Efficient and Parametric Reverberator for Room Acoustics Modeling

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Abstract

A computationally efficient digital filter structure for late reverberation modeling based on measured room acoustical data is presented. In our reverberator a dense response is obtained by inserting comb-allpass filters in comb filter loops which are connected in parallel. The sum of the comb filter outputs is fed back to their inputs. The advantage of the proposed structure is that a higher reflection density is obtained by a smaller computational burden than with former reverberators. To simulate the acoustics of an existing hall, the early reflections and the frequency dependent reverberation time are analyzed from measured room impulse responses. The data is used for deriving the parameters of the reverberator.

1 Introduction

Reverberation modeling by digital filters has been studied since Schroeder [1] introduced a method where parallel comb filters are used for producing artificial late reverberation. Later, a tapped delay line was suggested for producing the early reflections to create a more natural reverberation effect [2]. Moorer [3] presented a reverberation structure containing six parallel comb filters and one comb-allpass filter in series. He also studied the definition of the parameters of the IIR lowpass filters used after the delay lines to implement a frequency-dependent reverberation time. A further improvement for late reverberation modeling has been to use feedback delay networks (FDN) where the feedback from each delay line is connected to all other delay lines through a unitary feedback matrix [4], [5]. The advantages of FDNs are diffusion of reflections, an increasing reflection density versus time, and uncorrelated delay line outputs which still have energy from the modes of all the delay lines.

In [6] we introduced a system where real-time binaural room acoustics simulation was carried out using an efficient reverberator structure. In this work the IIR module used for late reverberation modeling is based on the principle of multiple feedback connection, like in the case of the FDN. The difference is that instead of using a feedback matrix the outputs of the parallel comb filters are summed and fed back to the inputs of the comb filters. Additionally, there are diffusive comb-allpass filters inserted in the comb filter loops which yields a further increasing reflection density versus time. As a result fewer parallel delay lines are needed than in Moorer's reverberator or in an FDN to achieve natural sounding reverberation. A dense reverberation is achieved with smaller amount of computation than in the compared structures.

In Section 2 we present the structure of the IIR filter used for late reverberation simulation. In Section 3 we compare the proposed structure with an FDN and Moorer's reverberator. The whole procedure of rendering stereophonic reverberation for a mono source signal, based on measured data of a binaural room impulse response, is presented in Section 4. Finally, Section 5 concludes the paper.

2 Novel Structure for Late Reverberation Modeling

The structure of our reverberator is illustrated in Fig. 1. The reverberator consists of several parallel feedback loops which contain a delay line DLk, a comb-allpass filter $A_k(z)$, and a lowpass filter $H_k(z)$. The lengths of the delay lines and the delays of the comb-allpass filters determine the memory requirement of the system. The lowpass filters implement the frequency-dependent reverberation time (similarly as in [4] and in [3]). One-pole lowpass filters of the type shown in Fig. 2 a) are typically used. The comb-allpass filters bring about a diffusing effect on the recirculating signal in the feedback loops. The feedback structure is a special case of an FDN, where the feedback matrix is unitary and circular with different gains on the diagonal than elsewhere in the matrix [4]. The + and - signs before summing for the output are to obtain a maximally flat frequency response [5].

The transfer functions of the comb-allpass filters are

$$A_k(z) = \frac{a_k + z^{-M_k}}{1 + a_k z^{-M_k}}, \tag{1}$$

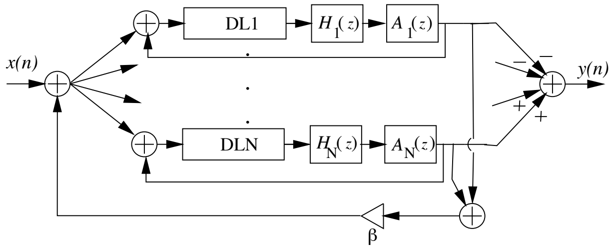

Figure 1: A digital reverberator consisting of parallel comb filters with diffusive and dispersive comb-allpass filters in the delay loops and with a feedback of the summed comb filter outputs.

