Reference Books:

[R1] S. K. Mitra, Digital Signal Processing, TMH edition 2006
[R2] Johny. R. Johnson, Introduction to Digital Signal Processing, PHI-latest edition
[R3] R.Babu ,Digital Signal Processing , SciTech Publication.