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Real Time Room Acoustic Simulation by a Reflection-Diffusion Reverberator

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Abstract: - A new model of artificial reverberator for real-time room acoustic response simulation is proposed. It's structure is inspired by Jot's model, but with the introduction of a not so much previously exploited aspect of the generation of the reverberation phenomenon. In fact, together with the reflection component, it also takes into account the presence of a diffusion component. The proposed reverberator makes massive use of All Pass filters in order to mix the multiple echoes produced by Comb filters, and to increase the echo density. Diffusion and reflection feedback matrices have been introduced, in order to adapt the reverberator to different kind of rooms.

Key-Words: - Reverberation, All Pass Filters, Comb Filters, Room Acoustic Simulation

1 Introduction

It's known that a sound produced inside an enclosed space is modified by the reflections of its wave fronts over the wall surfaces. This phenomenon deeply influences the auditory feeling of a listener and gives a particular personality to the sound. For example, church music can be more realistic if performed in a very reverberant place, capable of mixing and blending the sound produced by instruments and singers as a powerful resonance chamber.

On the contrary, a piano concert needs a dry ambient, even if with a bit of reverberation, being of primary importance the capability of distinguishing without ambiguity every note played by the pianist. So, it's of wide interest the artificial reproduction of the acoustical characteristics of a given ambient.

2 Reverberation

The reverberation phenomenon consists in the persistence of a sound emitted by an acoustic source, gradually attenuating during a given time interval, after the source has stopped. This is due to the multiple reflections of acoustic spherical waves on the walls of a listening room, attenuated by an absorption coefficient α that depends on the constitutive materials [1] of the walls.

As soon as the acoustic spherical waves hit the walls, the floor and the ceiling, they are reflected and retransmitted backward to the inner of the room. These new wave fronts go back to the listener, covering longer distances than the direct front. So, they are perceived by the listener in different times, producing this way the reverberation effect.

Reflected waves can be divided into two classes. The early reflections, i.e. waves reflected only once before reaching the listener; are essential for the brain to reconstruct the real dimensions of the room. A wave front can be reflected a lot of times before it is captured by the listener's ear. This kind of reflection belongs to the second class (late reflections) and it is particularly responsible of the generation of that dense mass of echoes that, arriving to the listener in random times, constitute the reverberating tail. In a room with 6 faces, we have six early reflections, while the number of late reflections can be enormous. The reverberation tail consists of millions of reflected sounds, reaching the listener in random times. Its energy decreases in time, because of air and walls' materials absorption. In Fig.1, the impulse response (IR) of a room is depicted.

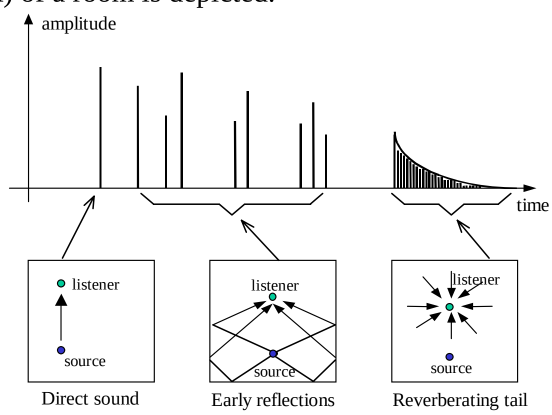

Fig.1 – Impulse response of a reverberating room.

The reverberation time T_{60} (time needed by the impulse response to decay to –60 dB with respect to its maximum level) quantifies the reverberation entity in a room. It is measured (Schroeder [2]) by integrating the IR to get the *Room Energy Decay* curve (EDC):

$$EDC(t) = \int_t^{\infty} h^2(\tau)d\tau \qquad (1)$$

where $h(t)$ is the IR. $EDC(t)$ is what remains of the acoustical energy in the IR at time t. T_{60} can be derived from EDC slope. Jot [3] extended the concept of *EDC* to bring out the frequency dependent nature of reverberation. He proposed a variation of *EDC* called *Energy Decay Relief*, or *EDR*(t,f), where reverberation decay is a function of time and frequency (Fig.2).

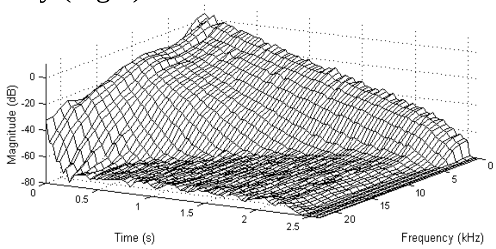

Fig.2 – Energy decay relief of a large hall.

