ENG EC416 Introduction to Digital Signal Processing

2008-2009 Catalog Data:
Prereq: ENG EC 401. Introduces techniques of digital signal processing and application to deterministic as well as random signals. Topics include representation of discrete-time random signals, A/D conversion, D/A conversion, frequency domain and z-domain analysis of discrete-time signals and systems, discrete-time feedback systems, difference equation and FFT based realizations of digital filters, design of IIR Butterworth filters, window-based FIR filter design, digital filtering of random signals, FFT-based power spectrum analysis. Includes lab. 4 cr.

Status in the Curriculum: Elective

Class/Lab Schedule:
LEC: 4 hrs/wk

Textbooks and other required materials:
None

Reference:
"Discrete-time Signal Processing," by Oppenheim, Schafer, Buck (Prentice Hall)
"Signals & Systems," by Oppenheim, Willsky, Nawab (Prentice Hall)

Coordinator:
S. Hamid Nawab, Professor, ECE Department

Prerequisites by topic:
ENG EC401: Signals & Systems
Basic knowledge of MATLAB

Goals:
To provide students with:
- Thorough understanding of the theory of A/D and D/A signal conversion, digital filtering and spectral analysis.
- Extensive experience in the design and implementation of digital filters and spectral analyzers, and in their application to real signals (e.g., speech, images).
- Detailed knowledge of a prototypical microprocessor-based framework for real-time digital filtering of analog signals.

Course Outcomes:
As an outcome of completing this course, students should be able to:
1) *Understand* how the combination of A/D conversion, digital filtering, and D/A conversion may be used to filter analog signals within a real-time microprocessor-based framework.
2) *Understand* the time- and frequency-domain concepts related to A/D conversion.
3) *Understand* the time- and frequency-domain concepts related to D/A conversion.

4) *Understand* the roles of oversampling, digital downsampling, and digital upsampling in the digital filtering of analog signals.
5) *Understand* the respective roles of the magnitude and phase of the frequency response of a digital filter.
6) *Understand* the concepts of phase delay and group delay of a digital filter.
7) *Understand* the relations between the DTFT, the DFT, and the FFT.
8) *Understand* the computational issues in the implementation of digital filters.
9) *Understand* the notion of random signals as an aid to filter design.
10) *Design* FIR filters using the Kaiser Windowing Method.
11) *Design* IIR Filters using the Bilinear Transformation Method.
12) *Implement and test* FIR filters.
13) *Implement and test* IIR filters.
14) *Implement* and test the periodogram.
15) *Implement and test* FFT-based frequency response computation for FIR filters.
16) *Implement and test* FFT-based frequency response computation for stable IIR filters.
17) *Discover* practical DSP applications through the use of Internet and other resources.
18) *Discover* the differences between the types of commercially available DSP chips.
19) *Write reports* on filter design projects
20) *Assess the societal impact* of DSP and related technologies *and the engineer's responsibilities* in this regard.

Course Outcomes mapped to Program Outcomes:

Program:	**a**	**b**	c	d	e	f	g	h	i	j	k
Course:	1-9	12-16	10-11		1-9	20	19	20	17	20	12-16
Emphasis:	5	3	4		5	2	3	2	3	2	3

1=not at all; 5=a great deal;

Contribution of Course to Meeting the Professional Component:
Engineering topics: 100%

Prepared by: S. Hamid Nawab, Prof. **Date:** Jan. 1, 2009