Digital Design of Audio Signal Processing Using Time Delay

Mauricio Perez[1], Rodolfo Coelho de Souza[1,2*], Regis Alves Rossi Faria[1,3*]

[1] Escola de Comunicações e Artes, Universidade de São Paulo
Av. Prof. Lúcio Martins Rodrigues, 443 – 05508-020, São Paulo, SP

[2] Departamento de Música da FFCLRP, Universidade de São Paulo
Rua Maria M. C. Teles, s/n – 14040-900, Ribeirão Preto, SP

[3] Escola de Artes, Ciências e Humanidades, Universidade de São Paulo
Rua Arlindo Bettio, 1000 – 03828-000, São Paulo, SP

mperez@usp.br, rcoelho@usp.br, regis@usp.br

Abstract

This poster describes the design in PureData of some audio signals processes in real time like *delay*, *echo*, *reverb*, *chorus*, *flanger* e *phaser*. We analyze the technical characteristics of each process and the psychoacoustic effects produced by them in human perception and audio applications. A deeper comprehension of the consequences of sound processes based on delay lines helps the decision-making in professional audio applications such as the audio recording, mixing, besides music composition that employs sound effects in pre-processed or real-time.

1. Introduction

The technique of time delay is simple and versatile. It is often used in audio signal processing for fixing a large set of technical problems, e.g. problems of sound diffusion in concert halls, or it is applied to audio effects that expand the capabilities of acoustic instruments, modifying and creating new timbres for the purpose of music composition.

In this poster, we chose to explore this second trend. Notice that, from the psychoacoustic standpoint, effects based of time delay are related to how the human hearing apparatus receives and interprets the delayed signals. We may understand them as repetitions, some feature of the acoustic space or as the timbre that results from transformations in the spectral dominium.

The first uses in music of sound effects based on time delay goes back to the 1940's. They were delay effects and short echoes that used tape loops in magnetic sound recorders. The procedure was called *tape delays*. The amount of the time delay was ruled by the distance between the reading and recording heads of the devices. This loop arrangement might generate an echo effect that could apply one or many repetitions of the signal to be added to the original signal on another recording device. Until the decade of 1970's this was the basic configuration for an echo system.

As only the digital domain concerns us here, time delay effects can be implemented using a function called *digital delay line*. According to Roads (1996, p.433), this delay type consists in "a data structure called a *circular queue*" in which a list of memory locations, disposed sequentially in the computer's memory, stores the numerical representation of audio samples.

2. Implementation of delay lines in PureData

The design of delay lines using the software PureData (Pd) (Puckette, 2006) can be implemented using the objects [delwrite~] and [delread~]. The first object is responsible for creating the circular buffer, as cited above, containing the audio samples, whereas the second object reads and reproduces them. Both objects receive two arguments on the right side. The first argument of [delwrite~] is the location that stores the buffer and the second stores the time of the buffer.

Roads (1996) remind us about the difference between two types of delay lines: those that use a fixed time of delay and those that use a variable time of delay. The difference is that unities of fixed delay time do not change their time of delay while they process the sound. However, in a unity of variable time of delay, this time can be changed at any moment by varying the reading pointers at each sample period. These two types of delay are also inherent to specific temporal processes. The first case, of unities of fixed time of delay, we can found in the most common processes of delay as delay proper, echo and reverb when generated by delay lines. The second case is used in processes like *chorus*, *flanger* and *phaser*. Therefore, we will start demonstrating the implementation that uses lines of fixed delay time, followed by those that use variable delay lines.

3. Efects with fixed delay lines: *delay, echo* and *reverb*

According to Roads (1996, p.435) fixed delay lines can be arranged in three categories with specific interval times which correspond to three categories of perceptual effects to the human hearing. These three interval times are: short interval times (up to 10 ms), medium interval times (from 10 ms to 50 ms) and large interval times (larger than 50 ms). Short interval times are perceived mainly in the frequency domain as artifacts added to the

[*] Supported by CNPq

original signal. When a delay line operates between 0,1 and 10 ms, it generates a *comb filter* effect that can reinforce frequencies of the original signal. Medium interval times are perceived as an ambience created around the original sound. This means that the signal is amplified by the sum of the original signal with other signals generated by the delay line. Therefore, the loudness of the original signal seems to increase. Finally, delay lines with large delay times generate a perception of sequential repetitions of the original signal. They correspond to larger spaces with more and distinct reflections.

We will deal now with the design of many delay lines to create the effects of *echo* and *reverb*. These effects can be created with an algorithm that generates a mechanism of feedback of the original signal into the unity of delay processing.

This way, with specific combinations between delay times and feedback gains we may implement different time processes based on fixed delay lines, as those described above, and as shown on Table 1.

