

EE-330

DIGITAL SIGNAL PROCESSING

DEPARTMENT	Electrical and Computer Engineering
COORDINATOR	Cynthia J. Finelli, Associate Professor of Electrical Engineering
CATALOG DESCRIPTION	Basic principles, design, and applications of digital signal processing systems are presented. Topics include: review of discrete-time signals and systems, the z-transform, discrete-time Fourier analysis, the Discrete Fourier Transform, the Fast Fourier Transform, digital filter structures, FIR filters, and IIR filters. The course includes extensive use of MATLAB and experimental design projects using real-time digital signal processors.
PREREQUISITES	EE-230, Signals and Systems
CLASS/LAB SCHEDULE	Three 60-minute class periods and one 120-minute laboratory session per week. (3-2-5)
TEXTBOOK	Vinay K. Ingle and John G. Proakis, <i>Digital Signal Processing Using MATLAB</i> . Brooks/Cole Publishing Company, Pacific Grove, CA, 2000 (or instructor's choice).
REFERENCE	<i>The MATLAB Student Version: Learning MATLAB Version 5.3</i> . The MathWorks, Inc., 1999.
CREDITS	Engineering science: 2 credits; Engineering design: 2 credits

RELATIONSHIP TO PROGRAM EDUCATIONAL OBJECTIVES

EE-330 contributes to students' achievement of the Electrical Engineering Program Educational Objectives (PEOs) as outlined below:

PEO1: Students learn a variety of digital signal processing concepts and complete real-world laboratory projects, thereby becoming better prepared for employment in their field.

PEO2: Students work in teams to complete the laboratory portion of the course, thereby improving their interpersonal skills.

PEO3: Students acquire self-confidence when completing the open-ended projects of the course.

PEO4: Students receive a solid foundation in digital signal processing and they design and implement digital signal processing systems, thereby preparing them for lifelong education.

COURSE LEARNING OBJECTIVES

Each student who receives credit for EE-332 will have demonstrated the ability to do all of the tasks listed below.

1. Determine the output of discrete-time system represented as difference equation, impulse response $h(n)$, transfer function $H(z)$, or frequency response $H(e^{j\omega})$. Determine whether or not the system is causal and/or stable.
2. Given one system representation (i.e, a difference equation, impulse response $h(n)$, transfer function $H(z)$, or frequency response $H(e^{j\omega})$), convert to any of the other representations.
3. Conduct spectral analysis using Fourier methods.
4. Derive the 4-point decimation-in-time and decimation-in-frequency FFT algorithms.
5. Sketch the direct form I and direct form II filter implementation of a system.
6. Discuss multiple design techniques of FIR and IIR filters and compare and contrast their characteristics.
7. From absolute or relative filter specifications, design an arbitrary FIR or IIR filter.
8. Implement a digital filter using a real time DSP microprocessor.

TOPICS

1. Discrete-time signals and systems—Basic signals and operations, discrete-time systems and their properties, the convolution sum and graphical convolution, time-domain solution to difference equations (3 classes)
2. z-transform—Definition, pairs and properties, the region of convergence, pole-zero diagrams and implications for stability and causality, computation of the inverse z-transform via partial fraction expansion (for unique and repeated roots, real and complex roots, and improper functions), the unilateral z-transform and its use to solve difference equations (3 classes)
3. Discrete-time Fourier analysis—the DTFT for aperiodic and periodic signals, Parseval's relationship, sampling and the Nyquist sampling theorem, analog signal reconstruction (3 classes)
4. Discrete Fourier Transform—Computing the DFT and inverse DFT, zero-padding, circular convolution, the Fast Fourier Transform (6 classes)
5. Digital filter structures—Cascade form, parallel form, direct-form I, and direct-form II filter implementation (2 classes)
6. FIR filters—filter specifications, the frequency sampling method, Gibb's phenomenon, window design, optimal equiripple technique (6 classes)
7. IIR filters—lowpass analog filters (Butterworth, Chebyshev type I and II, Elliptical), filter digitization via bilinear transform or impulse invariance, frequency band transformation (5 classes)
8. Tests (2 classes)

LABORATORY EXERCISES

1. Audio representation of digital signals.
2. Digital recording of audio signals.
3. Record and filter an electrocardiogram.
4. Predict machinery bearing wear.
5. Simulate room acoustics.
6. Final course project of instructor's choice.