ESE 531: Digital Signal Processing

Lec 1: January 16, 2020
Introduction and Overview

Lecture Outline

- Course Topics Overview
- Learning Objectives
- Course Structure
- Course Policies
- Course Content
- What is DSP?
- DSP Examples

Course Topics Overview

- Discrete-Time (DT) Signals
- Time-Domain Analysis of DT Systems
- Discrete Fourier Transform (DFT)
- Fast Fourier Transform (FFT)
- Discrete-Time Fourier Transform (DTFT)
- z-Transform
- Sampling of Continuous Time Signals
- Data Converters and Modulation
- Upsampling/Downsampling
- Discrete-Time Filter Design
- Special Topics

Learning Objectives

- Learn the fundamentals of digital signal processing
- Provide an understanding of discrete-time signals and systems and digital filters
- Enable you to apply DSP concepts to a wide range of fields
- Gain the ability to read the technical literature on DSP
- Apply the techniques learned in a final project encompassing many different application types

Learning Objectives

- In other words…

- Math, Math, Math*

*With MATLAB application for intuition

Course Structure

- TR Lecture, 4:30-6:00pm in DRLB A2
  - Start 5 minutes after, end 5 minutes early (~75-80min)
- Website (http://www.seas.upenn.edu/~ese531/)
  - Course calendar is used for all handouts (lectures slides, assignments, and readings)
  - Canvas used for assignment submission and grades
  - Piazza used for announcements and discussions

Course Structure

- Course Staff (complete info on course website)
- Instructor: Tania Khanna
  - Office hours – Wednesday 1-3 pm or by appointment
  - Email: taniak@seas.upenn.edu
    - Best way to reach me
- TAs:
  - Dhaval Bhatt
    - Office hours – Th 5-6pm, F 3-5pm
  - Yinghao Zhang
    - Office hours – W 5-6pm, Sat 10am-12pm

Course Structure

- Lectures
  - Statistically speaking, you will do better if you come to lecture
  - Better if interactive, **everyone** engaged
    - Asking and answering questions
    - Actively thinking about material
- Textbook
  - A. V. Oppenheim and R. W. Schafer (with J. R. Buck), Discrete-Time Signal Processing. 3rd. Edition, Prentice-Hall, 2010
  - Class will follow text structure… mostly

Course Structure - Assignments/Exams

- Homework – one week long (~10 total)* [25%]
  - Due Sundays at midnight
  - Combination of book problems and matlab problems
  - Lowest grade dropped
- Project – two weeks long [30%]
  - Work in pairs
  - Combination of different DSP applications
- Midterm exam [20%]
- Final exam [25%]

Course Policies

See web page for full details

- Turn homework in Canvas
  - Anything handwritten/drawn must be clearly legible
  - Submit code, graphs, test results when specified
  - NO LATE HOMEWORKS!
- Individual work (except project)
  - code, test simulations, analysis, writeups
  - May discuss strategies, but acknowledge help

Course Content

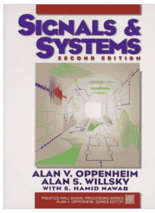
- Introduction
- Discrete Time Signals & Systems
- Discrete Time Fourier Transform
- Z-Transform
- Inverse Z-Transform
- Sampling of Continuous Time Signals
- Frequency Domain of Discrete Time Series
- Downsampling/Upsampling
- Data Converters, Sigma Delta Modulation
- Frequency Response of LTI Systems
- Signal Flow Representation
- Basic Structures for IIR and FIR Systems
- Design of IIR and FIR Filters
- Butterworth, Chebyshev, and Elliptic Filters
- Filter Banks
- Adaptive Filters
- Computation of the Discrete Fourier Transform
- Fast Fourier Transform

