PRINCIPLES OF DIGITAL SIGNAL PROCESSING
(ELECTIVE IV)

Course Code :13IT1112

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Pre requisites: Mathematics-I, Mathematics-II

Course Educational Objectives:

To give an understanding on the study that deals with the representation of signals as ordered sequences of numbers and how to process those ordered sequences.

- ✤ To understand the basics of signals and system by analyzing the various transformations available and determine their use to DSP.
- ✤ To study on the various digital filtering techniques and how to apply to DSP.
- ✤ To study on the ways to estimate signal parameters, and transform a signal into a form that is more informative.
- ✤ To give students a flavor on the applications of DSP in the areas of speech and image
- ✤ To understand Speech compression.

Course Outcomes:

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

- ✤ Learn about filtering methods based on DFT.
- ✤ Learn about structures of IIR filter design.
- ✤ Learn about Bilinear transformation.
- ✤ Learn about fir filter design.
- ✤ Learn about different applications in signal processing.

UNIT-I (12 Lectures)

SIGNALS AND SYSTEMS :

Basic elements of DSP – concepts of frequency in Analog and Digital Signals -sampling theorem – Discrete – time signals, systems – Analysis of discrete time LTI systems – Z transform – Convolution (linear and circular) – Correlation.

UNIT-II (12 Lectures)

FREQUENCY TRANSFORMATIONS:

Introduction to DFT – Properties of DFT – Filtering methods based on DFT – FFT Algorithms - Decimation – in – time Algorithms, Decimation – in – frequency Algorithms – Use of FFT in Linear Filtering – DCT.

UNIT-III (12 Lectures)

IIR FILTER DESIGN:

Structures of IIR – Analog filter design – Discrete time IIR filter from analog filter – IIR filter design by Impulse Invariance, Bilinear transformation, Approximation of derivatives – (HPF, BPF, BRF) filter design using frequency translation

UNIT-IV (12 Lectures)

FIR FILTER DESIGN:

Structures of FIR – Linear phase FIR filter – Filter design using windowing techniques, Frequency sampling techniques – Finite word length effects in digital Filters

UNIT-V (12 Lectures)

APPLICATIONS:

Multi rate signal processing – Speech compression – Adaptive filter – Musical sound processing – Image enhancement.

TEXT BOOKS:

1. John G. Proakis & Dimitris G.Manolakis, *"Digital Signal Processing – Principles, Algorithms & Applications"*, 4th Edition, Pearson education / Prentice Hall, 2007.

2. Emmanuel C..Ifeachor, & Barrie.W.Jervis, *"Digital Signal Processing"*,2nd Edition, Pearson Education / Prentice Hall, 2002.

REFERENCES:

1. Sanjit K. Mitra, *"Digital Signal Processing – A Computer Based Approach"*, Tata McGraw Hill, 4th Edition, 2007.
2. Alan V.Oppenheim, Ronald W. Jchafer & Hohn. R.Back, *"Discrete Time Signal Processing"*, 2nd Edition, Pearson Education, 2001.
3. Andreas Antoniou, *"Digital Signal Processing"*, 2nd Edition, Tata McGraw Hill, 2009.