Investigation on Simulation and Measurement of Reverberation for Small Rooms

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***Abstract*—This paper investigates the simulation of reverberation in a small room using image source method, Schroeder model, and Gardner model. It then compares the simulation results with measurement of reverberation in the real measurement room. It shows a large difference between the simulation and the real measurement of reverberation due to the over-simplified representation of the sound source, receiver, and room surfaces in the models. A reverberation simulation model including the real measurement of the source, receiver, and room surfaces will be developed to better match the real measurement of reverberation.**

Keywords-Digital Signal Processing; Reverberaton; Room Acoustics; Embedded Systems.

I. Introduction

Reverberation in a room is the result of many reflections of sound that occurs in that room. Reverberation can be separated into direct sound, early reflections, and late reverberations [1]. Room reverberation has been modeled in many applications. For example, it can be used to help predict the acoustical characteristics of the new finished spaces, to help create a virtual studio effect without building the actual room, or to help compare the effect of different absorbing materials and treatments.

Reverberation also has extensive uses in our daily life. For example, when we try to speak to a group of people outdoors, we need to speak louder than when we speak in a room. The reason is that when we speak indoors, the room reverberation helps keep sound energy localized in the room, raising sound pressure level and distributing sound throughout it. On the other hand, when we speak outdoors, many of the reflective surfaces are missing and much of the sound energy is lost.

Although reverberation is almost always around us, we still need to create reverberation effects in some cases because sometimes we are in environments with very little or poor reverberation. In the reverberation effect creation, it is often mistakable that a simple delay device with feedback can produce reverberation. Actually, a simple delay unit can only simulate reflections with a fixed time interval between them, but it cannot produce a very important characteristic of reverberation, that is, the rate of arriving reflections changing over time. There has been a lot of research on the simulation of reverberations [2-7]. However, most of the research focuses on creating reverberation effects for recorded or live music in large concert halls. Moreover, most of the research does not compare the simulation results with the real measurement of reverberation.

In this paper, we investigate the simulation of reverberation in a small room using the popular image source method, Schroeder model, and Gardner model. We then compare the simulation results with real measurement of reverberation in the real measurement room. It shows that there is a big difference between the simulation and the real measurement of reverberation due to the simple representation of the source, receiver, and room surfaces in the models.

The remaining part of this paper is organized as follows. First, the fundamentals of reverberation are reviewed. Next, the reverberation simulations using digital signal techniques, including the image source method, Schroeder model, and Gardner Model are described. Then, the measurement and simulation of reverberation in a real measurement room are compared. Finally, the paper is concluded with a summary of results.

II. FUNDAMENTALS OF REVERBERATION

In this section, the fundamentals of reverberation, including reverberation time and the methods to create reverberation effects are reviewed.

A. Reverberation Time

Reverberation time is a key parameter to characterize reverberation in a room. It is defined as the time required for reflections of a direct sound to decay by 60 dB below the level of the direct sound. The reverberation time is controlled primarily by the room size and the absorption coefficients of the room surfaces. The room surfaces determine how much energy is lost in each reflection. Highly reflective materials, such as tile floors, brick walls, and concrete windows, increase reverberation time. Absorptive materials, such as curtains, heavy carpet, and people, reduce reverberation time.

Wallace Clement Sabine was the first person to develop methods to estimate reverberation time. He found that reverberation time is proportional to the dimensions of room and inversely proportional to the amount of absorption. The equation he proposed is:

$$R_T = \frac{0.161 \cdot V}{S \cdot \bar{a}}, \tag{1}$$

where V is the volume of the room in cubic meters, S is the total surface area of the room in square meters, and $\bar{a}$ is the average absorption coefficient in the room. According to this equation, typical small rooms have reverberation time from 100 to 500 milliseconds.

B. Methods to Create Reverberation Effects

It is often desirable to create reverberation effects for recorded or live music. Many methods are available to create reverberation effects, among which the most common way is to use spring reverberators, plate reverberators, or digital reverberators [8].

In the old days, spring reverberators or plate reverberators were often used to create reverberation effects. Spring reverberators provide a relatively simple and inexpensive method for creating reverberation effects. A spring reverberator uses a transducer at one end of a spring and a pickup at the other end to create and capture vibrations with a metal spring. Many musicians have made use of spring reverberator by rocking them back and forth, creating a thundering, crashing sound caused by the springs colliding with each other.

