**DSP ALGORITHMS and SOFTWARE EEE 509**

**INTERNET CLASS -**

**Course Description**: This is a three credit course that is divided in three modules. The course starts with a review of the basics of signals and systems and continues with an introduction to discrete-time (digital) systems. The modules are described below:

**List of Contents**

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| **Module 1**   * Introduction to DSP * Review of analog signals and systems * The sampling theorem * Discrete-time (digital) filters * Impulse Response * FIR and IIR digital filters * Difference equations and convolution * Transient and steady state response of digital filters * Frequency response of digital filters * The z transform and its properties * The transfer function * Poles and zeros of the transfer function * Inverse z transform with the Residue Theorem * Deconvolution and System I.D. * Quantization Simulations * Introduction to MATLAB programming for DSP with programming examples * MATLAB Speech Processing Exercise * Frame by Frame Processing using MATLAB * Special Filters and their MATLAB implementation | **Module 2**   * FIR filter design * Linear phase filters * Design using windows * The Kaiser method * MATLAB Code Examples of the above * IIR Filter Design * Analog Filter Approximations and Impulse Invariance * The Bilinear Transformation * Butterworth filter design * Checychev Filter Design * Elliptic Filter Design * The DFT and the FFT * DFT Properties * DFT Computation * The FFT and its applications * Spectral Estimation using the FFT * The Cepstrum * Compression Applications * Speech Enhancement * MATLAB & J-DSP code examples of the FFT * Sampling Rate Changes * QMF Filter Bank Design * Pseudo QMF and MPEG * Lapped Transforms * Image Processing Filters and Transforms | **Module 3**   * Random Signal Processing * Stationary and Ergodic Signals * The mean, the variance, and the autocorrelation * Cross-correlations * The power spectrum * Random signal processing with digital filters * System Identification using the cross correlation and deconvolution * Application to channel estimation * Adaptive Filters * Adaptive noise and echo cancellation * Linear Prediction * Yule-Walker Equations * Introduction to speech processing applications * Vocoders and cellular telephones * Audio coding and applications to computer (internet) music * CELP Code in MATLAB * MATLAB code and Psychoacoustics in MPEG * MATLAB & J-DSP code examples of random signal processing and adaptive filters * Image Processing MATLAB Simulations |

**Books:**1. Digital Signal Processing, A Computer-Based Approach, Sanjit K. Mitra, 2011 4th Edition, McGraw-Hill   
2. Discrete-time Signal Processing, Oppenheim and Schafer, 3rd Edition , 2009, Prentice Hall (Pearson)  
3. A. Spanias, Digital Signal Processing; An Interactive Approach – **2nd Edition**, 403 pages, Textbook with JAVA exercises, ISBN 978-1-4675-9892-7,Lulu Press On-demand Publishers Morrisville, NC, May 2014.

**Pre-requisites**Signals and Systems (EEE 203 or similar)  
Pre-requisite by topic: Theory of simple analog linear circuits, the Laplace transform, the continuous Fourier transform, discrete time linear systems, The Fourier transform.