Schroeder's Reverberator:

The Earliest Digital Solution of Sound Reverberation

EECS 195 Final Project

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March 21, 2005

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Introduction:

Reverberation is a naturally occurring acoustical effect (Roads 472). It is the result of the many reflections of a sound that occur in a room. A sound will be reflected, if it is not absorbed or transmitted when it strikes a surface. For example, when you listen to music from the stereo system in your living room, besides a direct path from the speaker to our ears, there are many other ways the sound travels through to reach us. As shown in Figure 1, Sound waves can also take a longer path by reflecting off wall before arriving at your ears. Since the sound waves travel a slightly longer distance, the reflected sound wave will reach our ears a little later than the direct sound wave. The sound waves amplitude will be a little weaker, since the wall surfaces will absorb some of the sound energy. Depending on the size of the room, the time and amplitude differences varies. That is why we can tell the reflection effect in concert halls easier than in living room. In a big enough room, the reflection will repeat many times, and then series of delayed and attenuated sound waves, which is called echo, reaches our ears. That is how we feel the 'spaciousness' of a room (Sahdev 5). With respect to music, reverberation is the collection of all the reflected sounds (Lawson 1). The modern technology provides many ways to solve this reverberation problem. Since digital processing devices and computer technology have been developed in recent years, people are able to finish a complicate simulation to find digital solution of naturally producing reverberation. The earliest digital solution of this reverberation problem is established by Manfred Schroeder of Bell Telephone Laboratories in 1961(Schroeder). The solution is called Schroeder’s Reverberator, which is this paper’s main focus and will be explained more in detail in the following section.

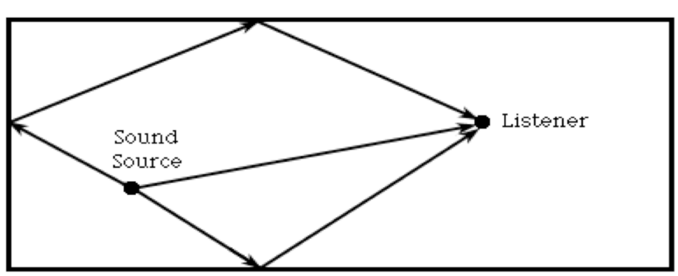

Figure 1.Sound Path (Sahdev)

Artificial Reverberation

Before the digital reverberator is established, there are some other ways for people to deal with reflected sound wave. The reverberation effect is one of the most important considerations for constructing theater, cinema, and concert hall. People study sound's nature and reproduce good reverberation effects by designing placement angle of speaker, building rough wall surface and specifically wall shape. A notable and useful technique is the echo chamber. Echo chamber is a small room with walls covered in sound reflecting material. A loudspeaker and a high-quality microphone are placed at different ends. The sound to be reverberated is played over loudspeaker and picked up by microphone. When all these condition are sympathetic, an excellent reverberated sound will be created (Roads 473). However, these units tend to have very uneven frequency response, falling off sharply at high frequencies, with the result that the sound is characteristically colored or blurred. As well, the echo density (i.e. the number of reflected repetitions per second) is often not high enough to avoid a 'fluttering' of the sound, particularly with very short percussive sounds (Smith 58). Because of the short points of artificial reverberation, digital reverberator plays more important roles in audio signal processing filed.

Digital Reverberation

1. Digital View of Sounds

As mentioned in the instruction section, Dr. Schroeder investigated the first digital reverberator and developed digital reverberation model in the 1960's.For studying the digital reverberator's work function, the nature of sound reflection should be known first. The Figure.2 shows the impulse response of a room. In DSP (digital signal processing) view, the reverberation effect can be broken into three parts (Roads 477):

