Digital Signal Processing

Syllabus

(2nd Jan 2019 to 3oth April 2019)

****

<u>UNIT-I</u> Review of signals and math behind DSP

- Vector spaces and subspaces
- Orthogonal and orthonormal bases
- Signal as a vector
- Classification of signals and systems
- Linear Time Invariant system analysis with Linear Constant coefficient Differential Equation (LCCDE)
- Frequency domain analysis of signals and systems.

<u>UNIT-II</u> Discrete Time signal Analysis

- Sampling Theorem
- Time domain analysis of signals
- Discrete time LTI system analysis with Linear Constant coefficient Difference Equation (LCCDE)

<u>UNIT-III</u> Frequency Domain Analysis of Discrete time signals

- Discrete Fourier Series
- Discrete Time Fourier Transform
- Discrete Fourier Transform
- Fast Fourier Transform

<u>UNIT-IV</u> Discrete time system Analysis

- Z-Transform
- Discrete LTI system analysis using Z-Transform and system function

<u>UNIT-V</u> Analog Filter Design

- Simple filters
- Butterworth, Chebyshev and elliptic analog filter design

<u>UNIT-VI</u> Digital Filter Design

- Digital IIR filter design from analog filters
- Digital FIR filter design
- Filter realizations
- Applications of filters for analysis of natural signal like speech, Image, ECG, EEG etc.

<u>References:</u>

- Proakis, John G. *Digital signal processing: principles algorithms and applications*. Pearson Education India, 2001.
- Mitra, Sanjit Kumar, and Yonghong Kuo. *Digital signal processing: a computer-based approach*. Vol. 2. New York: McGraw-Hill, 2006.
- Lyons, Richard G. *Understanding Digital Signal Processing, 3/E*. Pearson Education India, 2011.
- Smith, Steven. *Digital signal processing: a practical guide for engineers and scientists*. Elsevier, 2013.