**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
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**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.