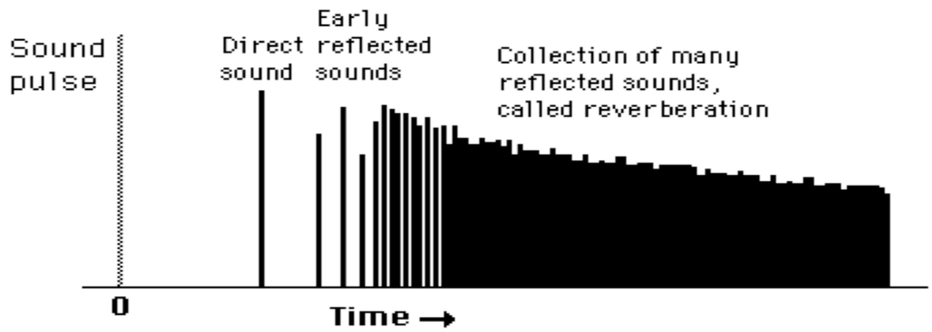

Figure 2: Parts of Reverberation (Lawson)

- Direct sound travels in straight path and is the first sound wave to arrive at listener's ears.
- Discrete early reflections hit the listener's ears just after the direct sound within 80 ms
- Fused reverberation contains thousands of closely spaced echoes but takes more than 80 ms to build up and then fade away

The mission of digital reverberator is to try to delay wave at certain frequencies, that will make sound wave arrive at the listener's ear at different time. Then, the reverberation effect is achieved.

2. Schroeder Reverberator

Schroeder Reverberator is a much simpler reverberator when compared with the later model and today's commercial reverberator. It only contains two components, recursive comb filter and allpass filter.

a. Recursive comb filter

y[n] = a * x[n] + b * x[n - delay] + c * y[n - delay]

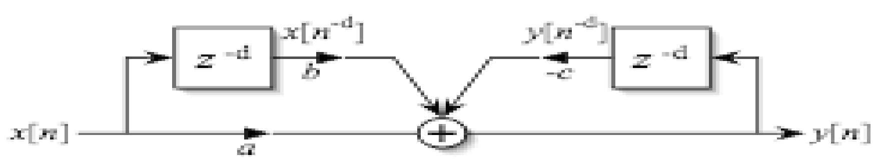

Figure. 3 Recursive Comb Filter (MSP)

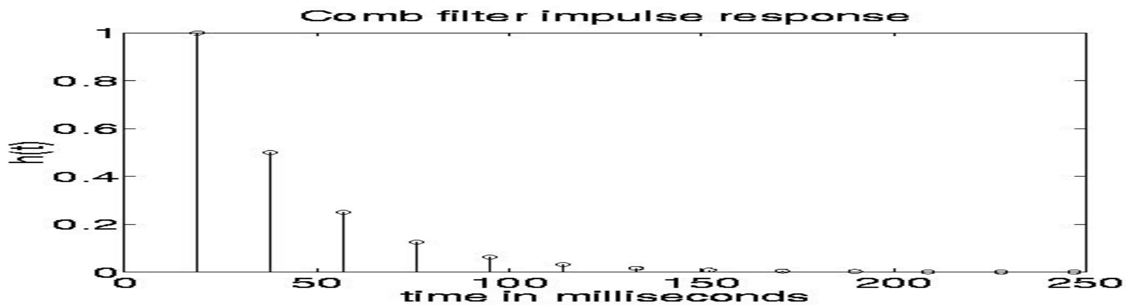

Figure 4: Comb Filter Output (Lawson)

Comb filter provides exponentially decaying impulse sequence, which can be thought of as simulating a single room mode. The comb filter shown above is a special case of an Infinite Impulse Response (IIR) digital filter, because there is feedback from the delayed output to the input. The feedback comb filter can be regarded as a computational physical model of a series of echoes. The echoes are exponentially decaying and uniformly spaced in time (Smith 226). The advantage is that the decay time can be used to define the gain of the feedback loop.

b. Allpass filter

$$y[n] = (-g * x[n]) + x[n - d] + (g * y[n - d])$$

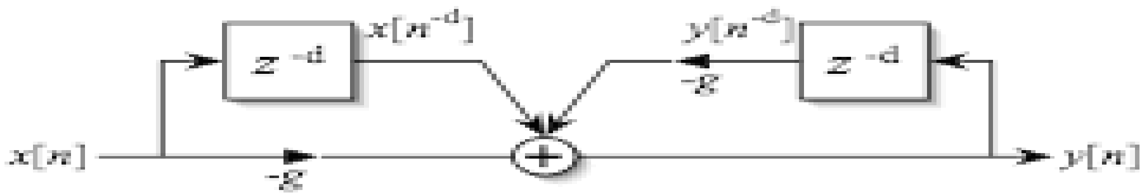

Figure 5: Allpass Filter (MSP)

