Digital Audio Signal Processing Assignment 1 Report

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# Introduction and Objectives

The purpose of this assignment is to produce a digital artificial reverb algorithm based on designs from Schroeder (1962) and Moorer (1979). This means modelling a section of “Early Reflections” based on room size and reflectiveness. The use of IIR Comb Filters to emulate room modes combined with all pass filters to increase the echo density from comb filters to produce an adequately dense and diffuse reverb.

Parameter control within the project intends to be intuitive and flexible, allowing the user control over more abstract parameters such as Reverb Time (RT60), Controls such as “spread” or “Reverb Time” allow the user to more easily manipulate the reverb signal than directly manipulating the delay time of a filter or its gain coefficient.

This report outlines the use of comb filters, all pass filters and early reflection networks (based on FIR filters) to produce the Moorer reverb, discusses the operation of the author’s reverb and how this relates to the filters used, displays the filter designs and their analysis, details results of the simulation, and concludes by evaluating the results of the exercise for improvements and further developments.

# Background / Literature Review

## Early Reflections

Moorer (1979) disucsses the use of an FIR filter to mimic the “Early Reflections” of a room. Early Reflections can be defined as the non-direct sounds arriving within 80ms of a direct sound. An array of b coefficients the same length as the maximum delay with a non-zero value at each delay point would create multiple taps of the audio. Howard and Angus (2017, p.289-293) discuss the effect of the absorptive nature of boundaries and attenuation over distance in air. These factors are important for determining total attenuation based on the room size and type.

## Late Reflections

To make parameter control more natural for the user, it is essential to translate terms like “a coefficients” or “b coefficients” that are used in the filters designing reverb into more intuitive terms. It was decided to give the user control over RT60 (labelled Reverb Time), which is the time taken for reverb level to reduce to -60dB of the input signal (Sabine, 1973). Thus, it is necessary to understand how the all pass and comb filter networks create an adequate reverb of given length. Schroeder (1962) states that an echo density of 1000 echoes per second is required for a flutter free reverberation. The echo density of a comb reverb can be found as follows:

where delay is in seconds.

Figure Echo density. Schroeder (1962)

The echo density of a system of parallel combs is equal to the summation of their individual echo densities. Schroeder (1962) recommends comb filter delay ratios being in the range 1:1.5. The delay times should be mutually prime to discourage overlap of reflections and reduce colouration. Pirkle (2013 pg 366) states that the value of gain applied to a comb filter determines the pole radii. Matching pole radii ensures matching decay rates, ensuring no colouration. The gain, then, can be calculated based on the shortest delay of the combs as follows:

where delay is in seconds.

Figure 2 comb filter gain. Pirkle (2013 p 366)

Schroeder’s desired modal density of 0.15 eigenfrequencies/Hz allows the echo density of the system to be calculated. Rearranging Pirkle (2013 p368) shows echo density to be as follows:

Figure 3 echo density of combs. Pirkle (2013 p 368)

In the case of the example given in Schroeder (1962), the echo density of four comb filters with delays or around 0.04s gives an echo density of 100. This is a factor of ten short of the Schroeder’s desired echo density of 1000, therefore, all pass filters are used to multiply this up to the required echo density. From Schroder’s choices of 0.005s and 0.0017s of delays to give the filters a multiplying factor of ~3, a “delay constant” from:

where n = number of all pass filters.

Figure 4 echo density of combs. Pirkle (2013 p 368)

This allows a single control, RT60 to set the gains of the comb filters when their ms delay is based on a fixed modal density and number of combs. The all pass filters are then calculated accordingly.

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## Signal Flow

The signal flow outlined in Fenton (2018) shows that the dry signal and the early reflections network are fed into the late reflection network. To blend the signal effectively, the output of the late reflections network is summed with the early reflection network to form a “wet” signal which is blended with the “dry” input signal.

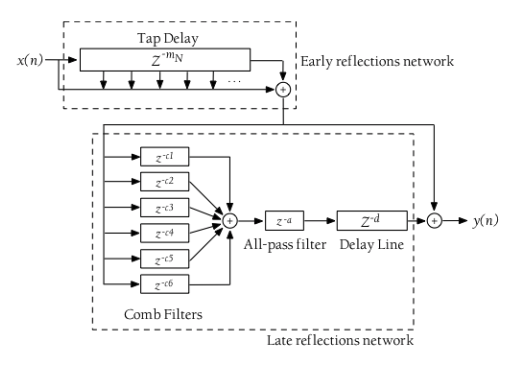


Figure 5 Algorithmic reverb block diagram

# Theory of Operation

## Early Reflection Network

Moorer (1979) uses an FIR network with b coefficient taps at points along the delay line. Implementing this method meant using a “dot multiplication” of a delay line against a b coefficient array, summing the results to produce one sample per loop, requiring a loop around the entire input audio file to produce an early reflection network containing all reflections. This is a costly method, requiring as many multiplications as the sample delay per sample for an entire audio file. A much more time efficient method, used in submitted code, is to create a new audio buffer per reflection, padding the start with 0s of the desired length and then attaching the audio file. These multiple reflections then allow specific filtering based on their distance, per reflection, simulating the high frequency damping effects of air. These individual reflections are given a gain multiplier relative to their distance, mimicking the attenuation effects of distance, and summed to create an early reflection network. Another advantage of using this system is that the direct signal is completely removed. Using the Matlab filter function will always include the direct sound due to a necessary 1 b coefficient in the first element of the array.

