DIGITAL AUDIO SYSTEMS REVIEW

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ABSTRACT

This piece of work will explore the natural phenomena of Reverberation and its recreation within Digital Signal Processing. The history of refabricating acoustic spaces through analogue and mechanical means will be examined; as well as a thorough dissemination of the physical and mathematical components of reverberation.

Spatial audio will also be discussed in terms of musical utilization of moving reverb to create a sense of change in stereo width.

With Digital signal processing has come major advancements in synthesizing reverberation effects as well as emulating not only mechanical reverberations but also actual acoustic spaces with the application of convolution. Automation (specifically panning automation) has made it easier and much more prevalent to create stereo effects in modern music and would not be possible without the advancements in DSP.

1. BACKGROUND

1.1. History

Reverberation within acoustic spaces has been sonically shaping music since the Gregorian period. The long decay of sound, induced by the large reflective churches they were produced in, helped create Gregorian chant. The Large reverb time smeared pitch and gave the vocalists an ethereal, larger than life quality. In almost all aspects of music since this period, Reverberation has been just as relevant as the instrumentation and orchestration. As technology advanced so did the emulations of reverberation. First, analogue efforts were achieved by routing dry audio from loudspeakers into a reverberant echo chamber, recording the output with a specifically placed microphone. Then came mechanical inventions; Spring and Plate Reverberation, which utilize metal transducers whereupon vibrations are created through electromagnetism and later picked up and added to the dry audio signal

1.2. Psychoacoustics

The way in which humans perceive sound is remarkable. The ability to discern between varying amplitude, frequency and timbre make our ability to hear one of our most vital senses. In reverberant situations our hearing process goes through a number of complex procedures. In connection with our outer ear anatomy (pinna), our brain can localize sound sources. Although we can't picture a room with just the sounds within them we can certainly make educated assumptions about the size by the magnitude reverberation time and the tone colour of the space.

1.3. Scientific

Reverb can be broken down into two portions: Early Reflections and Reverb Tail.

The Early Reflections happen milliseconds after we hear the direct sound due to the fact that the path to the listener is longer. Early Reflections can be broken up into orders based on the time they reach the listener due to longer paths around the rooms. The Early Reflections provide the timbre of the reverb and are directly correlated to the dimensions of the room.

Pre delay is the time difference between the direct sound and the beginning of the reverb tail. The Reverb Time, or RT60 (the time it takes the reflected sounds to decay 60dB below the direct sound) Pre Delay and Reverb Time are not only affected by the dimensions but also the absorption and reflective qualities of the room.

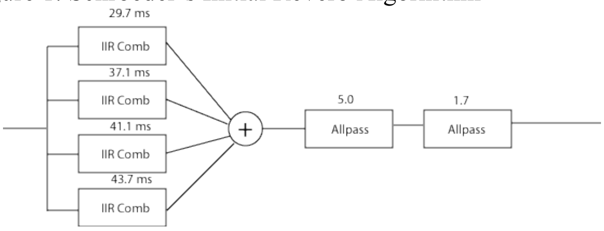
2. EARLY HISTORY OF DIGITAL REVERB

The First Digital Reverberations were formed with algorithms. Initially attempting to recreate/synthesize acoustic spaces as well as analogue and mechanical devices. It is questionable to whether the acoustic space recreation was adequate but these algorithms are still used in music production and devices such as the Lexicon are still studio standards.

2.1 Shroeder

Initial Reverberation algorithms were executed by a German physicist Manfred Shroeder. **[1]** He developed an algorithm to create concert hall reverberation digitally. It employed the use of Infinite Impulse Response (IIR) comb filters, because of their property of exponential decay, and allpass filters to create a sufficient density of reflections needed for realistic reverberation (around 1000 per second). The purpose of the comb filters is to control the reverb time, and the purpose of the allpass filters is to control the intensity of echo diffusion. The downside to Shroeders algorithm were that it didn't have a flat frequency response (Metallic sounding) on the input to ouput signal nor was it a realistic sounding reverb as it was ‘colourless’.