Effect	*delay time*	*feedback gain*
comb filter	1-10 ms	0,9
loudness boost	10 - 50 ms	0,5 - 0,3
short echo	50 ms	0,7
large echo	100 ms	0,7 - 0,95
short reverb	100 ms	0,3
large reverb	150 ms	0,5

Table 1: Effects based on delay time and feedback gain

The sonograms of Figures 1A to 1C, produced with the software Spek with audio samples in '.wav' format, mono/44100/32 bits, represent the spectral analysis of different processes applied to a sound sample according to the parameters of Table 1. They allow us to visualize the differences between these processes.

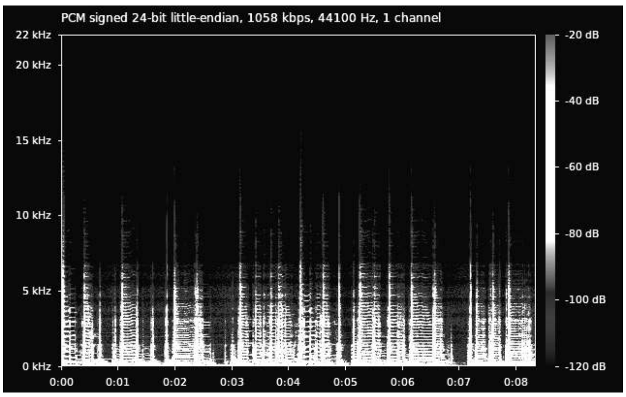

Figure 1A: Original sound

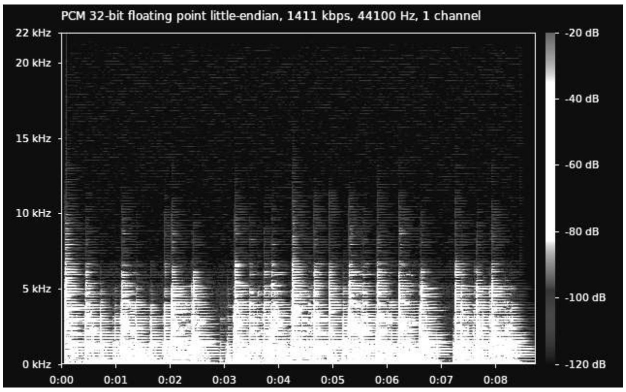

Figure 1B: Comb filter

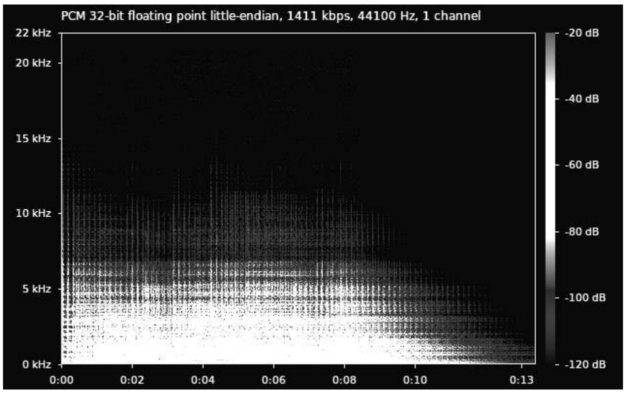

Figure 1C: Large echo

4. Effects with variable delay lines: *chorus*, *flanger* and *phaser*

As mentioned earlier, variable delay lines allow the change of the delay time while the audio signal is processed by the delay unity. This allows the creation of other effects based in delay lines: *chorus*, *flanger* and *phaser.* When we say that a delay line is variable, we are describing a unity of signal processing that has some element that varies constantly. In this case, what varies is the duration of the delay time. It can oscillate between a maximum and a minimum value. A low frequency oscillator (LFO) can implement this kind of effect. The LFO is used to control the delay time. Figure 2 shows an implementation of this model in Pure Data.

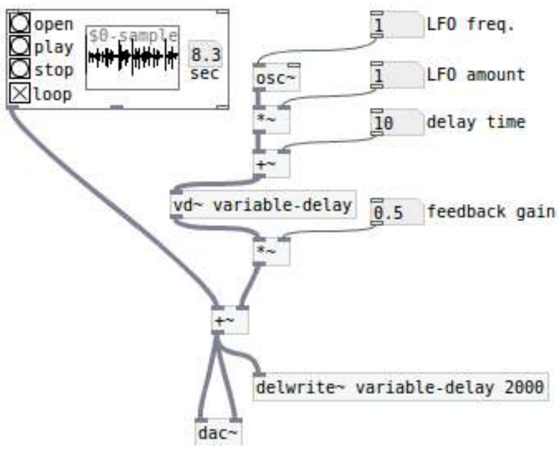

Figure 2: Model of variable delay with feedback

Based in this model of variable delay we can implement some kinds of effects. These effects have in common that same variable delay unity. However, each of them has some special features. One of these features is related to the variation of the delay time. In the case of *chorus*, for instance, this variation has to be set between 10 and 30 ms. In the case of *flanger*, the variation can occur between 1 and 20 ms, and in the case of *phaser* the LFO may vary from 1 to 10 ms.