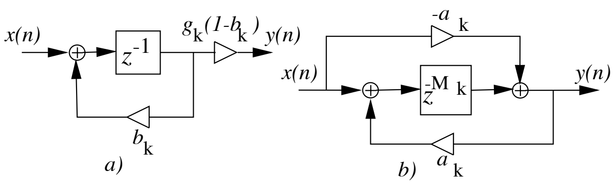

Figure 2: The block diagrams of the lowpass filter used in connection with each delay line and the comb-allpass filter used for diffusing the reflections.

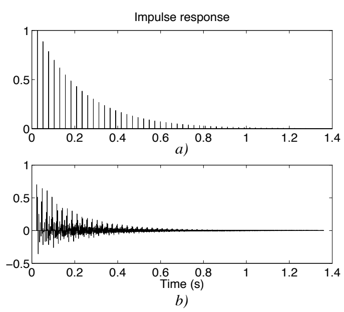

Figure 3: An impulse response of a single comb filter (a) and from a comb filter where a comb-allpass filter is inserted inside the feedback loop of a comb filter (b). The total delay line length in both cases is 1150 samples,and the feedback gain $g = 0.9$. The delay of the comb-allpass filter in the lower plot is 150 samples and the coefficient $a = -0.7$.

where $k = 1, 2, ..., N$, the filter coefficients are $-1 \leq a_k \leq 1$, M_k are the delay-line lengths, and N is the number of delay lines. One implementation structure of this transfer function is depicted in Fig. 2 b).

In Fig. 3 the impulse response of a comb filter and the same filter with a comb-allpass filter inserted inside the delay loop are plotted. It can be seen that when the filter $A_k(z)$ is added the regularity of the comb filter response is broken and soon becomes dense which is desirable in artificial reverberation.

3 Comparison of the Reverberators

The proposed filter structure containing four delay lines has been compared to an FDN containing 8 delay lines and a Moorer reverberator with 8 parallel comb filters. All the simulated structures have delay line lengths in the same range to simulate the same acoustic space, a concert hall with a reverberation

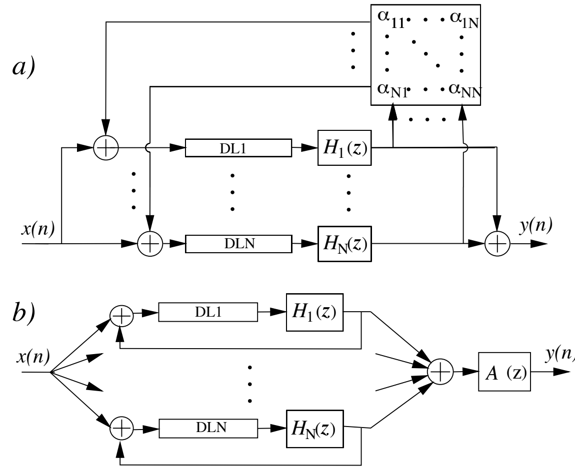

Figure 4: General structures of the FDN (a) and Moorer's reverberators (b).

time of about 1.5 s. The compared structures are illustrated in Fig. 4. The comparison of the time development of the reflection densities is shown in Fig. 5 and it can be seen that our reverberator with four delay lines develops a dense time response faster than an 8x8 FDN system, and the Moorer construction with 8 comb filters never reaches a larger reflection density than approximately 4000/s. Note that in Fig. 5 a) the reflection density of our structure increases quickly up to 10000/s which is a suggested minimum value in [7] for the reflection density to avoid flutter in the reverberation for short transient-like sounds.

The reflection density as a function of time has been obtained by computing the individual reflections in a 20 ms sliding window in each of which the peaks within 20 dB from the strongest reflection are counted for defining the density as was suggested by Griesinger [7]. The algorithm has been implemented by detecting the absolute values of the response which exceed the neighboring values.

