RUTGERS UNIVERSITY
School of Engineering
Department of Electrical & Computer Engineering

332:447 – Digital Signal Processing Design – Spring 2019

Course Description:

This course develops the theory, design, and implementation of DSP algorithms for a wide range of applications. The material is divided into the following topic areas:

Topic	*Lectures*
Real-Time Signal Processing	5
IIR Digital Filter Design	3
Fast Transforms and Applications	8
Signal Extraction and Modeling	7
Adaptive Signal Processing	4

- *Real-Time Signal Processing.* Filter realizations for real-time processing, circular delay-line buffers, digital audio effects, dynamic-range control, delay-based effects, reverb.
- *IIR Digital Filter Design.* IIR filter design methods using the bilinear and other spectral transformations of an analog prototype filter. Normalized lattice and other realizations. Quantization effects in digital filter structures.
- *Fast Transforms.* FFT, overlap-add and overlap-save methods of fast convolution, with applications to matched filtering. Short-time Fourier transform (STFT) implemented via the FFT or by DFT filter banks, with applications to time-scale and pitch-scale modification of speech and audio signals, such as the phase vocoder. Data compression, DCT, MDCT, and TDAC[†]. Two-dimensional FFTs and DCTs, image deblurring and denoising, and discrete wavelet transform, if time permits.
- *Signal Extraction and Modeling.* Data analysis methods for signal and trend extraction and prediction in business, finance, climate, biomedical, geophysical, and other time-series applications. Local polynomial filters for signal extraction and noise reduction. Predictive FIR filters, exponentially weighted moving average (EMA) filters, and other technical analysis tools in financial market trading. Regularized least-squares methods, Whittaker-Henderson smoothing filters. Introduction to *sparse signal processing* and modeling methods, such as the LASSO or BPD[‡], with applications to trend extraction, sparse deconvolution, and compressive sensing.
- *Adaptive Signal Processing.* LMS and RLS algorithms for adaptive signal and array processing with applications to signal estimation and prediction, noise canceling, channel equalization, system identification, echo cancellation, acoustic noise control, beamforming, and smart antennas.

Instructor:

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Office hours: Tuesday, 1:30–3:00 PM.

Text:

Lecture notes, textbooks, reference papers, and other course materials will be placed on Sakai.

[†]Discrete Cosine Transform, Modified DCT, and Time-Domain Aliasing Cancellation
[‡]Least Absolute Shrinkage and Selection Operator, Basis Pursuit Denoising

Prerequisites:

332:346 DSP, as well as working knowledge of MATLAB, some C, numerical linear algebra, and random signal concepts such as autocorrelation functions and power spectral densities.

Course Requirements and Grading:

The final course grade will be based on nine weekly design projects (7% per project), a midterm exam (13%), and a final project (24%). The weekly projects are an essential and *required* part of the course. They represent a substantial amount of work and may not be delayed or skipped.

A tentative list† of the course projects is given below. Project reports in PDF format are to be handed *in person, in class,* on the due date, and also uploaded to Sakai in a *single zipped file* that includes all required executable Matlab M-files, and generated data, audio, or image files.

Please adhere to the following formatting guidelines for your reports:

(a) Include a substantial discussion section that states the purposes of the project and presents and evaluates the results—include here all the relevant graphs and theory.
(b) Attach all computer code as an appendix at the end of your report. Unless specifically asked, please never include any long numerical listings of data points.
(c) Work alone on all projects.

Your project report should be formatted preferably with LaTeX and written as though you were teaching or presenting the subject to someone who knows some DSP but is not necessarily familiar with the topic. It should contain enough details and explanations to be self-contained and easily understood by such reader.

Projects:

	Project	assigned	due
1.	Removing unwanted echoes and reverb from recorded audio	01/28	02/06
2.	Digital dynamic range compressor/expander design	02/06	02/18
3.	Normalized lattice and coefficient quantization effects	02/18	02/27
4.	Fast convolution methods and matched filter applications	02/27	03/11
5.	STFT phase vocoder for time-scale and pitch-scale modification	03/11	03/27
6.	DCT data compression for audio and images, MDCT and TDAC	03/27	04/03
7.	Regularized and sparse methods of trend extraction	04/03	04/15
8.	Sparse modeling, regularized least-squares, LASSO/BPD	04/15	04/24
9.	Adaptive and block processing methods of linear prediction	04/24	05/06
10.	Adaptive filtering	04/24	05/09

Academic Integrity:

By taking this course *you agree* to work alone on your projects, and *you accept and adhere* to the Rutgers academic integrity policy described in:

`http://academicintegrity.rutgers.edu/academic-integrity-at-rutgers/`

Please note also that cell phones and laptops are *not allowed* during lectures or exams (except for taking class notes.)

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Please read also the IEEE code of ethics that should guide your professional life:

† subject to change as the course progresses

https://www.ieee.org/about/corporate/governance/p7-8.html