Figure 1. Schroeder’s Initial Reverb Algorhithm



2.2 Kendall and Martens

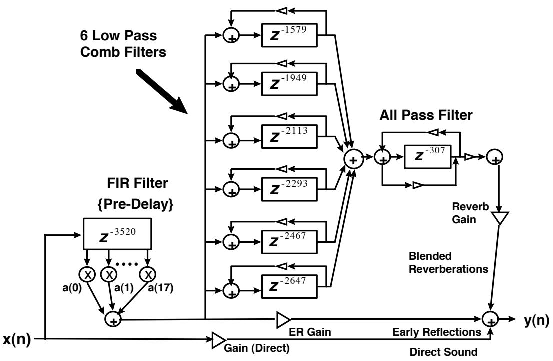
Kendal and Martens made a better real world approximation by examining Early Reflections in the digital realm.

[2] “We Plan to incorporate reverberation by generating an early reflection (echo) off the floor, spatialized to the same direction as the incident source. By varying the ratio of direct to indirect sound, we expect to strengthen distance cues. This is a first step to strengthening our acoustic room model. “

2.3 Moorer

[3] James Moorer then published an optimized version of the allpass delay, which used less multiplies, utilizing the early reflections.

Figure 2. James Moorer’s Algorithm

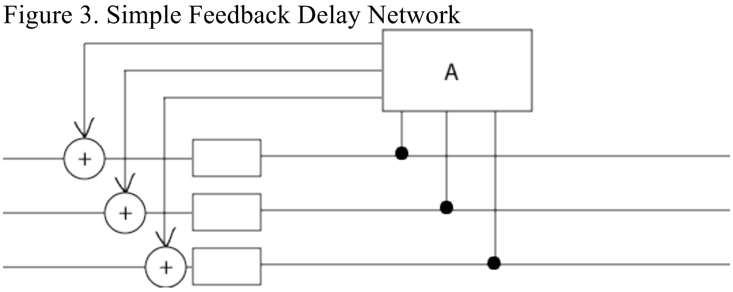


3. FEEDBACK DELAY NETWORKS

3.1 Puckette and Stautner

Reverberators utilizing Feedback Delay networks were pioneered by Puckette and Stautner. **[4]** They offered many advantages greatly in the spatial output of the system as well as explored the use of multiple arbitrary channel (all with decreased computational efficiency, when signal is dormant).

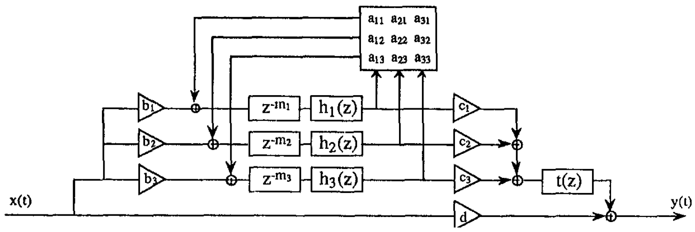
Figure 3. Simple Feedback Delay Network



3.2 Jean Marc Jot

In 1982 Jean Marc Jot’s expanded delay network with absorbtion filters h(n)(Z) and tone corrrector t(z), as he stated were necessary to ‘avoid unpleasant resonances in the response to short transients.’ **[5]**

Figure 4. Jot’s expanded delay network



4. DIGITAL SIGNAL PROCESSING BASICS

4.1 Impulse Response

The Impulse response of a system is a description of the system on an output. In a liner time invariant system (LTI) the impulse response completely characterizes the system. To understand the mathematics behind an impulse response of a system it is important to understand the Dirac Delta function. The Delta Function $\partial[n]$ is a single sample unit impulse with a value of 1 at the first sample and all other values equal to zero.