Plate reverberators were not used widely as spring reverberators because they are much more expensive. A plate reverberator uses an electromechnical transducer, similar to the driver in a loud speaker, to create vibration in a large plate of sheet material. A pickup captures the vibrations as they bounce across the plate, and the result is output as an audio signal.

Nowadays, the common way to implement reverberation is to use digital reverberators which use various signal processing techniques in order to create reverberation effects. Since reverberation is essentially caused by a very large number of echoes, simply reverberation algorithms use multiple feedback delay circuits to create a large, decaying series of echoes. A more detailed description of these computer models will be provided and the simulation results of the models will be presented in next section.

III. SIMULATION AND MEASUREMENT OF DIGITAL REVERBERATORS

Early digital reverberation techniques tried to simulate the room reverberation by using two types of IIR filters so that the output would gradually decay [1, 7]. One is the comb filter, which gets its name from the comb-like notches in its frequency response. The other is the allpass filter, which has the nice property that all frequencies are passed equally, reducing a coloration of the sound. Much of the early work on digital reverberation was done by Schroeder [1]. One of his well-known reverberator designs used four comb filters and two allpass filters. However, this design did not create the increasing arrival rate of reflections. Some improvements were made by Gardner [7].

The popular image source method can also be combined with these reverberation algorithms to create reverberation effects [7, 9]. Typically, a FIR filter made using the image source method is used to create the direct sound and early reflections, and IIR filters using the reverberation algorithms are used to create late reverberation. The early reflection and late diffuse reverberation are then combined to form the reverberation.

A. Image Source Method

The image source method has been a popular technique to simulate room acoustics [10-12]. The simulation of sound propagation from a source to a receiver within a rectangular room using the image source method is shown in Figure 1. The sound propagation is modeled as the superposition of the direct sound and a number of reflected sounds from the source to the receiver.

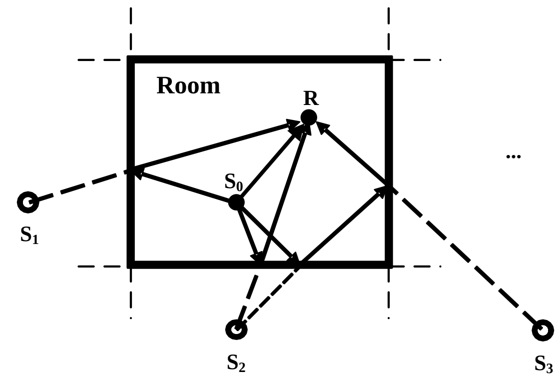


Figure 1: Simulation of sound propagation between two points in a rectangular room using image source method.

Image source method can be used to make the FIR filter by considering the composite contributions of each nearby virtual source to the receiver. The FIR tap delay lengths correspond to the sound travel time between the nearby virtual source and the listener. The FIR coefficient amplitudes are proportional to the reciprocal of distance to the nearby virtual source. The effects of absorption on the FIR coefficient amplitude should also be included since all reflections reduce the amplitude of the nearby virtual source by a factor of the reflection coefficient of the wall material. Actually, if an adequate amount of virtual source is considered, the image source method can also be used to calculate the whole reverberation.

B. Digital Reverberation Algorithms

Early work on simulating reverberation focused on the design of digital filters to model room responses. This work was based on the assumption that the perceptual difference between a real room and a greatly simplified simulation could be made small. As mentioned previously, Schroeder's initial reverberator design consisted of comb filters and allpass filters. A comb filter is a simple delay with feedback and its time domain impulse response of a comb filter is an exponentially decaying pulse train, which is shown in Figure 2.

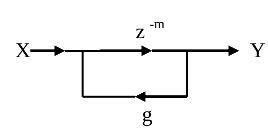


Figure 2: Flow graph of the comb filter

An allpass filter is just like a comb filter with a feed forward path around the delay, which is shown in Figure 3. In the all pass filter, the zeroes cancel the influence of the poles on the magnitude of the frequency response, resulting in a flat frequency response. However, the allpass filters affect the phase of signals.

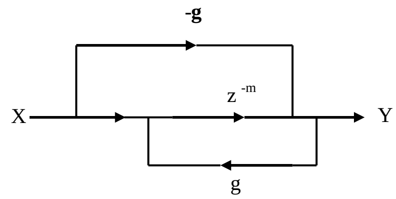


Figure 3: Flow graph of the allpass filter

The flow graph of one reverberator designed by Schroeder is shown in Figure 4. This reverbertor consists of four parallel comb filters feeding into two serial allpass filters. The parallel comb filters are used to simulate the echoes occuring between walls in a room since in the frequency domain, the peaks caused by the comb filters correspond to the normal modes of the room. However, the parallel comb filters do not supply a sufficient buildup of echoes for realistic diffuse reverberation. In order to increase the echo density, the output of the comb filters is fed into two allpass filters in series. Each allpass filter has a multiplicative effect on the number of echoes, but prevents coloration by taking advantage of the flat frequency response of the allpass filters [1].

