Implementation of 3D Audio Acoustic Modeling for Real Time Application using Blackfin Processor

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Abstract— Signal processing techniques allow us not only to compensate for the acoustics of a listening environment, but also to add special musical effects such as delay, echo, and reverb, if desired. The most common example is the audio equalizer that allows us to enhance the bass or treble content of an audio signal. What a listener hears in a room depends not only on the acoustics characteristics of that room, but also on the location of the listener.

In any listening space, an audio source consists not only of the direct sound, but also early echoes reflected directly from the walls and other structure, and a late sound or reverberation that describes multiple reflections (that get added to other echoes). The direct sound provides clues to the location of the source, the early echoes provides an indication of the physical size of the listening space, and the reverberation characteristics the warmth and liveliness that we usually associate with sounds. The amplitude of the echoes and reverberation decays exponentially with time. Room reverberation can be simulated with a set of IIR filters. An artificially add these reverberation effects to the music and make it sound richer.

Index Terms— binaural sound systems, FIR, IIR filters, reverberation.

I. INTRODUCTION

Reverberation is a very common and often unnoticeable phenomenon in our lives. The surrounding walls in a auditorium and in the office, the walls of the buildings in the street, every object around us reflects the sound that is propagated through the space.

The reverberation starts with production of a sound in a room [2]. The acoustic wave meets the walls, ceiling and other surfaces, where the energy is absorbed and reflected. The reflected energy leads to reverberation. If a direct path exists between the source and the listener, the listener will hear first the direct sound followed by reflections of the nearby surfaces, these being called early reflections. After some tens of milliseconds, the number of reflected waves becomes very large with a decreased evolution in time, characterized by a dense collection of echoes moving in all directions, their intensity being independent from the location in the room. These are called late reflections. Artificial reverberation tries to model the performance of a real concert hall and consists of a cascade of an early reverberator and a late one. The necessary time the sound pressure level to decay to 60 dB of its original value is defined as reverberation time. Due to absorption the reverberation time varies with the frequency; for instance, walls tend to absorb...
higher frequencies much more than low frequencies. Reverberation in a room usually is measured by the corresponding impulse response. The model of the impulse response consists of the direct signal followed by discrete echoes called early and late reflections. Artificial reverberation tries to model the performance of a real concert hall and consists of a cascade of an early reverberator and a late one.

A. Principle of Operation

For modeling both early and late reflections a complete reverberator consists of an early reverberator and a late one, usually connected in a cascade. An artificially added these reverberation effects to the music and make it sound richer. The basic elements of the reverberators are comb, all-pass filters, that are cascaded, nested or with feedback [5].

B. Literature Survey

The concept of reverberation and efficient reverberation algorithm are discussed by Norbert Toma, Marina Topa, Erwin Szopos[2], M. Topa. N. Toma, E. Szopos [1], [3]. As the work is implemented using ADSP-537 EZKIT reference is made to the manual of Analog Devices [10], [11]. The complete work proposed in the project is implemented using ADSP-537 Blackfin processor. The details about the processor are obtained by referring to the web sites [12], [13]. As the work is implemented using ADSP-537 EZKIT reference is made to the manual of Analog Devices [12]. The complete work proposed in the project is implemented using ADSP-537 Blackfin processor.

II. Design And Implementation

Sound processing is one of the applications of Digital Signal Processing. Different effects can be performed with various types of filters. The audio 3D effects are artificially generated using various signal processing circuits and devices, and they are increasingly being performed for different applications using digital signal processing techniques. They can be realized and done by Time-Domain Operations and Frequency-Domain Operations.

A. Digital Processing Techniques

The implementations can be broken down into efficient circular buffers and operations on delay lines. The recent developments in digital hardware have made reverb processors available at efficient, inexpensive prices that are portable and quite flexible. In early days the digital reverberation algorithms tried to mimic the room’s reverberation by using primarily consisted of two types of infinite impulse response (IIR) filters, so that the output would gradually decay. One of the commonly known such filter is the comb filter, which has got comb-like notches in the frequency response. The other primary filter is the all-pass filter. The all-pass filter has the nice property that all frequencies are passed equally, reducing a coloration of the sound. Much of the early work on digital reverb was done by Schroeder, and one of his well-known reverberator designs uses two all-pass filters and four comb filters. This design does not create the increasing arrival rate of reflections, and is rather primitive when compared to current algorithms.

