EE 453: Fundamentals of Digital Signal Processing, Fall 2010

Objective: To introduce the fundamentals of digital signal processing. The emphasis will be on analysis tools, the design of digital filters, and on the computation of the Discrete Fourier Transform (DFT). The theory developed in class will be complemented by MATLAB simulations and a real-time DSP laboratory.

Text: *Digital Signal Processing – A Computer-based Approach*, author Sanjit Mitra

Reference: *Signal Processing & Linear Systems,* B. P. Lathi (EE 350/351 textbook)

Prerequisite: EE 351 or EE 317/353 or an equivalent Signals/Systems introductory course. **This course assumes prior knowledge of the z-transform.**

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Office Hours: TBD

Grading: Homework and MATLAB Problems   20 %
Laboratory Experiments (3-4)   15 %
Midterms (2)   40 %
Final Exam   25 %

Midterm #1: Tuesday October 9 6:30 - 7:45 PM (room TBD)
Midterm #2: Thursday November 15 6:30 - 7:45 PM (room TBD)
Final: Time/location to be announced by the registrar's office.

Exams
Exams will be closed book and closed notes. You will be allowed to bring 1 sheet of self-prepared notes (8 1/2 X 11) to each exam, however.

Lab Experiments
There will be 3-4 lab experiments held in room 204 EEW throughout the semester.

Homework
Homework will be assigned regularly and will be due at the **beginning** of class on the due date. Most homework assignments will include computer problems, to be done using MATLAB. When submitting the homework, write only on 1 side of the paper, write your name and SS# on every sheet in the upper right corner, and STAPLE all pages together. Clearly mark your answers by underlying or enclosing them in a box. You may discuss your homework with your fellow students, but the work you submit must be your own. **Blatant homework copying will not be tolerated.** Homework solutions will be posted on the web.

To be fair to other students, **late HW will not generally be accepted.** See the instructor if you have extenuating circumstances which warrant an exception.

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COURSE OUTLINE

1. **Brief** review of discrete-time signals and systems topics (Ch 1, 2.1-2.8, Ch. 3, 4.1-4.2 ,4.6-4.11, Ch. 6, 7.9)
   overview of DSP -- discrete-time sinusoids – discrete time systems -- convolution -- difference equations – z transforms – DTFT -- sampling/reconstruction
   Laboratory #1: Sampling/reconstruction of continuous-time signals

2. Digital filter design basics (7.1-7.2, 7.4)
   ideal and non-ideal filters – pole/zero filter design – notch and comb filters

3. IIR digital filter design (4.4-4.5, 9.1-9.6)
   analog Butterworth/Chebyshev filters – filter design via bilinear transformation – filter design via impulse invariance – other filter design methods – frequency transformations
   Laboratory #2: Design and implementation of IIR filters

4. FIR digital filter design (7.3, 10.1-10.2, 10.5.1, 10.5.4)
   FIR filter design via windowing/truncation
   Laboratory #3: Design and implementation of FIR filters

5. Computer-aided digital filter design (9.7, 10.3, 10.5.2, 5.8)
   least squares – equi-ripple filter design – filter design via frequency sampling

6. Filter Realization and Error analysis (8.1-8.5, 8.12, 11.8-11.10, 12.1-12.7, 12.11)
   filter realizations – binary representation of numbers – quantization – effects of input quantization – effects of coefficient quantization – multiplication roundoff errors – limit cycle oscillations

7. Discrete Fourier Transform and the FFT (5.2-5.7, 5.10, 11.3)
   DFT definition and properties – Fourier Transform estimation via the DFT – DFT-based filtering – efficient computation of the DFT via the FFT
   Laboratory #4 (tentative): The Discrete Fourier Transform

8. Select advanced topics (4.3, 13.1-13.3)
   bandpass sampling – oversampling A/D converters – multirate DSP – adaptive digital filters