Course Description and Objective:

Introduction to Digital Signal Processing. Sampling Theorem, Discrete-time Fourier transform, power spectrum, discrete Fourier transform and the FFT algorithm, z-Transform, digital filter design and implementation.

OUTLINE

- Basic Signal and Systems
  - 1-D Signals and Filters
  - Random Signals
  - Multidimensional Signals
- Convolution
  - FIR Filters
  - Correlation
- Discrete-Time Fourier Transform (DTFT)
  - Transform Pairs
  - Properties
  - Relation to Continuous-Time Fourier Transform
  - Sampling Theorem
  - Power Spectrum
- Discrete Fourier Transform (DFT)
  - Transform Pairs and Properties
  - Computation via the FFT Algorithm
  - Circular Convolution
  - High-Speed Convolution
- Bilateral z-Transform
  - Transform Pairs and Properties
  - Relationship to DTFT
  - Partial Fraction Expansion
  - Difference Equations and IIR Filters
  - Structures for FIR and IIR Filters
  - Linear Phase
  - All-pass Filters
  - Minimum Phase
  - Spectral Factorization
- Digital Filter Design
  - Bilinear Transformation
  - Frequency Selective Design
  - Butterworth, Chebyshev, Elliptic Filters
  - Windowing
  - Chebyshev Approximation
  - Least-Squares
  - Wiener Filter (SNR minimization)