The lengths of the delay lines in samples at 44.1 kHz for all the compared reverberators are listed in Table 1. Table 2 shows their computational requirements. The amount of delay memory in our reverberator is 17585 words and in the other two structures 35407 words. Thus our reverberator reduces the memory requirement by almost 50 %. The amount of arithmetic operations in the proposed and the compared structures are 51, 182 and 52 per output sample. This implies that our reverberator is far more efficient than an FDN yet also exhibits a faster reflection density buildup. When compared to Moorer's structure the amount of computation is equal but the reflection density in our structure is much greater. Due to the fact that the total delay line length of the proposed structure is shorter than in the compared ones the theoretical modal density

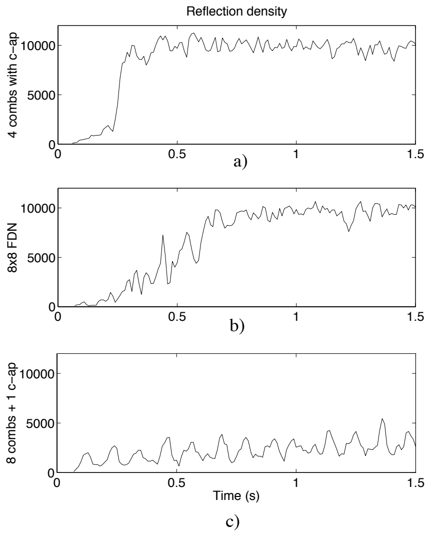

Figure 5: The reflection densities of our reverberator with four delay lines (a), and an 8x8 FDN reverberator (b), eight parallel comb filters connected in series with a single comb-allpass filter (c).

is reduced [7]. However, by choosing the delay line lengths properly the response can be kept free of perceptual coloration.

4 Simulation of the Measured Room Impulse Response

Fig. 6 presents the whole procedure of rendering stereophonic digital reverberation from a measured binaural room impulse response (BRIR). The different phases are described below.

4.1 Measurement and Analysis of the Room Impulse Response

The measurement of the simulated hall was done both monaurally and binaurally. The monaural response was used for octave band-filtered reverberation time calculation and the binaural response from the ears of a dummy head was used for obtaining incoherent early reflections for the simulated reverberation. We used an omnidirectional sound source for exciting the hall with pseudorandom noise, and the impulse response was obtained by deconvolution between the signals at the source and the receiver.

The analysis of the room impulse response was divided into analyzing the binaural early reflections and the frequency-dependent reverberation time in octave bands between 125 Hz-8 kHz. The reverberation time was a result of analyzing the average of monaural responses measured in three positions in the hall. The early reflections were analyzed from the individual reflection peaks in the early part of the response by detecting the delays and gains of the reflections during the first 100 ms.

The proposed structure				FDN and Moorer's reverberator		
delay	g	b	a	delay	g	b
2958+329	0.74	0.34	0.5	3287	0.71	0.38
4011+446	0.66	0.45	0.5	3824	0.67	0.43
4392+488	0.64	0.48	0.5	4250	0.64	0.48
4465+496	0.63	0.49	0.5	4562	0.62	0.50
				4745	0.61	0.52
				4832	0.60	0.53
				4946	0.60	0.53
				4961	0.60	0.54
$A(z)$ of the Moorer rev.				265	$a = 0.7$	

Table 1: Delay line lengths of the studied reverberators in samples at 44.1 kHz frequency, and the coefficients of the filters $H_k(z)$ and $A_k(z)$. The two numbers in our structure indicate the delay line lengths and the comb-allpass filter delay lengths.

	Arithmetic operations
Our structure	$11.5N + 3$
4 delay lines	51
8 delay lines	97
nxn FDN	$2N^2 + 7N - 2$
4x4	58
8x8	182
Moorer	$6N + 4$
4 delay lines	28
8 delay lines	52

Table 2: The amount of summing and multiplication operations of our reverberator compared to FDN and Moorer reverberators.

The reverberation time RT60 was calculated in octave bands to be used for estimating the parameters of the late reverberation producing part of the filter. The response was first filtered into 7 octave bands and the reverberation time was calculated by the backward integration method originally suggested by Schroeder [8].

