ECE 420 Lecture 1 Jan 14 2019

Course Overview

Course Administrative Info

• Webpage: https://courses.engr.illinois.edu/ece420/sp2019/

- Lecture: Mondays 2:00-2:50PM, 2013 ECEB
 - Lecturer: Dr. Jeffrey Brokish

- Labs: 2:00-3:50PM, 5027 ECEB
 - ABA: Tue, ABC: Wed, ABD: Thu, ABE: Fri
 - TAs
 - Grant Greenberg
 - Yu-Jeh Liu

Course Structure



- First half: 7 Structured Labs
 - Embedded DSP development framework
 - High-level (Python) \rightarrow Embedded (Android with Java/C)
 - Different signal modalities and interfaces: IMU, audio, visual
 - Basic DSP algorithms
 - Digital filtering
 - Spectral analysis
 - Auto-correlation analysis: pitch detection/correction
 - Image and multidimensional signal processing

- First half: 7 Structured Labs Format
 - Prelab [Individual]
 - Complete individually prior to lab, submit to TA
 - Quiz [Individual]
 - Overview of concepts from previous lab
 - Demo [Group]
 - Demonstrate work from previous week to TA, answer questions
 - Lab work [Group]
- No prelab, quiz or demo this week

- Second half: 'Student Choice' Group Projects (subj. to approval)
 - Start with an Assigned Project Lab
 - Explore implementation of a DSP algorithm from the literature
 - In Python, 2 week duration
 - Jumping off point for the Final Project
 - Final project
 - Proposal (the pitch) and Design Review
 - Deliverables and 3 weekly milestones
 - Final Project Demo and Presentation
 - Final Report (and optional Video)
 - Recommended not to wait until week 7 to start exploring options

- Grading
 - Structured Labs 40%
 - Assigned Project Lab 10%
 - Final Project 45%
 - Final Quiz 5%
- For more detailed breakdown consult the course website

Embedded Digital Signal Processing

Embedded Digital Signal Processing (DSP)

- "Signal": physical quantity that carries information
- **"Processing"**: series of steps to achieve a particular end
- "Digital": done by computers, microprocessors, or logic circuits
- "Embedded": part of a complete device (hardware), often with real-time constraints

Background for DSP



Example: Speech Recognition using DSP



Digital Cameras

Original After DSP





www.dxo.com

Multimedia Compression





- Provide the crucial technology for:
 - WWW with multimedia content (e.g. audio, image, and video)
 - DVD
 - Digital cameras, camera phones
 - MP3, iPod

Medical Imaging: Ultrasound (US), Computer Tomography (CT), Magnetic Resonance Imaging (MRI), ...



www.imaginis.com/ct-scan





DSP Appliances



Smart Phones



Example Smartphone Chip



Best Practices in Developing DSP Software: Systematic Debugging

- First, develop and test DSP algorithms in high-level languages (Python, MATLAB)
 - More rapid development, more extensive tools available
 - Use of test/training signals
 - Tuning of algorithmic parameters
 - Quantification of algorithm performance (SNR, detection accuracy)
 - Examination of intermediate / final signal outputs in various domains
 - Provides a reference for comparison against embedded implementation
- Then, port tested algorithms into embedded platform (Android)
- Sometimes, may need to go back and refine algorithms

Practical Considerations

- Reducing power is *critical* for mobile real-time devices
 - Battery drain is #1 reason for users to turn off an app
- Ways to save power
 - 16-bit fixed point, not floating point
 - Low clock speed/voltage through parallelism
 - Simple, low-power microprocessor architecture
 - Program in low-level languages
 - Use hardware accelerators, or dedicated computing units



Upcoming Labs

- Lab 1 Pedometer
 - Not as signal processing heavy
 - Introduction to development tools and Android platform
- Lab 2 Digital Filtering
 - Signal processing at its finest!
 - Work with audio signals, filter design, and the OpenSL framework

$$x_a(t) \longrightarrow A/D \longrightarrow Digital Filter y(n) \longrightarrow D/A \longrightarrow y_a(t)$$

Filter Design: Mapping Analog to Digital Frequencies

If we sample an analog signal $x_a(t)$ to obtain a digital signal $x_d[n] = x_a(nT)$ using the sampling frequency $f_s = 1/T$, then their Fourier transforms are related by:

$$X_d(\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} X_a\left(\frac{\omega - 2k\pi}{T}\right).$$

Hence, assuming no aliasing (i.e. $X_a(\Omega) = 0$ for $|\Omega| \le \pi/T$) then an analog frequency $\Omega = 2\pi f$ (where $|\Omega| \le \pi/T$) is mapped to a digital frequency

$$\omega = \Omega T = \frac{2\pi f}{f_s}.$$

In particular, the Nyquist frequency $f = f_s/2$ is mapped to $\omega = \pi$.

Digital Filter Implementation

Given a digital filter

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1 z^{-1} + \ldots + b_K z^{-K}}{1 + a_1 z^{-1} + \ldots + a_L z^{-L}},$$

then the filtering by H(z):

$$x[n] \longrightarrow H(z) \longrightarrow y[n]$$

can be implemented for each n as:

 $y[n] = (b_0 x[n] + b_1 x[n-1] + \ldots + b_K x[n-K]) - (a_1 y[n-1] + \ldots + a_L y[n-L]).$

Examples of Filters



Digital Filter Design

- Typically filters have a target frequency response expressed in the Fourier domain
- We will be filtering in the spatial domain, and for most applications achieving a 'perfect' frequency response is impossible
- One of the biggest constraints will be computational limits
- How do you get the most out of a filter?
 - Design criteria 'optimal' in some sense
 - FIR vs. IIR complexity vs. fidelity

Summary

- Lab 1 this week (no prelab)
- No lecture next Monday (Jan 21)
- Lab 2 next week (with prelab)
- Let's go do some amazing work!