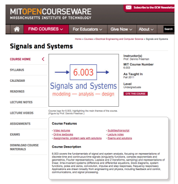
Course Content

ESE531 Spring 2020 Working Schedule

Wk	Lect.	Date		Lecture	Slides	Due	Reading
1	1	1/16	Th	Intro/Overview	[lec1] [lec1_6up]		review course webpage completely
2	2	1/21	T	Discrete Time Signals & Systems, Part 1			2.1-2.2
	3	1/23	Th	Discrete Time Signals & Systems, Part 2		HW0 Diagnostic Quiz (in Canvas)	2.3-2.4
3	4	1/28	T	Discrete Time Fourier Transform			2.5-2.7
	5	1/30	Th	Z-Transform			3.0-3.1
		2/2	Su			HW 1 MATLAB Tutorial	
4	6	2/4	T	Inverse Z-Transform			3.3
	7	2/6	Th	Sampling and Reconstruction			4.0-4.3
		2/9	Su			HW 2	
5	8	2/11	T	DT/CT Processing of CT/DT Signals, Impulse Invariance			4.4-4.5
	9	2/13	Th	Downsampling/Upsampling and Practical Interpolation			4.6-4.6.3
		2/16	Su			HW 3	
6	10	2/18	T	Non-Integer and Multi-rate Sampling			4.6.4-4.7.2
	11	2/20	Th	Polyphase Decomposition and Multi-rate Filter Banks			4.7
		2/23	Su			HW 4	
7	12	2/25	T	Data Converters and Noise Shaping			4.8-4.9
	13	2/27	Th	Frequency Response of LTI Systems			5.0-5.4
		3/1	Su			HW 5	
8	14	3/3	T	All-pass Systems, Min Phase Decomposition			5.4-5.6
		3/5	Th	Midterm Exam, during class in DRLB A2			
9		3/10	T	SPRING BREAK -- no class			
		3/12	Th	SPRING BREAK -- no class			
10	15	3/17	T	Generalized Linear Phase Systems			5.7
	16	3/19	Th	Design of IIR Filters			7.0-7.2
		3/22	Su			HW 6	
11	17	3/24	T	Design of FIR Filters			7.3-7.5
	18	3/26	Th	Optimal Filter Design			7.7

Signals and Systems Review





- Diagnostic Quiz in Canvas
  - Complete by 1/23 for credit
  - Review in HW 0
    - Also Matlab Tutorial

What is DSP

DSP is Everywhere

- Sound applications
  - Compression, enhancement, special effects, synthesis, recognition, echo cancellation,…
  - Cell phones, MP3 players, movies, dictation, text-to-speech,…
- Communication
  - Modulation, coding, detection, equalization, echo cancellation,…
  - Cell Phones, dial-up modem, DSL modem, Satellite Receiver,…
- Automotive
  - ABS, GPS, Active Noise Cancellation, Cruise Control, Parking, Driverless Cars…

DSP is Everywhere (con't)

- Medical
  - Magnetic Resonance, Tomography, Electrocardiogram, Biometric Monitoring…
- Military
  - Radar, Sonar, Space photographs, remote sensing,…
- Image and Video Applications
  - DVD, JPEG, Movie special effects, video conferencing…
- Mechanical
  - Motor control, process control, oil and mineral prospecting,…

Signal Processing

- Humans are the most advanced signal processors
  - speech and pattern recognition, speech synthesis,…
- We encounter many types of signals in various applications
  - Electrical signals: voltage, current, magnetic and electric fields,…
  - Mechanical signals: velocity, force, displacement,…
  - Acoustic signals: sound, vibration,…
  - Other signals: pressure, temperature, biometrics…
- Most real-world signals are analog
  - They are continuous in time and amplitude
  - Convert to voltage or currents using sensors and transducers

Signal Processing (con't)

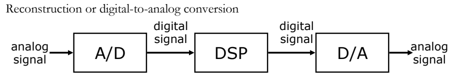
- Analog circuits process these signals using
  - Resistors, Capacitors, Inductors, Amplifiers,…
- Analog signal processing examples
  - Audio processing in FM radios
  - Video processing in traditional TV sets

Limitations of Analog Signal Processing

- Accuracy limitations due to
  - Component tolerances
  - Undesired nonlinearities
- Limited repeatability due to
  - Tolerances
  - Changes in environmental conditions
    - Temperature
    - Vibration
- Sensitivity to electrical noise
- Limited dynamic range for voltage and currents
- Inflexibility to changes
- Difficulty of implementing certain operations
  - Nonlinear operations
  - Time-varying operations
- Difficulty of storing information