To yield a practical number of early reflections for real-time processing reflection peaks were combined to approximately 40 discrete reflections for both channels within 100 milliseconds from the direct sound. From a single tapped delay line the early reflections are taken out separately to each output channel. The gains $e_{R,k}$ and $e_{L,k}$, and the delays $D_{R,k}$ and D_L, k of the k^{th} reflection were obtained by combining the reflections in the early part of the response. The obtained early reflections were low-pass filtered to simulate the air and reflection surface material absorption which is a function of frequency.

The delay lengths of the IIR module were defined according to the duration of the early reflections so that the first outputs from the late reverberation unit overlap with the early reflections. The lengths of the other delay lines are kept within a close range from the first one to obtain a good resonance density at the low frequencies. According to Schroeder's suggestion the ratio of the longest and the shortest delay length is kept smaller than 1.5:1 and the lengths mutually incommensurate to avoid superposition of reflections in the time domain and clustering of the resonances

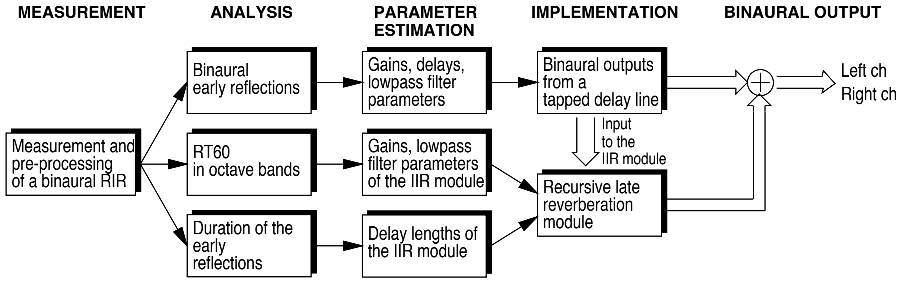

Figure 6: A block diagram of the room response simulation process.

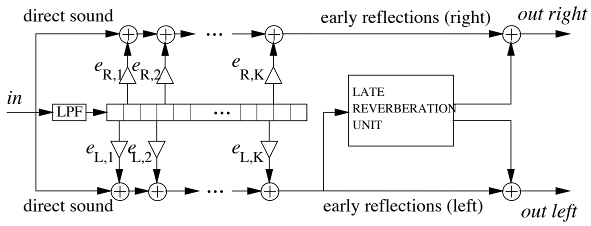

Figure 7: The structure of the reverberator where the early reflections are fed to an IIR late reverberation module from which incoherent outputs are taken for both channels.

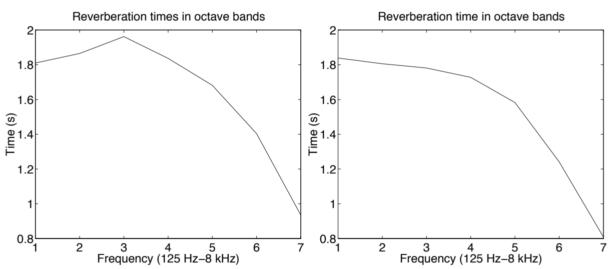

Figure 8: The octave band reverberation times of the original and the artificial responses.

in the frequency domain [1]. Also care has been taken that the total length of the delay lines is at least 0.25·RT60 as suggested in [4].

4.2 Implementation and Results

In Fig. 7 the structure of the whole reverberator including the delay line for the early reflections and the late reverberation IIR block is shown. The incoherent responses for two channels are derived from the binaural early reflections and incoherent outputs from the late reverberation filter [4]. In Fig. 8 the reverberation time in octave bands of the measured and artificially produced responses are illustrated. It can be seen that the reverberation times match quite well. The fit can be further improved by using higher order filters $H_k(z)$ in the structure.

5 Conclusions and Further Work

In this presentation a novel structure for reverberation modeling was suggested. In addition we have performed measurements and analysis of room impulse responses to obtain parameters for early reflections and late reverberation. The proposed reverberator was compared to other known structures and found to be efficient both from the point of view of computational load and memory requirements.

Extensions to our model include directional processing of the direct sound and early reflections using Head-Related Transfer Functions (HRTF) [9], and tone correction by adding a filter which models the initial frequency response of a measured room impulse response [4].

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