Lab #9: End of Semester - Real-time DSP Project

Purpose

The purpose of this lab is to expand on the DSP algorithms that you have previously used in earlier labs to experiment and create new signal processing techniques. Thus, the highest points will be awarded for innovative DSP algorithms that go beyond what we have covered earlier. In lab and class we have implemented the following: examples of aliasing, resampling, echo & reverb, FIR Filters, IIR Filters, and a FFT based spectrum analyzer. Your project may use these components as tools for your larger system and should also include new techniques that you have researched and/or created on your own.

The lab is broken up into two parts: Part I is worth 40% of a regular lab and consists of a research/development/experimentation stage. You should research algorithms similar to those you plan to implement by obtaining papers, books and other material related to your subject. You should then propose your algorithm and test it out with sound files in Matlab. All of this should be shown to your TA during your upcoming section for maximum points (40%). Part II is then the implementation of your equations/Matlab testing on your DSP board. This is worth 160% and will be judged by the TAs, our class and myself during a presentation the last week of class. For this presentation, you should be prepared to introduce your topic and answer questions relating to equations/algorithms that you have implemented.

Points will also be awarded for efficient software implementation and meeting real-time specification: processing must take less time than the time to receive a new sample or block of samples from 32-48 KHz.

You may also receive points for new DSP hardware additions such as: a working SD card interface, Nordic wireless communication and other hardware interfaces that we have not covered in class/lab this semester. Additional points may also be awarded unique GUIs for controlling the DSP and other equipment that is part of your project.

Here is a quick summary of some of the proposed ideas from past and present semesters:

Vocoder (band pass filter bank or FFT used to determine a main carrier frequency and then applies this to another synthetically generated sound), elementary speech recognition for simple control, spectrum analyzer that uses computed frequency information to alter/enhance music or speech (vocoder, auto-tune, pitch shifters), guitar special effects (significantly better phasor, flanger, chorus, filter effects than what was done in lab), sound synthesizer (additive synthesis, FM modulation, granular synthesis, theremin and many others that you can research online). Also check out a reference in my office, "Numerical Recipes in 'C'".

Points/Grading Break-down

Part I	40%	;Research materials (10%), proposed equations & algorithms (10%), Matlab and/or ‘C’ test results (20%)
Part II	160%	;Explanation of your design (15%), software efficiency/buffering techniques/optimized routines (15%), ;real-time operation (10%), new hardware and/or software (20%), complexity of the task (40%), ;wow factor/novel new idea (40%), bug-free function (10%), clean sound (10%)

Suggestions for Reducing/Eliminating Noise

Use battery power instead of USB/wall wart. Electrolytic/Tantalum/Mono caps on all voltage references. The ferrite bead is your friend. Remind me to discuss this and routing signals for minimal noise in class.