Digital Signal Processing

- Represent signals by a sequence of numbers
  - Sampling and quantization (or analog-to-digital conversion)
- Perform processing on these numbers with a digital processor
  - Digital signal processing
- Reconstruct analog signal from processed numbers
  - Reconstruction or digital-to-analog conversion



- Analog input → analog output
  - Eg. Digital recording music
- Analog input → digital output
  - Eg. Touch tone phone dialing, speech to text
- Digital input → analog output
  - Eg. Text to speech
- Digital input → digital output
  - Eg. Compression of a file on computer

Pros and Cons of Digital Signal Processing

- Pros
  - Accuracy can be controlled by choosing word length
  - Repeatable
  - Sensitivity to electrical noise is minimal
  - Dynamic range can be controlled using floating point numbers
  - Flexibility can be achieved with software implementations
  - Non-linear and time-varying operations are easier to implement
  - Digital storage is cheap
  - Digital information can be encrypted for security
  - Price/performance and reduced time-to-market

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- Cons
  - Sampling causes loss of information
  - A/D and D/A requires mixed-signal hardware
  - Limited speed of processors
  - Quantization and round-off errors

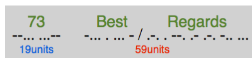
DSP Examples

Example I: Audio Compression

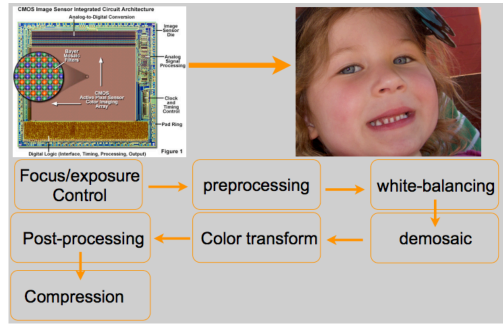
- Compress audio by 10x without perceptual loss of quality
- Sophisticated processing based on models of human perception
- 3MB files instead of 30MB
  - Entire industry changed in less than 10 years!

Historical Forms of Compression

- Morse code: dots (1 unit) dashes (3 units)
  - Code Length inversely proportional to frequency of character
    - E (12.7%) = . (1 unit) Q (0.1%) = --.- (10 units)
- "92 Code"
  - Used by Western-Union in 1859 to reduce BW on telegraph lines by numerical codes for frequently used phrases
    - 1 = wait a minute
    - 73 = Best Regards
    - 88 = Loves and Kisses



Example II: Digital Imaging Camera



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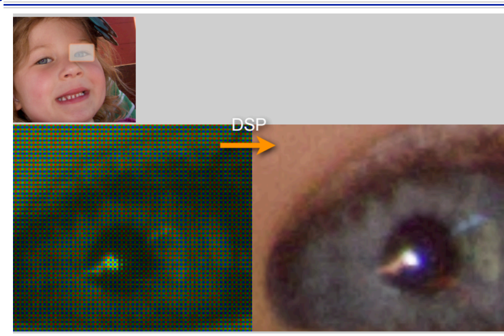

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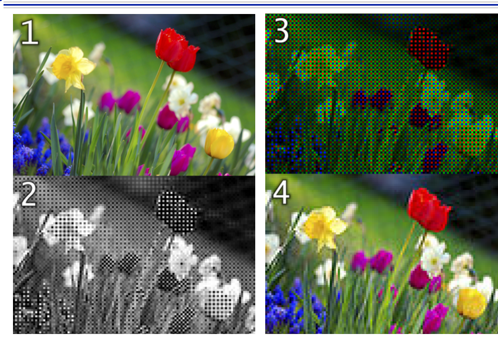

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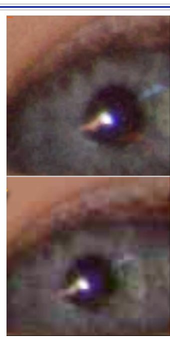


- Compression of 40x without perceptual loss of quality.
- Example of slight over compression: difference enables 60x compression!

