Lecture 5: Audio Effects Processing

Introduction

As the title suggests, this lecture gave an extensive overview of various audio effects and principles of their operation. In particular, it covered delays, flanging and phasing effects, chorus, pitch shifting/time stretching, compression. Moreover, it covered more advanced effects and techniques such as guitar amplifier modelling. In this diary I will give a brief overview of the covered topics and provide an implementation example of a delay based flanging effect.

Overview

Delay

Delay is an audio effect where a delayed version of the signal is added to the original. This results in an output sounding like an echo. Originally, this effect was achieved physically using a tape delay machine. The signal would be recorded by a record head, thereafter the delay input would be outputted by the playback head and added back to the original signal resulting in the desired effect. Delay length can be changed by moving the delay handle. Nowadays, there are various digital delay implementations using digital filters such as digital echoplex model which uses single tap IIR filter as its feedback delay, etc.

Flanging and Phasing

Flanging is an effect achieved when an audio signal and its slowed down or slightly delayed copy are playing simultaneously. This creates a wooshing and sweeping sound effect which can also be observed in various places around us, such as hissing sound source passing us, sound of a fountain reflected from a staircase or a jet airplane flying over us. This effect has also been used extensively in various music genres, especially as a guitar effect. As a part of this diary a simple digital delay flanger is implemented where the delay time of the signal is varied using an LFO (low frequency oscillator) based on various signal shapes, e.g. square, sine and sawtooth. Obtained delayed signal is then added to the original to produce the flanging effect.

Chorus

Chorus endeavors to create a single source sound like multiple sources. It achieves this by adding an array of variously delayed versions of the original signal. As a result of the nature of its implementation, chorus architecture can be used to achieve various effects such as vibrato, flanger, doubling, echo and the chorus itself.

Dynamic Processing

Dynamic processing involves applying compression or expansion to the audio signal which results in reduced or expanded dynamic range of the output, respectively. Main components of the compressor are threshold, ratio and the knee. Threshold determines the signal level at which the compressor is activated, ratio controls how much gain reduction is applied with respect to the signal exceeding the threshold and knee controls the gain transition when the signal crosses the threshold. Compressor is used in wide range of applications such as radio, TV broadcasting, live PA systems, music, etc. In modern pop music it is used extensively as a "glue" when mixing the track. The activation of the compressor is also often controlled by another source. This type of compression is called sidechain compression and is extensively used in EDM music to achieve the "pumping" effect of the track - it is often tied to the kick drum and whenever the kick drum activates, the compression is applied to certain tracks which makes them duck.

Audio Effect Modelling

Block-Based Distortion Modelling

Block based distortion modelling is performed by measuring the filter using sine-sweep method and then the parameters of the nonlinearity are adjusted in order to minimize the error, i.e. we want them as close to our goal.

Neural Guitar Amplifier Modelling

Amplifier (and other effects) modelling using neural networks is a fairly new approach in the audio signal processing domains. It aims to use neural networks to approximate the characteristics of the desired effect. This is done by sampling of the desired output to create a training dataset. Thereafter, the neural network is trained iteratively using the dataset by comparing its output to the desired output and adjusting the parameters accordingly.

Conclusion

This lecture covered many common audio signal processing effects and as this area is ever evolving, I believe that in the future we will have access to even more various effects used both for practical and creative purposes. The area of modelling using deep learning looks especially promising as this approach hasn't been used before and could yield some quite interesting results.