Typically, a finite impulse response (FIR) filter is used to create the early reflections, and then IIR filters are used to create the diffuse reverberation. Low pass filters may be used to model the air absorption.

B. Parameters of reverberation

- **Reverb Decay**: The reverb decay indicates how you how long the reverb can be heard after the input stops. The reverb decay is typically in terms of milliseconds, which can be thought of as something like the reverb time.
- **Gate Time**: This parameter applies to gated reverbs. The gate time is simply the length of time that the reverb is allowed to sound. This may also refer to the length of a reverse reverb.
- **Predelay**: The predelay is the amount of time before the first reverberations of a signal are heard, i.e. the time before the first early reflection in the impulse response. In some cases, the predelay may be defined as the time before the late reflections are heard. More complex reverberation units may actually allow you to set the predelay for both the early and late reflections. For simulation of real environments, the predelay for the early reflections should always be smaller than for the late reflections are shown Figure 1.
III. RESULTS AND DISCUSSIONS

Room reverberation [6] can be simulated with a set of IIR filters. The reverberation algorithm simulates the sound reflections in an enclosed environment such as rooms, concert halls, etc. These audio effects enhance the listening experience of audio playback in an environment with little or poor reverberation. For example, during listening music with headphones, there is no reverberation from the surroundings that is being added to the music. Artificially add these reverberation effects to the music and make it sound richer.

Reverberation consists of three components: direct sound, early reflections, and late reflections (or reverberation). Direct sound takes a direct (shortest) path from the sound source to the receiver (or listener). Early reflections, which arrive within 10 to 100 msec after the direct sound, are caused by sound waves reflected once before reaching the listener. The late reflection is produced after 100 msec and forms the reverberation. The reverberation time $t_{60}$ is defined as the time needed for the impulse response to decay by 60 dB. Reverberation time is often used to characterize the reverberation of an enclosed space. A large room such as a concert hall has a long reverberation time between 1.5 and 2s, whereas a small meeting room has a reverberation of few hundredths of a millisecond.

A digital reverberation algorithm was proposed by Schroeder [4] to model the reverberation effects in a room. As shown in Figure 2, this algorithm simulates room reverberation with four comb filters $C_1(z)$, $C_2(z)$, $C_3(z)$, and $C_4(z)$ connected in parallel, followed by two cascaded all-pass filters $A_1(z)$ and $A_1(z)$.

Figure 1: A room impulse response with the pre-delay parameters labelled

- **Gate Decay Time:** Some units with gated reverbs will also provide this parameter, which controls how the gate is actually applied or ‘closed’. A very short gate time means that the reverb is cut-off rapidly. Longer decay times mean that the reverb is given some time to fade away gradually.

- **Gate Threshold:** Rather than apply a gated reverb to an entire signal, could very well only gate the reverb depending on signal levels. Typically, the gate on a reverb will be kept open (the impulse response is not truncated) when signals are above this value, but as when the signal drops below the threshold, the gate closes and the number of reflections is reduced. The gate will open again when the signal rises back above the threshold. Some gated reverbs may use a threshold that is not user programmable.

![Figure 2: Block diagram of a digital reverberation algorithm](image)
The transfer function of the comb filter is expressed as,

$$ C_i = \frac{1}{1 - \alpha z^{-D_i}}, \quad i=1, 2, 3, 4 \quad \ldots \quad (1) $$

Which is the IIR filter with $D_i$ poles equally spaced ($2\pi/D_i$) on the unit circle. The coefficient $\alpha_i$ determines the decay rate of the impulse response for each comb filter. The comb filter increases the echo density and gives the impression of the acoustics environment and room size. However, it also causes distinct coloration of the incoming signal. All-pass filters prevent such coloration and emulate more natural sound characteristics in a real room. The transfer function of the all-pass filter is expressed as,

$$ A_i(Z) = \frac{-\alpha_i + z^{-D_i}}{1 - \alpha_i z^{-D_i}}, \quad i=5, 6 \quad \ldots \quad (2) $$

By cascading the all-pass filters with comb filters, the impulse response of the overall system becomes more diffuse.