Computational Photography

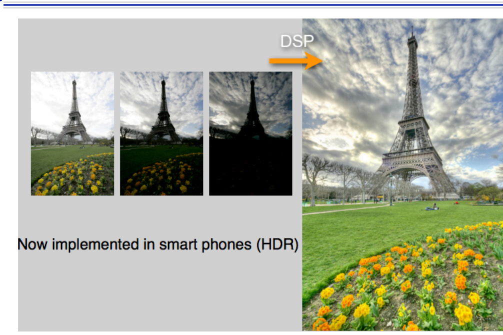


Image Processing

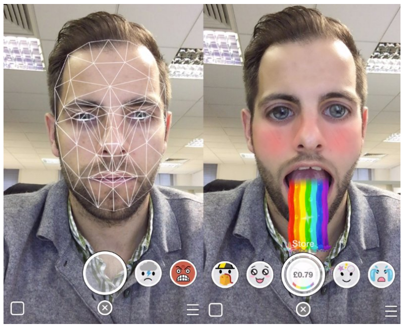
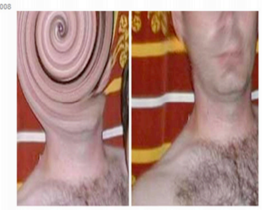


Image Processing - Saves Lives

Canadian 'swirl face' jailed in Thailand

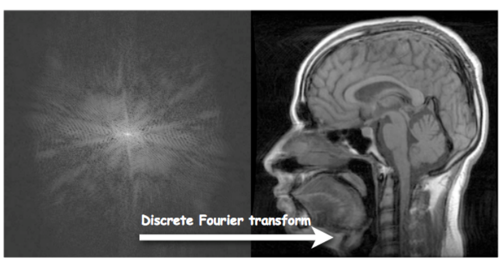
August 15, 2008



Images released by Interpol in 2007 show the 'unswirling' of the internet pictures that led to the capture of Christopher Paul Neil.

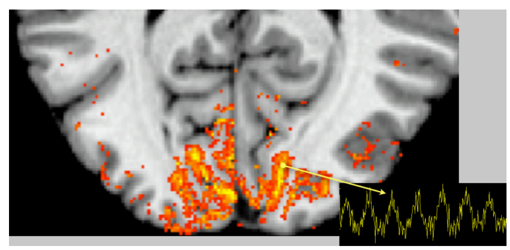
Example III: MRI

k-space (raw data) Image



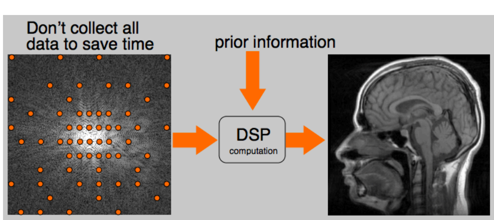
fMRI example

- Sensitivity to blood oxygenation
  - response to brain activity Convert from one signal to another

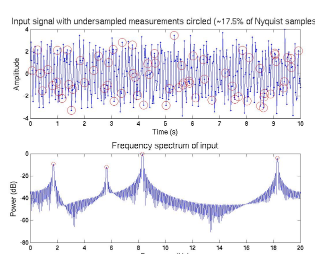


Compressive Sampling

- Compression meets sampling



Example: Sum of Sinusoids

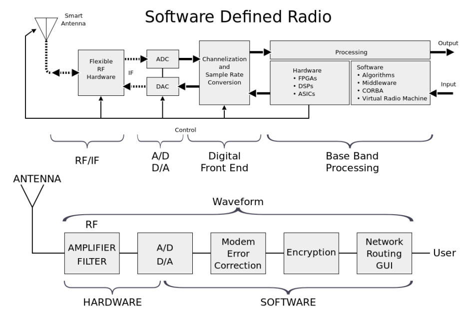


- Sense signal randomly M times
  - $M > C \cdot \mu^2(\Phi, \Psi) \cdot S \cdot \log N$
- Recover with linear program

Example IV: Software Defined Radio

- Traditional radio:
  - Hardware receiver/mixers/demodulators/filtering
  - Outputs analog signals or digital bits
- Software Defined Radio:
  - Uses RF front end for baseband signal
  - High speed ADC digitizes samples
  - All processing chain done in software