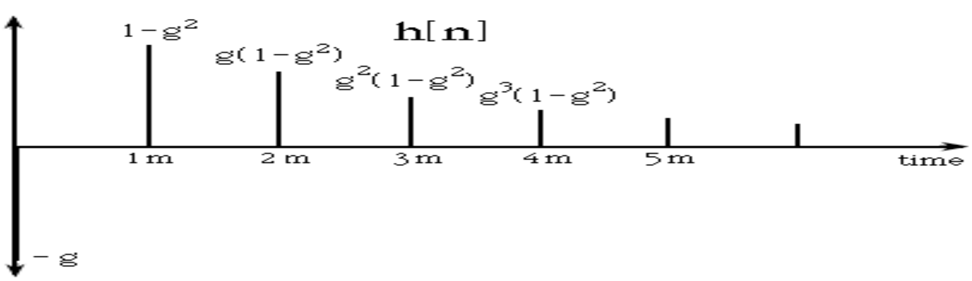

Figure 6: Allpass Filter Output (Lawson)

The allpass filter is an important building block for digital audio signal processing systems. It is called ``allpass" because all it transmits all frequencies of steady state signal equally well (Roads 478). In other words, the amplitude response of an allpass filter is 1 at each frequency, while the phase response , which determines the delay versus frequency, can be arbitrary (Smith 439).

C. Patches

Schroeder's original design is to connect 5 allpass filter in series. The feedback intensity is amplified which means that the echo density is enlarged. “ Each allpass can be thought of as expanding each nonzero input sample from the previous stage into an entire infinite allpass impulse response. For this reason, Schroeder allpass sections are sometimes called impulse expanders or impulse diffusers. His suggestion of using of allpass filters is especially brilliant because there is no such nature phenomena (Smith 430) However, in another word, this kind of reverberation doesn’t exist in nature, and the design is only archived in the simulated world.

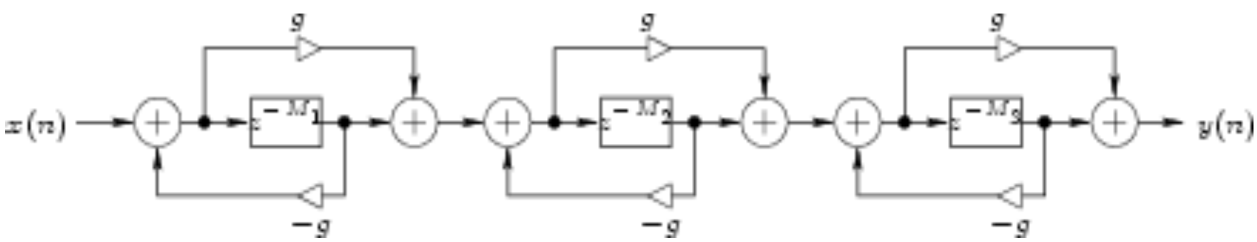

Figure.7 Schroeder's allpass design (Smith)

So, later on, Schroeder put comb filter into his new design, and got better performance. This design is recognized as Schroeder Reverberator. Figure. 8 shows its patches, and Figure.9 and Figure.10 shows its in more detail.

