EE 438 DIGITAL SIGNAL PROCESSING WITH APPLICATIONS

Sem. 1 and 2. Class 3, Lab. 3, cr. 4

***Prerequisite*:** EE 301 and 302.

Engineering Science: 1.5 credits

Engineering Design: 2.5 credits

Course Description:

The course is composed of three topical units designed to support laboratory experiments and a design project. The first lecture unit, Foundations, is followed by two units chosen from a set of three possible topics. The choice and order of these second two units varies depending on the design project. For example, if the design project topic is on speech recognition, then the Speech Processing unit is covered first so the the maximum time is available for projects. *Foundations:* the review of continuous-time and discrete-time signals, and spectral analysis; design of finite impulse response and infinite impulse response digital filters. *Image processing:* two dimensional signals, systems, and spectral analysis; image enhancement; image coding; and image reconstruction. *Speech processing:* vocal tract models and characteristics of the speech waveform; short-time spectral analysis and synthesis; linear predictive coding. *Array processing:* basic radar principles; representation of propagating waves; delay-and-sum beamformer; array pattern.

Throughout the course, the integration of digital signal processing concepts in a design environment is emphasized. As part of the laboratory, students work in teams of four on a semester-long design project, which requires trade-offs between performance and cost.

Objectives:

This course will treat a broad range of Digital Signal Processing (DSP) topics. It will strengthen the student's understanding of the foundations of DSP, introduce the students to three major application areas: speech processing, image processing, and array signal processing, and provide extensive hands-on design experience.

Required Texts:

Discrete-Time Signal Processing, Alan V. Oppenheim and Ronald W. Schafer, Prentice-Hall International, Inc., Englewood Cliffs, New Jersey, ISBN 0-13-216771-9.

Digital Signal Processing Applications , edited by J. P. Allebach, C. A. Bouman, and M. D. Zoltowski, (Custom text available from Purdue Armory).

Course Outline:

1.0 Foundations 6 weeks

- 1.1 Continuous-time and discrete-time signals and spectral analysis (CTFT and DTFT)
- 1.2 Continuous-time and discrete-time systems
- 1.3 Sampling
- 1.4 Decimation and interpolation
- 1.5 Z Transform
- 1.6 Discrete Fourier Transform (DFT)
- 1.7 Probabilistic methods in digital signal processing
- 1.8 Quantization

Two Units from topics 2.0, 3.0 and 4.0

2.0 Image Processing 3 weeks

- 2.1 2-D signals and systems
- 2.2 Image processing and enhancement
- 2.3 Image coding
- 2.4 Computed tomography

3.0 Speech and Audio Processing 3 weeks

- 3.1 Speech models and characteristics
- 3.2 Short-time Fourier analysis and synthesis
- 3.3 Differential Pulse Code Modulation (DPCM)
- 3.4 Linear predictive coding

4.0 Array Signal Processing 3 weeks

- 4.1 Basic radar principles
- 4.2 Representation of propagating waves
- 4.3 Beam forming

5.0 Project Preparation 2 weeks

6.0 Examinations 1 week

List of Experiments:

The first 7 experiments deal with the laboratory environment, linear systems, A/D and D/A conversion, filter design, and general digital signal processing techniques. The six remaining experiments deal with the three application areas covered by the course. There are 3 experiments on image processing, 2 on speech processing, and 1 on beamforming. Generally, 3 lab periods will be devoted to the design project, so that 11 of the available experiments are selected depending on the project topic.

1) *Introduction to the Laboratory*: Introduction to the Hewlett-Packard Workstations and X-windows environment, use of the Entropic software, communication with other ECN machines.

2) *Linear Systems:* Demonstrations of properties of linear systems.

3) *Sampling and Quantization:* Demonstration of sampling principles, design of optimal quantizers, design trade-offs in D/A conversion.

4) *Design of Finite Impulse Response (FIR) Filters:* Design of FIR filters using the Kaiser window, the Parks-McClellan algorithm, and a minimum weighted mean-squared error criterion.

5) *Design of Infinite Impulse Response (IIR) Filters:* Design of Butterworth and Chebyshev Type I IIR filters, and comparison of the performance of FIR and IIR filters.

6) *Analysis of Climatic and Stock Market Data:* Design of digital signal processing algorithms for the analysis of temperature and precipitation records, and the generation of buy/sell indicators from stock market data.

7) *Digital Processing of EKG Signals:* Design of both linear and nonlinear (morphological) filters for removal of sinusoidal interference and baseline drift in EKG signals.

8) *Enhancement and Restoration:* Design of grayscale transformations for contrast enhancement, quantization, and feature selection. Design of spatial filters for image sharpening, blur removal, and descreening.

9) *Image Coding:* Design of image coding algorithms via three different approaches: block truncation coding, transform coding using the discrete cosine transform (DCT), and predictive coding.

10) *Computed Tomography:* Algebraic reconstruction technique (ART), demonstration of reconstruction based on Fourier slice theorem, sampling effects.

11) *Time and Frequency Domain Analysis of Speech:* Examination of temporal characteristics of speech, analysis of speech via spectrograms, relation between short-time Fourier transform and the spectrogram.

12) *Speech Vocoder:* Design of a speech coding system based on linear prediction with voiced/unvoiced classification, pitch extraction, and gain computation.

13) *Beamforming:* Demonstration of beamforming process, design of receiver array configuration and receiver delays to achieve desired beamforming objective.