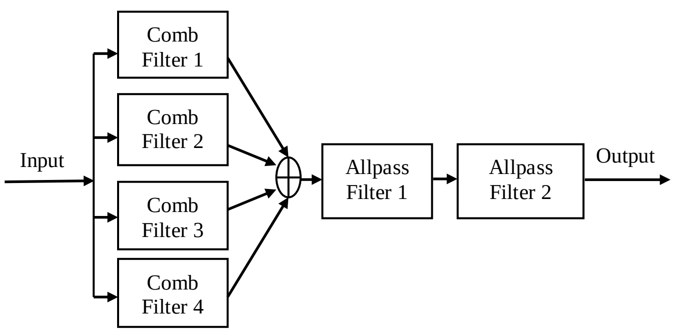


Figure 4: Block diagram of Schroeder's reverberator

Instead of using allpass filters in series as in the Schroeder reverberator, Gardner managed to combine them in a way that led to an exponential buildup of echoes as occurs in real rooms. He used a nested allpass filter, which embeds an allpass filter into the delay element of another allpass filter [7]. In a nested allpass system shown in Figure 5, the echoes generated by the inner allpass filters will be recirculated to their inputs via the outer feedback path. The number of echoes generated in response to an impulse will therefore increase over time as what we expected rather than remaining constant as with a standard allpass filter.

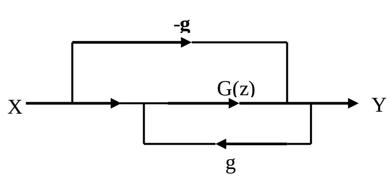
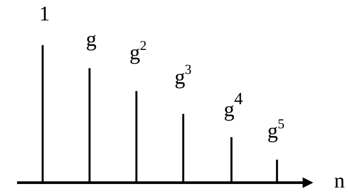


Figure 5: Flow graph of the nested allpass filter. Here G(z) is also an allpass filter.



C. *Simulation and Measurment of Reverberation*

In this sub-section, the simulations of room reverberation using the popular image source method and the digital reverberation models are shown. A real measurement of reverberation is also presented. The subjective comparison between the simulations and the measurement is then performed.

Before the room reverberation was modeled, the measurement room dimensions, the source position, and the receiver position were measured. The layout of the measurement room is shown in Figure 6. The rectangular measurement room is 3.37 meters long, 3.03 meters wide, and 2.39 meters high. The sound source is localized in the position of (0.31, 0.28, 1.02) meters and the receiver is in the position of (1.57, 1.49, 1.02) meters. The reflections coefficients are 0.90 for the walls, 0.85 for the ceiling, and 0.85 for the floor. The temperature in the measurement room was 19° C, giving a sound velocity of 342.8 m/s.

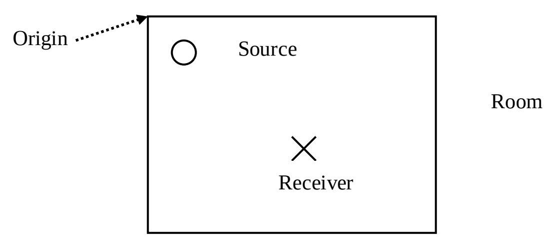


Figure 6: The layout of the measurement room

A MATLAB program was developed to implement the image source method to simulate the reverberation of the room. The reflection order was set to 20, resulting in 68921 virtual sources, which is sufficient to simulate the reverberation. The simulated reverberation result is shown in Figure 7. It is shown that the reverberation decays very quickly.