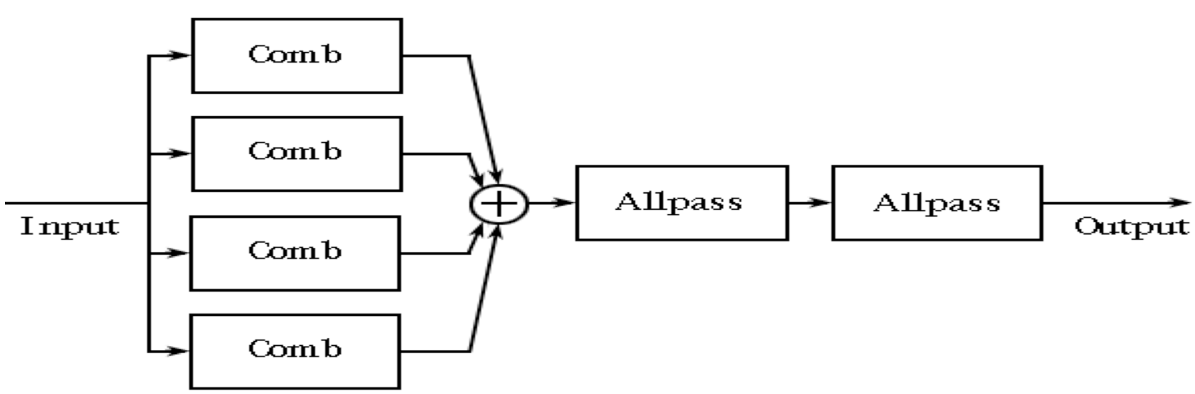

Figure 8 : Schroeder’s Reverberator(Lawson)

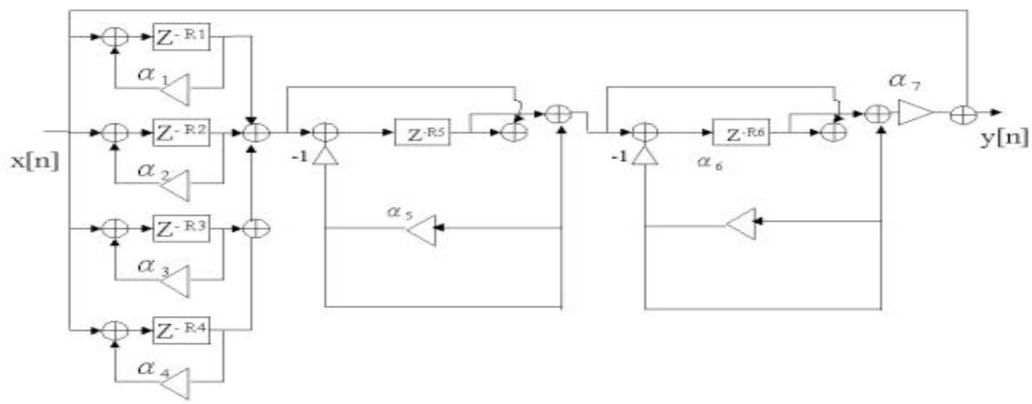

Figure.9 Schroeder reverberator in detail (Shadev)

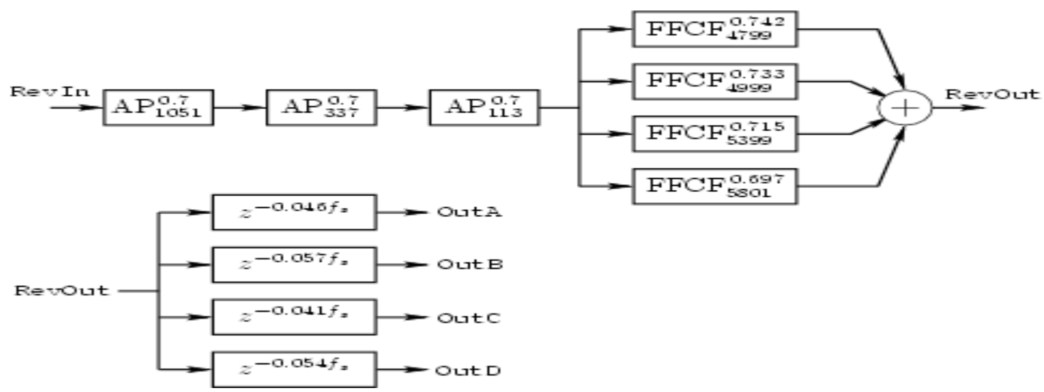

Figure.10 classic Schroeder Reverberator JVRev (Smith)

In this design, comb filter are connected in parallel. Its function is to reduce the spectral anomalies. It will attenuate the frequency that pass through other comb filter. The allpass filter is still connected in series to avoid the nonuniform amplitude response. By controlling the delay time, which is also called loop time, of each filter, the sound waves will become spectral and reverberated.

