Final Project

Proposal Due Monday 04/17/17
Report Due Wednesday 05/05/17

Reading:

- For project ideas, browse links on the website, search the web, and read the recommended textbooks.

Overview:

The purpose of the class project is to encourage you to combine the material you learned in this class with your own ideas to create an interesting digital audio processing project. The project should be completed in groups of two and students are free to choose their own partners. The final project will count for 25% percent of your grade labeled "Projects" on the syllabus.

Each group will turn in a project report. The report should be composed using a word processor (e.g., Word, Pages, LaTeX, etc...) and include a cover page with full names of all group members. The remaining pages should contain a description of (1) the idea behind your project, (2) the materials you studied to start your project, (3) the materials you created to accomplish your project, and (4) an overall summary of what you learned during the process.

Your project grade will be based on: (1) the work done to accomplish the results and (2) the report. Submission should be done electronically through Sakai with the report PDF attached along with a ZIP file containing all code necessary to reproduce your results.

Plagiarism is a very serious offense in Academia. Any figures in the paper not generated by you should labeled "Reproduced from [...]". Any portions of any simulation code (e.g., MATLAB, C, etc...) not written by you be clearly marked in your source files. The original source of any mathematical derivation or proof should be explicitly cited.

Proposal: The first task is to pick a partner and a project idea. Then, submit a short proposal (e.g., 2 paragraph description) to me. I will provide feedback to help you find something reasonable that can work in the time allotted.

Project Ideas:

1. **Your Own Idea:** Combine the material you learned in this class with your own ideas to create an interesting digital audio processing project.

2. **Advanced Physical Modeling for Synthesis:** Building on the guitar synthesizer lab, the idea is to model a more complicated instrument (e.g., see `http://www.jstor.org/stable/3681331`) and/or to learn the parameters of an instrument model from sampled audio to make the synthesis more realistic. For example, see

   `https://pdfs.semanticscholar.org/a3b5/89e460d43fff6057a8d85db77a28d7b3c09e.pdf`

3. **Auto-Tune Digital Effect:** Combine a pitch-estimation algorithm with a pitch-shifting algorithm to create your own "auto-tune" effect. The idea is estimate the pitch, quantize

pitch to a desired value (e.g., the nearest note in a scale), and then pitch-shift the input signal to match the quantized note. It would great if this were implemented as a Matlab audio plugin so that it used in real time and compiled into a VST audio plugin.

4. **Perceptual-Based Audio Compression:** Using a filter-bank front-end, design perceptual-based audio encoder that quantizes coefficients in different frequency bins with different step sizes. For best performance, one should use an MDCT filter bank (for details on the MDCT, see `http://pfister.ee.duke.edu/courses/ece485/mdct.pdf`). The basic steps are: (i) compute filter-bank outputs, (ii) compute perceptual masking thresholds, (iii) choose quantization step sizes to keep noise below masking thresholds, (iv) quantize transform coefficients, and (v) invert transform to hear the effect of quantization. A short outline pdf will be released for this project to aid interested students.

5. **Test the Effect of Multiple Microphones using Array Processing:** Write MATLAB code to model the response of multiple microphones to multiple audio signals at different spatial locations. Compute the spatial-frequency response of a linear array with optimal combining weights. Simulate the a noisy environment with multiple noise and signal sources. By combining the signals from multiple microphones, can you generate a single signal with increased SNR? This topic will also be covered in a lab. We will also provide a multiple-microphone audio sample from which you can separate multiple individuals who are speaking.

6. **Pattern Classification for Vowel Recognition:** The goal of this project is write a program that automatically labels an unknown vowel sound. There is a database of recorded vowel sounds available from `https://homepages.wmich.edu/~hillenbr/voweldata.html`. The idea is to use a perceptually-weighted filter bank (e.g., MEL Cepstral coefficients) to map 40ms of speech into a feature vector. Then, a classifier is trained and tested using labeled samples. For interested students, we will provide a rough outline of a suggested method. Note: this is an advanced project that will likely require more effort than others.