Creative Computing II

Applying Audio Effects

2nd March 2010

This lab sheet covers the application of digital audio effects.

1. This part helps to explain the return value of *Octave*'s `fft` operator, using as source material a real musical signal.

   (a) Obtain the provided `scale.wav` file, and use the `wavread` and `sound`[1] *Octave* functions to play it. What is the sampling rate for that sound file?

   (b) Plot the first thousand samples of the first channel. Can you determine any apparent regularity in your plot? If so, estimate the period of the regularity in samples and hence the frequency of that regularity.

   (c) Take the Fourier Transform of the first thousand samples of the first channel, and plot the magnitude of it. Note the symmetry around the mid-point.

   (d) Plot the magnitude of the Fourier Transform you obtained over the first 100 frequency bins, and identify the bin number of the highest peak.

   (e) The bins are in ascending order of frequency, each corresponding to an integer multiple of $\frac{1}{T}$, where T is the total time of the signal whose transform was taken. Compute the frequency corresponding to the bin identified in part 1d

   (f) Repeat parts 1b to 1e with the thousand audio samples starting at 39001, 78001, 117001, 156001, 195001, 234001 and 273001. You may wish to write functions to automate some of this work. Note all the frequencies that you obtain, and attempt to relate them to the musical scales discussed last term.

2. This part illustrates the effects of basic filtering techniques on simple musical signals.

   (a) Using the same `scale.wav` example file, take the Fourier Transform of an entire channel's signal. Verify that taking the inverse Fourier Transform (with `ifft`) produces the original signal.

   (b) Removing a particular frequency component from a signal is equivalent to setting its Fourier component to zero. Do this for all frequency bins between 3001 and 311012 (inclusive).

   (c) Listen to the sound produced by taking the inverse Fourier Transform of your modified spectrum, and comment on the relationships between the new sound, the old one, and the frequency components removed.

3. This part takes you through applying the acoustics of a highly reverberant space (York Minster) to a simply synthesized piece of musical audio.

   (a) First, we need an impulse-response for York Minster. Visit `http://space-net.org.uk/node/54` and download the 'Stereo - ORTF' file; listen to it, and think about how that sound relates to an impulse-response.

[1] If necessary, obtain the `sound.m` function from the session on 25th November.

(b) Read it into *Octave* using `wavread`, using the way of calling `wavread` which tells you the sample rate of the wave data; note that sample rate.

(c) Obtain `toccata.wav` from the teaching website, and read that file into *Octave* using the `wavread` function, again noting its sample rate.

(d) You should find that the two audio files were sampled at different rates. In order to treat one as the input to the other treated as a system, they must have the same sample rate. Use the `resample Octave` operator to convert one of the signals to the other's sample rate.

(e) Try convolving the two signals together to produce the York Minster system's output from the input musical signal. How long are you prepared to wait?

(f) Use the Fourier Transform method for implementing convolution to apply the York Minster acoustic to the musical signal. How long did the computation take?

(g) Use `wavwrite` to write out the resulting data as an audio file, and listen to it using a media player.

Other Resources:

- Spatial Audio Creative Engineering Network. `http://space-net.org.uk/`
- Stereo Room Impulse Responses, from Rebecca Stewart, Queen Mary, University of London; from University of London External System.
  `http://www.londonexternal.ac.uk/current_students/programme_resources/cis/news/docs/Stereo%20IRs.zip`