ELEC484/532 Audio Signal Processing

Assignment 1. Summer 2007 (10 pts)

The main goal of this assignment is to motivate you to experiment with various digital audio effects for creative purposes and to gain insight about some of the ideas we dicussed in class.

Read the questions carefully and do not hesitate to contact me via email if you have any questions or need some clarifications.

The assignment is due **June 11th** in class. For each question you should describe what you did, what programming environment you used, show screenshots/figures/plots and provide all the relevant code. For the audio files you create the easiest would be if you could put them on a website and provide me with a link. If you don't have access to a website then you can send it by email, give me a CD or transfer them using a usb key to my laptop. You can use any text processing environment you want but what you hand me must be in **pdf** format. Any programming environment, tool is also acceptable but you should mention it explicitely in your report.

There are 4 question each of which is worth 2 points. The last two points will be based on the quality and completeness of your report. If you feel like reading about something related to this assignment that won't help your grade in any way check out:
`http://en.wikipedia.org/wiki/Musique_concrete`

Enjoy, George Tzanetakis

Question 1 (2 pt)

Matlab or Octave are probably the easiest tools to use for this question. You can do this question either using Matlab or Octave. As we discussed during the lectures the class of sinusoids of a particular frequency are closed under the operation of addition. That means that adding two sinusoids that have the same frequency with different amplitude and phase results in a sinusoid at the same frequency. Write Matlab code to show that this property does NOT hold for sawtooth waves of the same frequency but with different amplitude and phase. Provide the corresponding plots to illustrate this as well as sounds.

Question 2 (2pts)

Choose 5 samples from the .wav file on the dafx.de website. Manipulate them using sound editing software (some ideas are provided on the course webpage - Audacity would be my recommendation) to create new sounds. Organize these new sounds in a sound collage between 15-30 seconds that is musically interesting and demonstrates concepts covered in class. Export your project to a .wav file. Write a summary of your composition and what tools you used. Try to have a range of frequency conent, both low and high, include at least three tracks in your audacity project, and creativy DOES count.

Question 3 (2pts)

This question is similar to Question 2 but you will be restricted to one input sound: white noise and one type of digital audio effects: filters (either FIR or IIR). You can design/implement as many filters as you want but you will need to document how you designed them, what type they are etc. Parametric filters such as the ones described in the DAFX textbook that allow flexible control of their center frequency/bandwidth can help you create some really interesting sounds. You can either provide a 15-30 second collage as in the previous question or if you feel ambitious write a graphical user-interface or patch in an interactive environment such as Max/MSP, Pd, Chuck, Marsyas that allows a listener to “interact” with your composition.

Question 4 (2pts)

Generate the sound corresponding to a sampled square wave. Then generate the sound corresponding to sampling a square wave that has no harmonics above the Nyquist frequency so there is no aliasing. Compare the sounds with and without aliasing. Plot the corresponding Magnitude Spectra. Can you find an interesting use for intentional aliasing ?