EECS 360 Signal and System Analysis
Lab 6 Matlab and Audio

1 Reading and introduction

Reverberation, in psychoacoustics and acoustics, is the persistence of sound after a sound is produced. A reverberation, or reverb, is created when a sound or signal is reflected causing a large number of reflections to build up and then decay as the sound is absorbed by the surfaces of objects in the space which could include furniture and people, and air. This is most noticeable when the sound source stops but the reflections continue, decreasing in amplitude, until they reach zero amplitude.

Convolution reverb is a process used for digitally simulating reverberation. It uses the mathematical convolution operation, a pre-recorded audio sample of the impulse response of the space being modeled, and the sound to be echoed, to produce the effect. The impulse-response recording is first stored in a digital signal-processing system. This is then convolved with the incoming audio signal to be processed[2]. A recommend reading is in the following link: http://people.eecs.ku.edu/ esp/doku/doku.php?id=aseepaper.

2 Exercise 6.1 Sampling

Sampling. In signal processing, sampling is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal)[1]. Sampling can be done for functions varying in space, time, or any other dimension, and similar results are obtained in two or more dimensions. For functions that vary with time, let $s(t)$ be a continuous function (or "signal") to be sampled, and let sampling be performed by measuring the value of the continuous function every T seconds, which is called the sampling interval. Then the sampled function is given by the sequence:

$$s(nT), \text{for integer values of } n. \tag{1}$$

The sampling frequency or sampling rate, F_s, is the average number of samples obtained in one second (samples per second), thus $F_s = 1/T$. Using what you learned in class plot a figure including:

- subplot out a sine wave function with sampling rate of 6283.2Hz, time duration of 0.01 seconds, frequency of 100Hz;
- subplot out a sine wave function with sampling rate of 6283.2Hz, time duration of 0.01 seconds, frequency of 1000Hz;
- subplot out a sine wave function with sampling rate of 6283.2Hz, time duration of 0.01 seconds, frequency of 2000Hz;

Are the sampling results still sine wave? Do you see some strange?

3 Exercise 6.2 Create your own music

Plot out your *my_song* with the new tone matrix in class with tones : *my_song*=[1 1 5 5 6 6 5 8 4 4 3 3 2 2 1]. Note if you plot out all the results of *my_song*, the plot will look very ugly. So try to plot every other 213 samples. How to plot out a signal every other 213 samples? This process is called signal extraction.

4 Exercise 6.3 Make your music more fantastic

Download the two impulse response of concert halls, for each impulse response : Subplot the impulse response; subplot your generated song(the song inside the concert hall). In your report, you should have two figures and each figure includes two subplots. Tell how fantastic convolution can make your make is. What is the difference between signals after and before processing?

Note: there are two channels in the impulse response signal, so you have to "process" your music with each of the channel and then combine them, i.e, your final music is a matrix, the first column of your final music is the processing result of your music with the first column of the impulse response; the second column is the processing result of your music with the second column of the impulse response.

5 Exercise 6.4 Bonus (+10 points)

Write a code that can play out whole a new music. The Matlab code and song files for this bonus exercise only need to be sent by email.

References

[1] https://en.wikipedia.org/wiki/Sampling_(signal_processing)

[2] https://en.wikipedia.org/wiki/Reverberation