The delay lengths in the system are calculated based on the user parameters “room size”, “latest ER”, and “number ERs”. Room size determines the width of the room, with height and length calculated proportionally with height:width:length ratios of 0.62:1:1.62 (Radiobomb, 2012). The source and virtual “microphone” are assumed to be at a position 1/3rd of the way along each wall. The shortest distance from the source to the boundary is then multiplied by two to calculate the time of the first reflection and an array is created of length number ERs up to the time of the latest ER to calculate the timing of each early reflection. The Early Reflections network can be switched on or off.

## Comb Filters

The comb filters used in the late reflections network are IIR filters. The delay time of the combs are calculated based on the echo density of the entire comb system (based on Schroeder’s fixed 0.15 modal density value of eigenfrequencies). This is then divided by the number of combs to calculate the echo density of a single comb. This is then expanded up using Schroeder’s recommended 1:1.5 ratio for comb delay values. The calculated values are then fed to a Matlab function to find the next prime number with regards their sample delay values to ensure there is no overlap of comb pulses. The gain value for the entire comb network is calculated from the shortest delayed comb as per figure 2. This is duplicated to create a stereo effect, with every other comb having an inverse gain value in the right channel as shown in Pirkle (2013, p376).

The comb filters, in this system, are placed after the all pass filters as their transfer functions multiply in the z domain (Pirkle, 2013, p.376).

## All Pass Filters

The all pass filter gains are set by the user parameter “Spread” and the delay times are calculated based on the echo density from the combs which need multiplying to achieve the desired echo density proposed by Schroeder (1962) of 1000. These all passes are scaled by +/- 10% with the smaller filter first for a larger echo density. The multiplying factor is calculated based on the 40ms reverb example given in Schroeder (1962). The RT60 is then achieved by gain manipulation. The all pass filtering is achieved manually within Matlab following Schroeder’s block diagram. This allows the all pass filtering effect to be achieved faster than with the filter function. This can be found in the file AllPassFilter.m

## Delay Input for Late Reflections

A single IIR filter is used to create a musical delay with a delay time calculated by a tempo input. The output is then summed with the input to the late reflections network, producing a soft rhythmic pulse applied only present in the late reflections. The filter delay is determined by the tempo variable and the note length (determined by division, isTripleted, and isDotted) for the delay system. The feedback gain is used to determine the decay of the filter. The overall gain is used to blend the delayed signal into the late reflections network. There is also an option to filter the feedback gain of the network with a variable cutoff frequency.

## Chorus

A chorus effect is can be added to late reflection network producing a modulated pitch and timing effect based on a random phase offset generated within a given chorus “depth”. Random number generators create a series of offset values which are filtered to reduce high frequency content. These random offsets produce a random detune/timing effect relative to the source audio. Mutltiple voices of these random offsets are created and summed. The resultant output is then low pass filtered to reduce any unpleasant extra high frequency content.

## Error Checking

Delay checking has been implemented for every user control variable. This ensures that the filters remain stable and room sizes and reverb times remain within reasonable boundaries.

# Filter Design and Analysis

# Results and Conclusions

Critique – ER network isn’t great. ER attenuation calculation could be improved by ACCURATE DISTANCE MODELLING. Use of 80ms as max would have been better and then could accurately calculate delay positions. Better use of room size and wall number. Could base geometry on regular shapes?

Chorus – doesn’t sound great. The random offsets probably aren’t calculated very well and the extra HF content is probably a result of that. NB- this effect is reduced with shorter delay times

Does it work with transients??

# References

Fenton, S. (2018). *Digital Reverb Study.*[PowerPoint Slides]. Retrieved from https://brightspace.hud.ac.uk/d2l/le/content/9674/viewContent/75209/View.

Pirkle, W. C. (2013;). *Designing audio effect plug-ins in C++ with digital audio signal processing theory*. New York: Focal Press. doi:10.4324/9780203573310

Howard, D. M., & Angus, J. (2017). *Acoustics and psychoacoustics* (Fifth ed.). Abingdon, Oxon;New York, NY;: Routledge.

Radiobomb. (2012). *Sound 101: The “Golden Acoustic Ratio”*. Retrieved from https://radiobombfm.wordpress.com/2012/09/15/sound-101-the-golden-acoustic-ratio/.

* Schroeder, M. R. (1962). Natural sounding artificial reverberation.*Journal of the Audio Engineering Society, 10*(3), 219-223.
* Moorer, J. A. (1979). About this reverberation business.*Computer Music Journal, 3*(2), 13-28. doi:10.2307/3680280