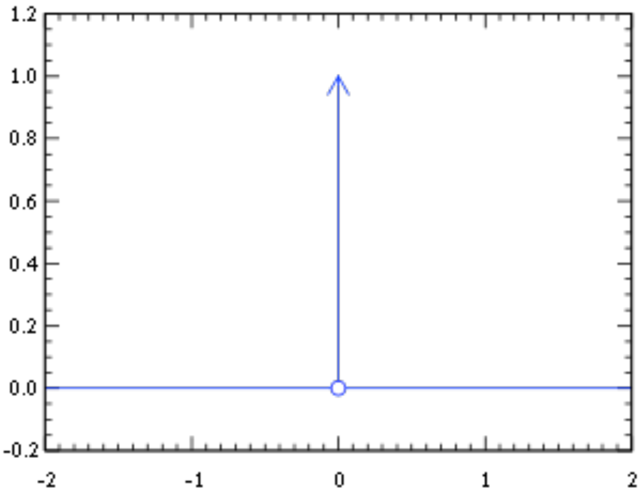


Figure 5. Delta Function

An ideal impulse response would provide equal energy across the frequency spectrum. This would portray a more accurate modeling of the acoustic space across all frequencies.

Physically collection an impulse response of a acoustic space include:
Loudspeaker Microphone setups
Sharp Transient : clap burst of white noise
Sine Wave sweep

4.1 Convolution

Acoustic modeled Convolution Reverb is a relatively new digital processing tool. Impulse responses taken of an acoustic space can be convolved with an impulse signal to take on the sonic characteristic of that space. There are still inherent weaknesses mostly due to FIR being relative to the static placement of the recording equipment used to take it. The whole process of convolution is also quite cpu intensive. That being said it is still the most accurate, natural and real sound reverberation available in the digital world.

Convolution is the 'multiplication' of an input signal by an impulse response (summed, sample by sample) resulting in an output.

$x[n] * h[n] = y[n]$

$$x[n] * h[n] = y[n]$$

Figure 6. Impulse response h[n] of Starter pistol fired in Humprecht castle plotted in Matlab

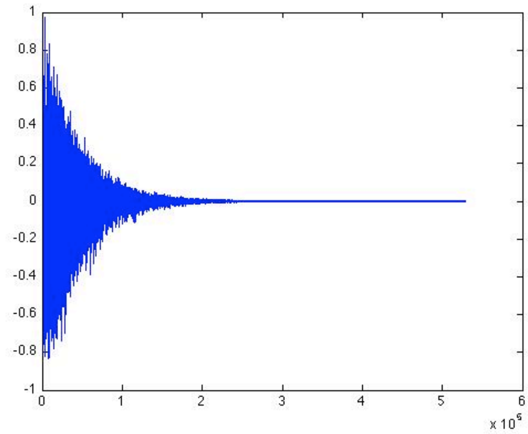
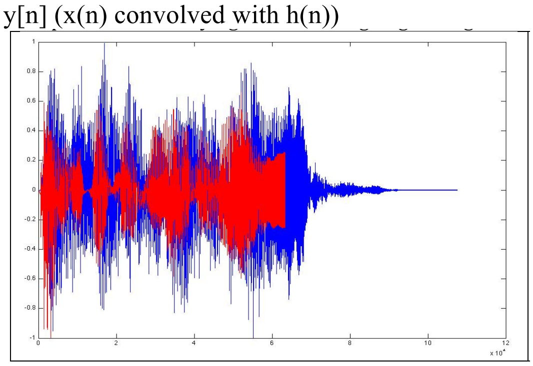
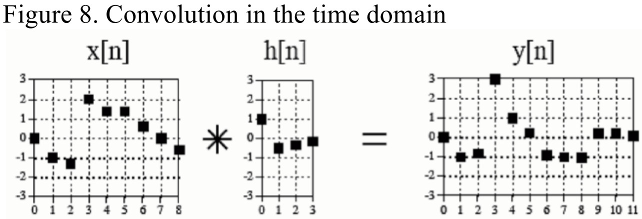


Figure 7. Red input signal x[n] and blue is output signal y[n] (x(n) convolved with h(n))



Direct convolution can be done in the time domain by storing each sample of the impulse response as a coefficient of an IR filter. Direct convolution becomes easily impractical if the length of the target response exceeds small fractions of a second.