The digital reverberation algorithm shown in Figure 3 can be simulated with MATLAB. Figure 4 specifies the time delays of 40, 44, 48, and 52 msec for the four comb filters. The all-pass filter has a delay of 7 msec and gain=0.4. The reverberation algorithm is running at 48 kHz. Setting all feedback coefficients, $\alpha_i, i = 1 \ldots 4$, of the comb filter as 0.75. The simulation result with the input wave file liutm_48k_mono.wav is shown in Figure 3. The simulation result with variable gain of comb filters is shown in Figure 4. The comb filters has a gain of 0.6284, 0.6237, 0.6189, 0.6143 respectively with all-pass gain of 0.4 are shown in Figure 4.

This section implements the reverberation algorithm in real time with the Blackfin EZ-KIT. The performance of the previous floating-point MATLAB implementation must be re-evaluated with the fixed-point (1.15) format before porting to the Blackfin processors. The simulation results for all the comb filters are shown in snapshot Figure 5, Figure 6, Figure 6, and Figure 8 respectively.
Figure 5: Snapshot of Comb0_left and Comb0_right output

Figure 6: Snapshot of Comb1_left and Comb1_right output
Figure 7: Snapshot of Comb2_left and Comb2_right output

Figure 8: Snapshot of Comb3_left and Comb3_right output
In addition to the generation of reverberation shown in Figure 2, this project also implements the early reflection portion and direct path of the impulse response as shown in Figure 9. This reverberation structure, proposed by Moorer [9], produces a more realistic reverberation sound than that proposed by Schroeder (Fig. 2).

![Figure 9: Moorer's digital reverberation structure](image)

To build and run the project we have to connect stereo music to the audio input port of the EZ-KIT. Press SW11 for the BF537 to listen to the reverberation version of the music and then press SW10 for BF537 to switch back to the pass-through mode. Three gains ($g_1$, $g_2$, and $g_3$) can be individually controlled to emphasize the significance of the early reflection generator, reverberation, and direct path signals. It is also made possible to turn off the contribution of different paths by setting the corresponding gains to zero.

![Figure 10: Snapshot of All-pass_left and Allpass_right output](image)

IV. CONCLUSIONS

In this paper considered the Schroeder's digital reverberation algorithm and Moorer's digital reverberation structures are implemented successfully on ADSP-537 EZ KIT for the generation of reverberation effects. This paper also presents some techniques for artificial reverberation used in enhancement systems. For modelling both early and late reflections a complete reverberator consists of an early reverberator and a late one, usually connected in a cascade. The basic elements of the reverberators are comb, all-pass filters that are cascaded, nested or with feedback. The early Schroeder's reverberator algorithm provides a set of equally spaced impulses. Real rooms have in their impulse response a diffuse sound energy component between early...
This phenomenon was simulated by introducing all-pass filters. Best results are obtained combining the Griesinger early binaural reverberator with all-pass filters. The late reverberation comprises exponentially decaying diffuse impulses with a high density. The classical late Schroeder's reverberator is built with comb and all-pass filters; it has many disadvantages as, annoying sound and impossibility to specify a relation between the reverberation time and frequency. The improved reverberator solves the problem with the help of the absorbent all pass filters.

The echo filter is also serves as a basis for several other audio special effects like chorusing, flanging, phasing etc. A spatial sound effect as in concert halls can be achieved by cascading an early and late reverberator. The best complete reverberator consists of a Griesinger early binaural reverberator, all-pass filters and absorbent all-pass late reverberator. Its impulse response and spectrogram is similar with the one obtained for a real room and is much better than the one of a classical Schroeder's reverberator.

REFERENCES