We can see that IR energy decays more slowly at low frequencies, because air and walls absorption is more incisive at high frequencies.

3 Artificial reverberation

Schroeder was the first trying to realize a simple reverberator, based on the properties of the *comb filter* (*plain reverberator*), as in Fig.3:

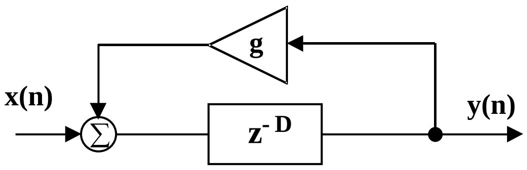

Fig. 3 – plain reverberator

The comb filter transfer function is as follows:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{z^{-D}}{1 - g \cdot z^{-D}} \qquad (2)$$

To simulate the frequency dependent absorption by air and walls, Schroeder inserted a low-pass filter (LP) in the feedback loop (Fig.4), obtaining the following transfer function:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{z^{-D} - g_1 \cdot z^{-(D+1)}}{1 - g_1 \cdot z^{-1} - g_2 \cdot z^{-D}} \qquad (3)$$

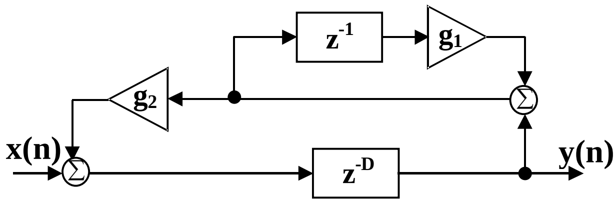

Fig. 4 – Comb filter with Low-Pass in feedback loop.

In order to increase time echoes density, Schroeder exploited the properties of an all-pass filter (Fig. 5). This filter is not inspired by a physical phenomenon: it's presence is justified only by the necessity of increasing the time echo density, without adding any coloration to the sound. Moreover, Schroeder cascaded multiple all-pass filters to achieve a further increase of echoes.

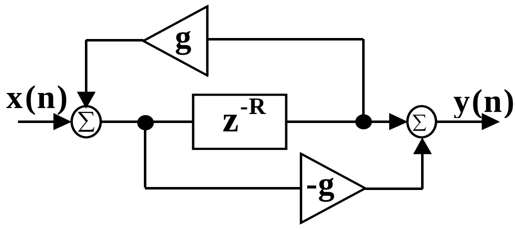

Fig. 5 – Al-pass filter.

To achieve a good echo density, Schroeder [3] used both comb filters (in parallel) and all-pass filters (in series), to obtain a structure as in Fig.6. Comb delay lengths are chosen prime between them, in order to avoid an overlapping of peaks.

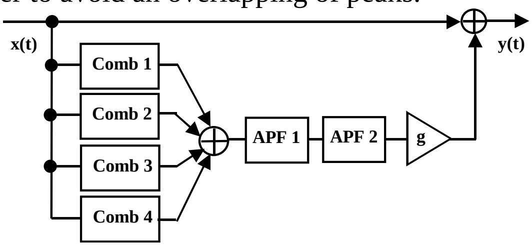

Fig. 6 – Schroeder's reverberator.

3.1 Jot's reverberator model

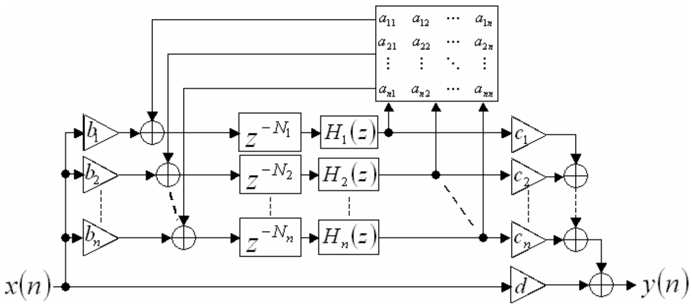

Fig. 7 –Jot's reverberator.

This architecture [4] is characterized by some kind of an IIR subsystem, obtained by coupling the comb filters by means of a feedback coefficients matrix *G*.

The presence of this feedback is justified by the following needs. Firstly, to obtain a time remixing of echoes, to cancel the periodicity introduced by comb and FIR filters an then to give more realism to the reverberating tail. The elements in the main diagonal of the *G* matrix are all null, to avoid that the output of a comb filter be fed back at it's same input.