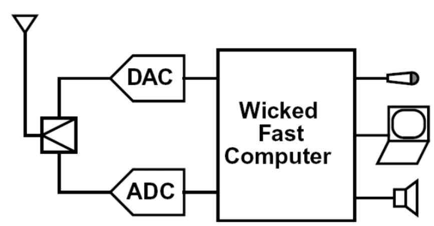
Software Defined Radio



Software Defined Radio

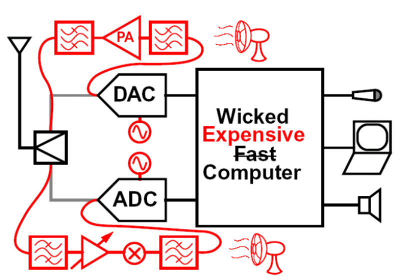
- Advantages:
  - Flexibility
  - Upgradable
  - Sophisticated processing
  - Ideal Processing chain
  - not approximate like in analog hardware
- Already used in consumer electronics
  - Cellphone baseband processors
  - Wifi, GPS, etc....

Software Radio Vision



[Schreier, "ADCs and DACs: Marching Towards the Antenna," GIRAFE workshop, ISSCC 2003]

Software Radio Reality



[Schreier, "ADCs and DACs: Marching Towards the Antenna," GIRAFE workshop, ISSCC 2003]

Shameless Plug

- If you are interested in how Analog to digital converters work and how to make them
- Take ESE 568!
- Good to know both sides of the system

Future of ADC design

- Today's ADCs are extremely well optimized
- For non-incremental improvements, we must explore new ideas in signal processing that tackle ADC inefficiency at the system level
  - Compressed sensing
  - Finite innovation rate sampling
  - Other ideas?

Filter Design Example

Optimal Filter Design

- Window method
  - Design Filters heuristically using windowed sinc functions
- Optimal design
  - Design a filter h[n] with $H(e^{j\omega})$
  - Approximate $H_d(e^{j\omega})$ with some optimality criteria - or satisfies specs.

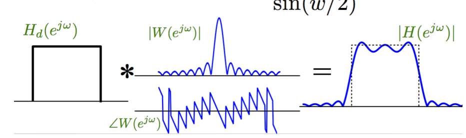
FIR Design by Windowing

- Desired filter,

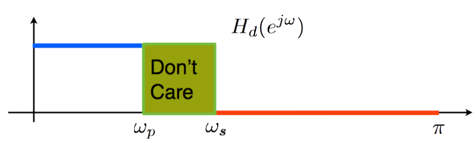
$$H(e^{j\omega}) = H_d(e^{j\omega}) * W(e^{j\omega})$$

- For Boxcar (rectangular) window

$$W(e^{j\omega}) = e^{-j\omega\frac{M}{2}} \frac{\sin(w(M+1)/2)}{\sin(w/2)}$$



FIR Design by Optimality



- Least Squares:

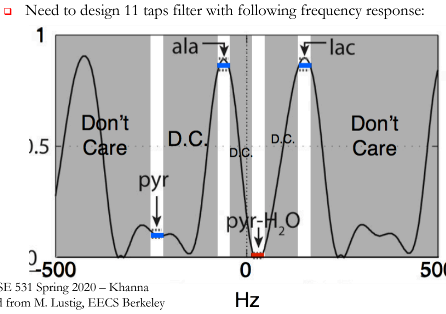
$$\text{minimize} \int_{\omega \in \text{care}} |H(e^{j\omega}) - H_d(e^{j\omega})|^2 d\omega$$

- Variation: Weighted Least Squares:

$$\text{minimize} \int_{-\pi}^{\pi} W(\omega)|H(e^{j\omega}) - H_d(e^{j\omega})|^2 d\omega$$

Example of Complex Filter

- Larson et. al, "Multiband Excitation Pulses for Hyperpolarized 13C Dynamic Chemical Shift Imaging" JMR 2008;194(1):121-127
- Need to design 11 taps filter with following frequency response:



Admin

- Find web, get text, start HW 0 and assigned reading…
  - http://www.seas.upenn.edu/~ese531
  - https://piazza.com/upenn/spring2020/ese531/
  - https://canvas.upenn.edu/
- Diagnostic quiz due 1/23
- Remaining Questions?