Figure 8. Convolution in the time domain



Convolution can also be done in the frequency domain with the latter being significantly faster. Given the Fast Fourier Transform (FFT) of an impulse response as well as a discrete FFT of an input signal, a convolution can be made by multiplying, sample by sample, and then transforming back into the time domain. The drawbacks of this are that it requires a lot of CPU power, but then again with the advancements in cpu threading this shouldn't be a problem.

Panning Automation (Stereo Width) With Convolution Reverb Summed with Dry Mono Signal

5. AIMS

To create a more musical utilization of convolution reverb by combining the spatial audio effect of panning.

Psychoacoustically, our ears are more perceptive to change, and in a musical setting this is vital in creating tension and release.

By employing a sense of movement to the reverb a greater sense of width and thus change is achieved. This can be achieved by utilizing two separate reverbs on both channels and pre delaying one of the channels. Another way is to reverse one of the wet signals and sending it to one channel with the original wet sent to the other.

The most controllable method is to actually automatically pan the reverb across the stereo field.

6. METHODS

Can be broken up into four parts.

1. Original mono Signal and Impulse Response
2. Convolution of Original Signal and Impulse response to give %100 wet signal
3. Panning effect on wet signal (3 effects will be examined):
   a. one mono reverb signal that that pans left to right at a given rate of samples and
   b. Split mono signal that starts fully panned left and right and pan towards the centre for the full duration of the wet input signal
   c. The reverse of b
4. Combination of both original dry mono signal and panning wet signal in assignable amplitude ratios.

7. RESULTS

Although there are already a vast array or digital reverbs on the market with a vast array of assignable variables including stereophony controls, experimentation with convolution reverb in a musical context seems to be overlooked.

Convolution reverb code is simple and has been experimented with and discussed in my Lab report 1.

Successful utilization of a panning script ('panning.m' attained from Vinay Kumar Tadepalli on the matlab central file exchange) on the wav written wet signal has yielded workable results. Sufficient tweaking Summing of Both signals is yet to be completed as well as an incorporating script that performs all of these functions in one.

(Will be completed by Lab Report 2)

8. DISCUSSION

Issues will arise in the summing of both wet and dry signals. The durations have to be matched and the rate of panning must be related to this duration and I've yet to discern how to achieve this at present.

The decay or Rt60 time cannot be varied in my proposed code (unless a different impulse response is used) but it can be measured with the Hanning Transfer. The Hilbert transform computes the discrete time analytical signal of the impulse response.

$$Y = \mathrm{Re}\{X\} + j \cdot X\sim$$

where $X\sim$ is the Hilbert transform of the vector

$\mathrm{Re}\{X\}$.

It would be interesting to employ a variable of amplitude ratio (wet:dry summing signal). This is probably the most important variable in terms of musical usage.

There are many variables that can be explored and code written for, in the lead up to my second Lab report, but time restraint and inexpierience with Matlab will be the deciding factor.

9. CONCLUSION

Presently, Convolution reverb doesn't really present itself as a forerunner in music production. It has however, proven itself in post production, (film and television) where the recreation of real spaces is pivotal in creating a believable final product (see plug ins IR-1, Altiverb and TL spaces).

With the ongoing change in style of music production from the big reverb overproduction in the 80's to the raw nature of the stripped back 90's it is impossible to tell what will be a trend in the future.

Thus it is important to keep experimenting with effects (such as panning automation of convolution reverb) as this is what will shape the sounds and music of tomorrow as it has throughout history.

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Further Reading

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