4 Reflection-Diffusion Reverberator

Our aim is to improve the characteristics of the reverberating tail, in terms of further remixing of echoes, a more uniform energy distribution in the time-frequency spectrum of the IR, and in a less audible "periodicity effect" in the time response of comb filters.

Being inspired by the previous models, particularly by Jot's model, we have developed a new reverberator (Fig. 9a). The meaning of all the quantities appearing in these figures is the following:

$g_1^C \ldots g_n^C$: gains to control the amount of reflections in the input signal.

D : gain to control the amount of direct signal

$z^{-M_1} \ldots z^{-M_n}$: delays that compensate the different activation time of the comb filters

$g_1^{X_2} \ldots g_n^{X_2}$: (X=R, D) coefficients of the comb filters

$\boldsymbol{M}^D, \boldsymbol{M}^R$: diffusion/reflection feedback matrices

$z^{-N_1} \ldots z^{-N_n}$: comb filter delays

$H_1^X(z) \ldots H_1^X(z)$: (X=R, D) low pass filters that affect the input signal reflection; take into account the frequency dependent absorbing behavior of a room.

$g_1^{D_1} \ldots g_n^{D_1}$: diffused portion of the signal. $g_1^{R_1} \ldots g_n^{R_1}$: reflected portion of the signal. $g_i^{R_1} = 1 - g_i^{D_1}$

$APF_i^{D_j}(z)$: (j=1, 2) all pass filters introduced to increase time echo density, but at the same time reducing the number of parallel comb filters needed to achieve the same results. They have different delay times (5 ms and 1.7 ms), but identical gain; the impulse response of an *APF* is characterized by peaks decreasing in time.

In order to better understand the role played by the different blocks of the architecture, some considerations are needed:

- the low pass filters set in the forward path of the comb filters take into account the absorbing behavior of the walls.
- the output low pass filter, $H_y(z)$, takes into account the absorbing behavior of the air.
- the activation delay times of the comb filters are different from their delay times. Nevertheless, adding these delays, we obtain the same result for each comb filter and this result must equal the delay time of the first reflection.
- to further increase the echo time density, we have added two output *APFs.*
- $\boldsymbol{M}^R$ has a role in the modelling of the reflection contribution to the reverberation effect and aims to bring back to the input of one comb filter the outputs of all the other filters.
- $\boldsymbol{M}^D$ takes into account the presence, at the point of incidence of a sound ray, of the diffusion contribution to the reverberation effect.

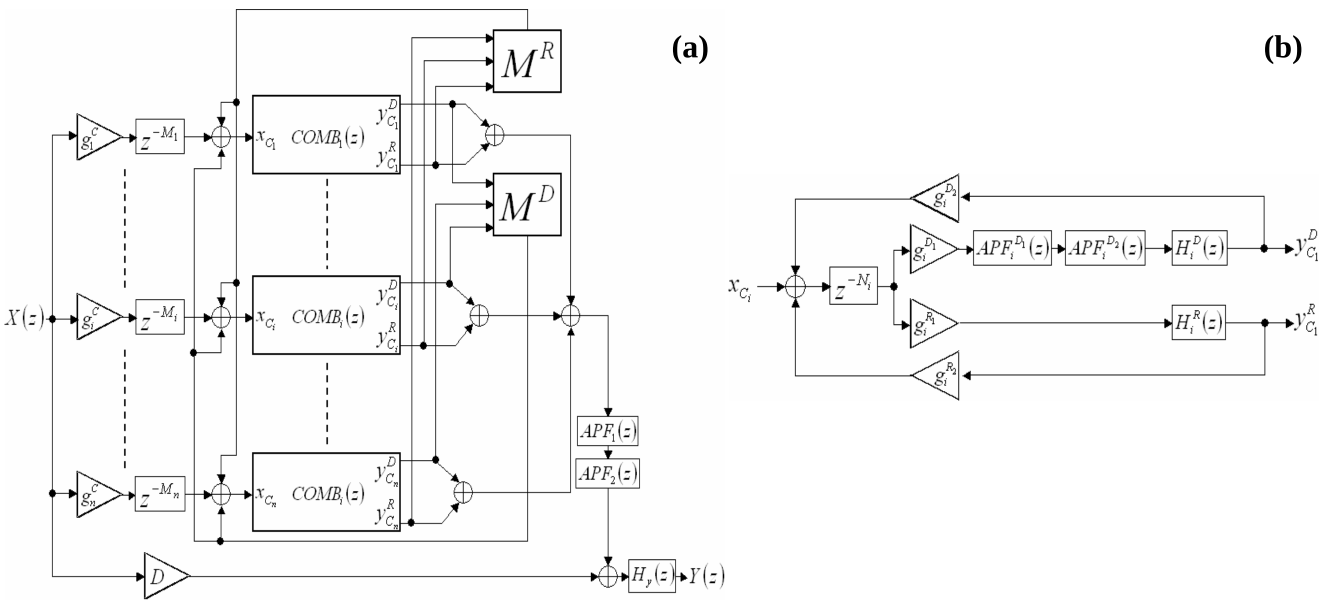

Fig. 9 (a) structure of the proposed reverberator – (b) structure of the COMB subsystem.

