Department of Electrical and Computer Engineering
American University of Beirut

# EECE 491 — Discrete-Time Signal Processing
# Course Information

**Catalog description:**

Digital Signal Processing (DSP) is at the heart of almost all modern technology. This course introduces the fundamentals of DSP systems, including properties of discrete-time linear systems, digital filter design, sampling and reconstruction, A/D and D/A conversion, quantization, discrete-time Fourier analysis, spectral analysis, sample-rate conversion, FFT and fast convolution, filter structures and realizations, and multirate DSP and filter banks. The course also discusses applications of DSP in areas such as speech/audio processing, autonomous vehicles, and software radio. It includes a project related to implementations of DSP applications on embedded processors.

**Credit hours:** 3 credits

**Required or elective:** Elective for ECE and CCE students.

**Prerequisites:**

– *By course*: EECE 340 (Signals and Systems Course)
– *By topic*: Basic knowledge in signals and systems.

**Textbook:**

– A. Oppenheim and R. Schafer, *Discrete-Time Signal Processing,* 3rd ed., Pearson Education, 2009, ISBN 978-0131988422.

**References:**

– J. Proakis and D. Manolakis, *Digital Signal Processing,* 4th ed., Pearson Education, 2007, ISBN: 9780131873742.
– S. Mitra, *Digital Signal Processing: A Computer-Based Approach,* 4th ed., McGraw-Hill, 2011, ISBN: 978-0077366766.
– C. Johnson, Jr., W. Sethares, and A. Klein, *Software Receiver Design*, Cambridge University Press, 2011, ISBN 978-0521189446.
– T. Welch, C. Wright and M. Morrow, *Real-Time Digital Signal Processing from MATLAB to C with the TMS320C6x DSPs*, CRC Press, 2nd ed., 2011, ISBN 978-1439883037.

**Computer usage:**

– Simulink, Matlab DSP/Audio/Communication toolboxes
– TI embedded processor development kit

**Course Objectives**

– Develop skills for analyzing and synthesizing algorithms and systems that process discrete time signals, with emphasis on realization and implementation.
– Explain the theory and concepts behind the operation and construction of DSP systems.
– Analyze basic DSP building blocks including: analog-to-digital (A/D) and digital-to-analog (D/A) converters, digital filters, spectrum analyzers, sample rate converters (up-sampling and down-sampling), and the fast Fourier transform (FFT) algorithm.
– Design digital filters to meet target magnitude and phase responses.
– Design and implement these building blocks and use them effectively in applications.
– Evaluate DSP systems and justify choices among alternative designs.

**Course Topics**

– DSP overview: Signals, systems, and transforms

- Fourier series and transforms: Continuous-time (CT) Fourier series; discrete-time (DT) Fourier series; CT Fourier transform; DT Fourier transform; discrete Fourier transform (DFT)
- DFT spectral analysis
- Sampling, A/D and D/A converters
- LTI discrete-time systems in the transform domain: Impulse response, convolution, difference equations, z-transform, system analysis, BIBO stability.
- Frequency response of LTI systems: lowpass, highpass, and Bandpass Filters
- Linear-phase, generalized linear phase, and minimum-phase systems
- DT processing of CT systems
- Digital filter structures and realizations
- IIR digital filter design: bilinear transformation and impulse invariance
- FIR digital filter design: truncation, windowing, frequency sampling, optimal min-max filter design
- Down-sampling, up-sampling, and oversampling A/D and D/A
- Digital interpolation and noise shaping
- Fast Fourier transform (FFT), fast convolution
- Introduction to multi-rate DSP and multi-rate filter banks
- Quantization effects in fixed-point implementations of filters
- Practical implementations of DSP applications on embedded processors

**Course Learning Outcomes**

Upon successfully completing this course, the student should:

- Be familiar with the terminology that is used in the DSP field
- Understand the basics of DT-LTI systems
- Understand how to sample continuous-time signals.
- Know how to perform discrete-time processing of continuous-time signals.
- Know how to perform continuous-time processing of discrete-time signals.
- Determine what the best discrete-time filters are.
- Know how to design FIR and IIR DSP filters.
- Be able to analyze the effect of finite-precision on numerical and hardware implementation.
- Determine the relationship between the Fourier transform, Fourier Series, Discrete Fourier Transform, and the Fast Fourier Transform.
- Understand how to use the Discrete Fourier Transform to analyze continuous time signals.
- Understand the decimation-in-time and decimation-in-frequency FFT algorithms.
- Know how to do fast convolution using the FFT.
- Understand the basics of downsampling and upsampling, oversampling A/D and D/A, digital interpolation, and noise shaping.
- Know how to design a basic multirate filter bank.

**Class schedule:** Two 75-minute lectures or three 50-minute lectures per week

**Evaluation methods**

1. Homework 10%
2. Project 20%
3. Midterm(s) 35%
4. Final Exam 35%

**Professional component:** Engineering topics: 100%

**Person(s) who prepared this description and date of preparation**

Mohammad M. Mansour, September 2015

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