In the sonograms in Figures 3A and 3B, we may visualize – mainly looking at the spectral content – the different patterns produced in the resulting sound signal by each of these different processes of time delay using variable delay lines. A clear distinction between these processes and the processes with fixed delay lines is that their main characteristic concerns changes in the temporal/morphological domain of the sound signal, except maybe in relation to the *comb filter* effect.

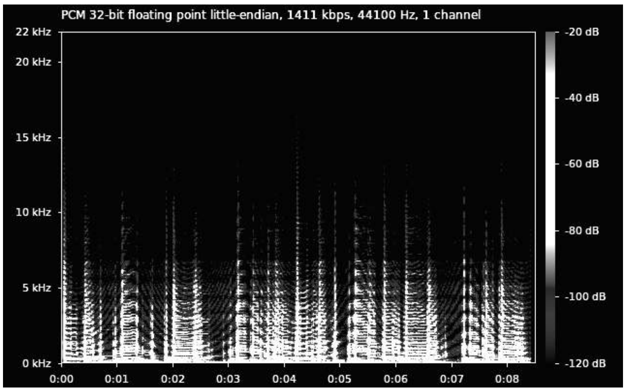

Figure 3A: Phaser

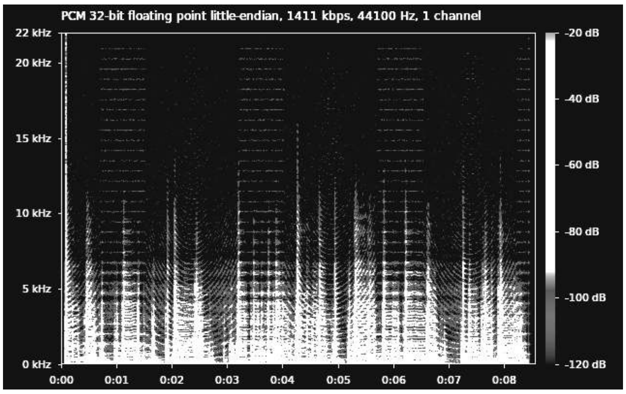

Figure 3B: Flanger

5. Conclusions

The understanding of the different types of signal processing, generated by fixed or variable delay lines, enhances the decision making process in situations when we face professional audio problems, from technical or aesthetical points of view. This can happen in a simple sound recording session, in an audio mixing station, or during the composition of an electroacoustic music that employs pre-processed or real-time sound effects.

It can be quite useful being able to differentiate between the results of processes such as *comb filter*, *phaser* and *chorus*. These effects result from delays that generate time displacement between their repetitions. This causes changes in the harmonic spectrum. The awareness of time elements can be useful in many situations, for instance in room reverberation, identification of obstacles in sound trajectory or recognizing phase cancelation in a recorded sound.

Therefore, in these cases, acknowledging the time prevalence in the situation can help the decision making to avoid attention only to the frequency domain, what can lead the sound engineer to use, for instance, a frequency filter to change the spectrum involuntarily. Indeed, we may create desirable effects with other strategies, like setting a short time displacement between similar tracks, as we use to do in voice or instrumental unison doubling. The result is an enlarged sound ambience and reinforcement of harmonic partials produced by constructive interference.

Similarly, we must be aware of changes in the spectrum domain generated by basic effects as reverbs and simple delays, as generated by delay lines, because they may be desirable or not. The superposition of repeated sound materials usually produces reinforcement of certain frequencies, generating an effect aesthetically desirable or just distortion. Our experience tells that this situation can happen when recording in a room with large reverberation. For the performer the sound seems nice but the signal for microphone caption can already be saturated at the source.

These experiments also demonstrate that the human perception of delays shows a double standard. Delays larger than 50 ms are interpreted by our brain as isolated repetitions while shorter delays just change certain frequency components of a single sound event.

What has been presented in this paper was a reasoning for the implementation of many types of signal processing with time delays. The kind of reasoning used was based in the creation of delay lines by digital means. This way we emphasize that there is not a single way to implement these processes, even digitally or using individual delay lines without feedback. The implementation with delay lines, in a digital environment, helped didactically the understanding of how are produced and how we perceive the effects based on time delays and that use less resources of digital processing.

References

Mauricio di Berardino. *Muselectron: Tutorial su PureData, Max/MSP e Axoloti.* Available at: https://muselectron.blogspot.com.br/p/pure-data.html Accessed in April of 2019.

Miller S. Puckette. *The Theory and Technique of Electronic Music.* World Scientific Publishing. 2006.

Jean-Michel Réveillac. Musical Sound Effects: Analog and Digital Sound Processing. Iste Editions, 2018.

Curtis Roads. *The Computer Music Tutorial.* The MIT Press, 1996.