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- With $\boldsymbol{M}^D = 0$, we can implement an improved version of Jot' reverberator. For test purposes, both matrices $\boldsymbol{M}^D, \boldsymbol{M}^R$, when not null, were randomly filled with values between –0.4 and +0.4

5 Simulation results

Every experiment was carried out at a sampling frequency of 16 kHz. With some specific non zero values, the presence of both reflection and diffusion matrices introduces a resonance that reveals itself to be very hard to eliminate. This is why, after some non successful experiments, we decided to make tests always zeroing one of the two matrices. In any case, we skillfully varied some parameters in order to reach a uniform reverberation. Tests are illustrated by both time waveforms and frequency spectrograms. Among the others, a worth mentioning test is the one that simulates Jot's model by zeroing matrix $\boldsymbol{M}^D$ ($D = 0.9$):

$g_1^C \dots g_n^C$	1	1	1	1	1	1
$g_1^{D_1} \dots g_n^{D_1}$	0.2	0.2	0.2	0.2	0.2	0.2
$g_1^{R_1} \dots g_n^{R_1}$	0.9	0.9	0.9	0.9	0.9	0.9
$g_1^{D_1} \dots g_n^{D_1}$	0.9	0.9	0.9	0.9	0.9	0.9

Table 1

We confirmed that, for Jot's model, it's not so easy to guarantee a non resonating output (Fig. 11):

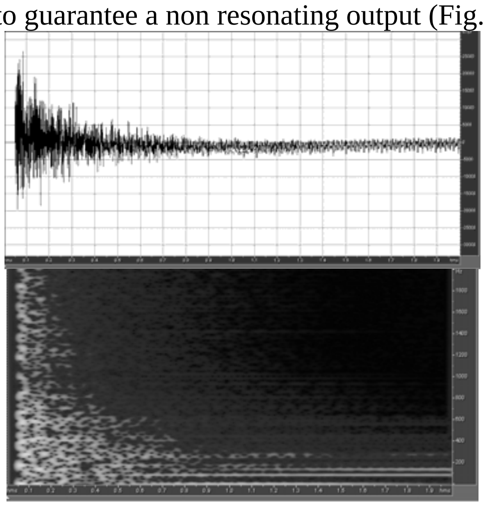

Fig. 11 – output waveform suffering from resonance

As we can see, energy is not uniformly distributed in the spectrum and Comb's periodicity are clearly visible.

In the following test, we succeeded in obtaining a reverberation not affected by resonance (all parameters values as before except for matrix $\boldsymbol{M}^D, \boldsymbol{M}^R = 0$). The characteristics of this test's results can be observed in the following figures:

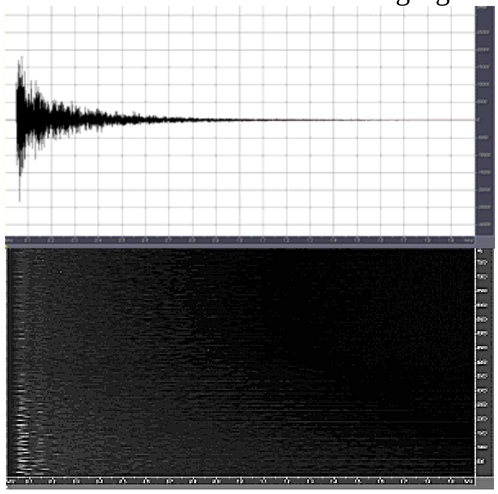

Fig. 12 – output waveform without resonance

As we can see, time echo peak density is good, as results with Jot's model, too. Moreover we have also obtained a noticeable uniformity in the spectrum of the output signal, with respect to the previous test.

6 Conclusion

A new model of artificial reverberator for real-time room acoustic response simulation has been proposed. It's an evolution of Jot's model, with the introduction of the modeling of the diffusion effect. As a consequence, our model can better reconstruct the characteristics of a real reverberating tail. Instability phenomena have been controlled, and a more uniform frequency response has been reached.

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