Development in Digital reverberation

In 1970 Schroeder extended his original design to incorporate a new component, Muiltitap Delay Line. This component is used to simulate the early reflections. The new design became the core part of the most commercial reverberator.

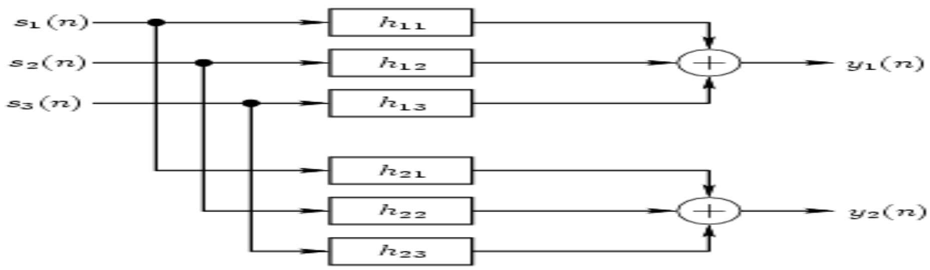

Figure.11 Tapped Delay Line FIR filter (Lawson)

After that, more scientists developed their reverberator with more functions and better performances. A notable achievement is Moorer's reverberator, which investigated in 1976.In Moorer's design, a lowpass filter inserted in the feedback loops to alter the reverberation time as a function of frequency (Hut 5). The lowpass filters implement a dc-attenuation, and a frequency dependent attenuation.

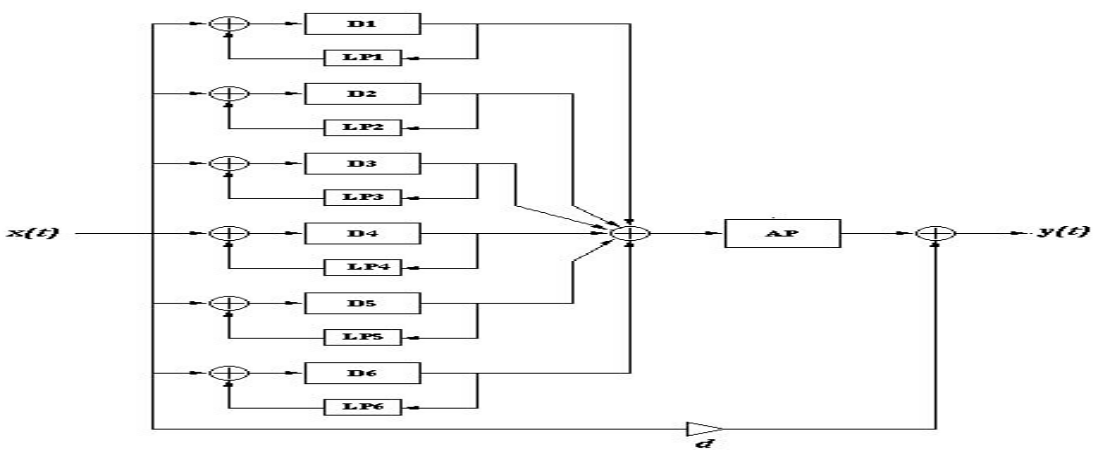

Figure.12 Moorer's Reverberator (Hut)

Conclusion

Schroeder reverberator is very simple and has less performance when comparing to fancy commercial reverberator with today's techniques. However, as a pioneer in the world of digital reverberation, Schroeder's work consists of building up the foundation of audio signal digital reverberation. Based on his algorithms, the comb filter and allpass filter became the "brick" of all the digital reverberator, and multitap delay line brought better performance. So, today, instead of wasting time to collect reflected sound wave in a small chamber room, we can sit in front of the computer and just click mouse button.

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