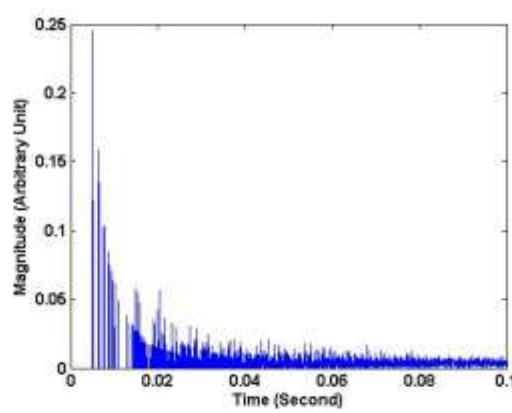


Figure 7: Reverberation simulation using the image source method

Then, the image source method was used to simulate the direct sound and early reflections of the room. The first 20 milliseconds of the direct sound and early reflections generated by the image source method were extracted and then fed into the IIR filters to simulate the late reverberations. The IIR filters were implemented using Schroeder model and Gardner model respectively. The direct sound, early reflections, and late reverberation were then combined to form the whole reverberation. The reverberation results using these two models are shown in Figure 8 and Figure 9 respectively. It is shown that the magnitude of the late reverberation decrease exponentially for both models, but it is hard to tell which one is better by just looking at the simulation of reverberation.

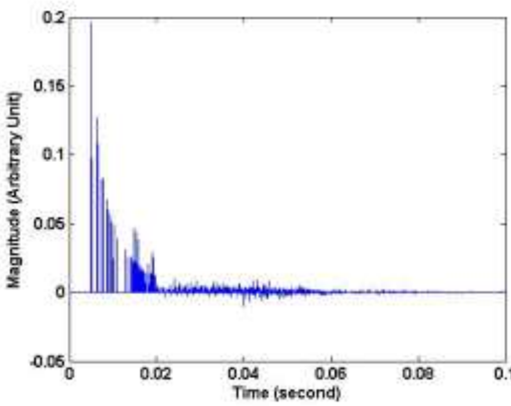


Figure 8: Reverberation simulation using Schroeder model

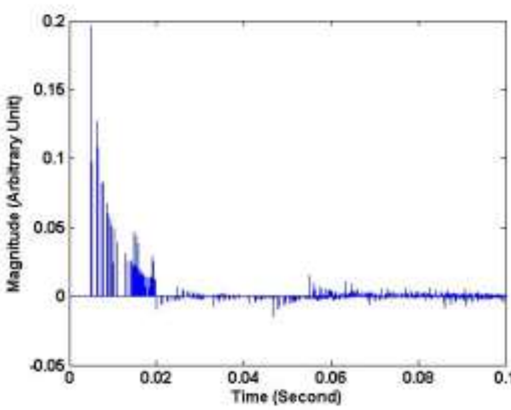


Figure 9: Reverberation simulation using Gardner model

The reverberation in the real measurement room was also measured using the SMAART software **[**13**]**. Here the sound generated from the speaker acts as the sound source and a high quality microphone acts as the receiver. The measurement of the room reverberation is shown in Figure 10. It can be seen that the direct sound and some early reflections are very similar to the simulation results. However, the late reverberations are stronger than that simulated by the image source method or the two models. It should also be noticed that there are some energy components among the early reflections, which seem to be the joint effects of the measurement errors, the polar responses of the speaker or microphone, and the scattering of the room surfaces.

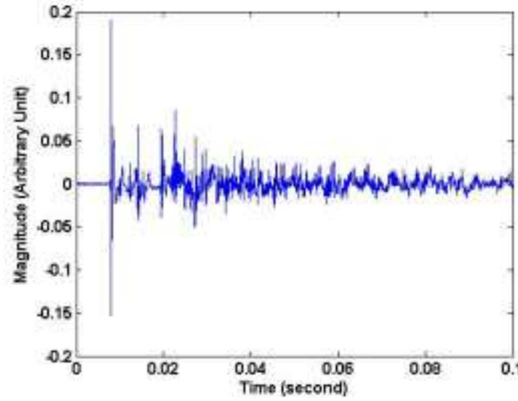


Figure 10: Reverberation measurement of the real room

The quality of the reverberation simulation can now be judged subjectively by listening to the output sound signal of the reverberator. Both male speech and female speech were fed into the two reverberators, the system using the image source method, and the system with the real measurement of reverberation. By listening to the output sound of these systems, we found that the real room has stronger reverberations than any of the simulation systems. It appeared that the Gardner model had a result more similar to the real room, which also showed that the Gardner model has a better performance than the image source method and the Schroeder model in modeling room reverberation.

IV. CONCLUSION

This paper briefly reviewed the fundamentals of reverberation and its simulation methods. It then compared the simulation and the measurement of reverberation. It showed that the simulations using the image source method, Schroeder model, and Gardner Model have similar results. But they differed substantially from the real measurement of reverberation. The difference might be due to the over-simplified representation of the source, receiver, and room surfaces in the reverberation models. A reverberation model that includes the real measurement of the source, receiver, and room surfaces is being developed to better match the real measurement of reverberation.

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