IT8071 DIGITAL SIGNAL PROCESSING

DETAILED SYLLABUS

OBJECTIVES:

- To understand the basics of discrete time signals, systems and their classifications.
- To analyze the discrete time signals in both time and frequency domain.
- To design lowpass digital IIR filters according to predefined specifications based on analog filter theory and analog-to-digital filter transformation.
- To design Linear phase digital FIR filters using fourier method, window technique ·
- To realize the concept and usage of DSP in various engineering fields.

UNIT I DISCRETE TIME SIGNALS AND SYSTEMS

Introduction to DSP – Basic elements of DSP– Sampling of Continuous time signals– Representation, Operation and Classification of Discrete Time Signal–Classification of Discrete Time Systems– Discrete Convolution: Linear and Circular–Correlation.

UNIT II ANALYSIS OF LTI DISCRETE TIME SIGNALS AND SYSTEMS

Analysis of LTI Discrete Time Systems using DFT–Properties of DFT–Inverse DFT– Analysis of LTI Discrete Time Systems using FFT Algorithms– Inverse DFT using FFT Algorithm.

UNIT III INFINITE IMPULSE RESPONSE FILTERS

Frequency response of Analog and Digital IIR filters–Realization of IIR filter–Design of analog low pass filter–Analog to Digital filter Transformation using Bilinear Transformation and Impulse Invariant method–Design of digital IIR filters (LPF, HPF, BPF, and BRF) using various transformation techniques.

UNIT IV FINITE IMPULSE RESPONSE FILTERS

Linear Phase FIR filter–Phase delay–Group delay–Realization of FIR filter–Design of Causal and Non-causal FIR filters (LPF, HPF, BPF and BRF) using Window method (Rectangular, Hamming window, Hanning window) –Frequency Sampling Technique.

UNIT V APPLICATIONS OF DSP

Multirate Signal Processing: Decimation, Interpolation, Spectrum of the sampled signal – Processing of Audio and Radar signal.

OUTCOMES:

At the end of the course, the students should be able to: ·

- Perform mathematical operations on signals.
- Understand the sampling theorem and perform sampling on continuous-time signals to get discrete time signal by applying advanced knowledge of the sampling theory.

- Transform the time domain signal into frequency domain signal and vice-versa. ·
- Apply the relevant theoretical knowledge to design the digital IIR/FIR filters for the given analog specifications.

TEXT BOOK:

1. John G. Proakis & Dimitris G.Manolakis, ―Digital Signal Processing – Principles, Algorithms & Applications‖, Fourth Edition, Pearson Education / Prentice Hall, 2007.

REFERENCES

1. Richard G. Lyons, ―Understanding Digital Signal Processing‖. Second Edition, Pearson Education.

2. 2. A.V.Oppenheim, R.W. Schafer and J.R. Buck, ―Discrete-Time Signal Processing‖, 8th Indian Reprint, Pearson, 2004.

3. 3. Emmanuel C.Ifeachor, & Barrie.W.Jervis, ―Digital Signal Processing‖, Second Edition, Pearson Education / Prentice Hall, 2002.

4. 4. William D. Stanley, ―Digital Signal Processing‖, Second Edition, Reston Publications.