Digital Signal Processing Techniques for Non-exponentially Decaying Reverberation

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Abstract

In this paper we show several digital signal processing techniques that can be used for non-exponentially decaying artificial reverberation. Traditional recursive filter techniques used for simulating the diffuse part of reverberation produce an exponentially decaying reverberation. We show how traditional reverberation algorithms can be modified and combined to create non-exponentially decaying reverberation. The techniques presented here can be used for interesting musical effects and speech enhancement. As an application example, a real-time system using the Motorola DSP56002 digital signal processor is presented.

1 Introduction

Artificial reverberation has been used for dozens of years to add reverberation to studio recordings. The first digital reverberator was developed by Schroeder [1] over 30 years ago. His solution was based on a recursive structure constructed of parallel comb filters in series with two allpass filters. Later, Moorer [2] improved Schroeder's algorithm and studied the definition of filter parameters so that more natural-sounding reverberation could be created. A further extension of the parallel comb filter approach was presented by Väänänen *et al.* [3] who included allpass filters inside the delay-line loops to obtain more rapid build-up of the impulse response.

Nowadays, the most sophisticated structure to create diffuse reverberation is the Feedback Delay Network (FDN) [4, 5, 6]. The main advantages of the FDN are the diffusion of reflections and an increasing reflection density versus time. In addition, the FDN produces uncorrelated delay-line outputs which still have energy from the modes of all the delay lines, which enables the usage of high-quality reverberation also in multi-channel sound reproduction. A review of digital reverberation algorithms was recently presented by Gardner [7].

There are not many publications about reverberators for producing musical effects rather than for simulating natural spaces, although digital reverberators are commonly used for interesting effects. Recently, Dattorro [8] discussed the design of such reverberators.

The recursive filter structures used in traditional reverberation algorithms typically produce an exponentially decaying tail. Similar behavior can be observed also in real concert halls, but there is no need to constrain the artificial reverberation effects only to this kind of response.

There are applications where the shaping of the reverberation tail is desired such as sound effects in music recording field. The shaped reverberation is a particularly useful effect for drums and other percussion instruments. Typically used reverberation for percussion has a fast build-up and a sudden decay. The sudden decay can be useful also for speech and vocal tracks. The reverberation increases the energy of the speech signal and sudden decay ensures that it does not degrade the speech intelligibility.

In Section 2 of this paper we review potential techniques for producing reverberation with a non-exponential tail. We are focusing on linear and time invariant techniques and the main emphasis is on techniques suitable for real-time applications. New approaches include a combined reverberator structure and a modified comb filter. A real-time signal processor implementation of one of the proposed techniques is described in Section 3. Section 4 concludes the paper.

2 Techniques

The shaping of the reverberation is a much easier task if the signal processing is done off-line. The off-line processing is an attractive solution for post production, but certainly not for live presentations. In the following sections we will discuss efficient techniques suitable also for real-time processing.

2.1 Convolution

The most straightforward way to create arbitrarily shaped reverberation is to use the standard convolution operation for whole reverberation. The convolution is more efficiently performed in the frequency domain. The fast convolution operates on data in large blocks causing considerable input-output delay.

The zero-delay convolution or compromise between efficiency and delay can be achieved by splitting convolution into smaller separate convolutions as proposed by Gardner [9]. Fast convolution techniques make it possible to use very long filters up to several seconds [10]. However, the algorithm is fairly complex and the required memory for long reverberation is much larger than available in a typical DSP platform.

A hybrid solution can be used when a fast build-up is desired. The initial attack part is implemented with a short FIR filter and the latter decaying part of the reverberation can be implemented with a traditional reverberation algorithm.

2.2 Block processing

Non-causal processing is not directly applicable to real-time applications, but can be approximated with block processing. The technique has been used for realization of linear-phase IIR filters [11,12]. The fundamental problem with the concept is the truncation noise that is caused by segmentation. The truncation error can be minimized using larger data blocks or rearranging the polynomials in the IIR filter [12]. However, the length of the impulse response used in reverberation filters makes the approach less attractive for reverberation use and is not discussed any further here.

2.3 Combined systems

Traditional recursive structures can be connected as shown in Figure 1. The main idea is to use reverberation units with slightly different parameters. First they have almost identical output and after that the difference is increasing. Figure 2 presents an example output using two FDN structures with slightly different reverberation times. The reverberation times are chosen long enough to make the effect more visible. The reverberation algorithms that have an increasing echo density and long reverberation time can benefit from this technique. Echo density of the reverberation algorithm can be made higher before the amplitude reaches the maximum level. Additional shaping is possible by adjusting g_1 and g_2 parameters (Figure 1).

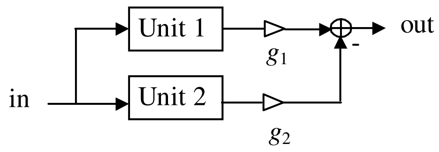

Figure 1: The block diagram of combined systems.

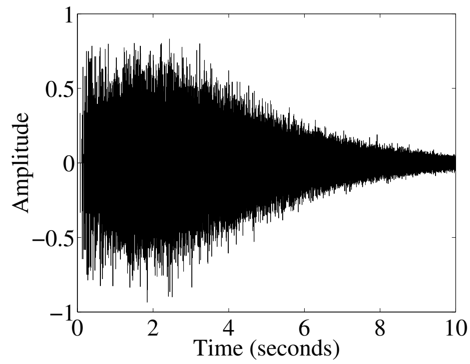

Figure 2: The impulse response of combined FDN filters.

2.4 Modified comb filter

The comb filter is the basic building block of many reverberation algorithms [7]. In this section it is shown how the comb filter can be modified for producing different decay curves.

The amplitude of the comb filter output can be adjusted using one additional FIR filter before the comb filter. If the modification is synchronized to the length of the delay line used in the comb filter we can change the amplitude of the circulating impulse in the delay line, or cancel it completely. The output is piecewise an exponential decaying signal. An example output is shown in Figure 3. A similar modification technique can be used also for allpass filters, but the output of the resulting filter is not truly allpass.

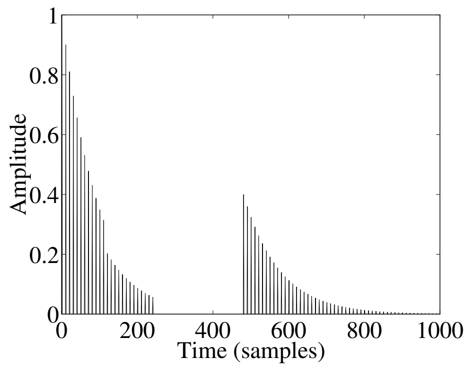

Figure 3: The impulse response of the comb filter modified by the FIR filter.

If two different exponential decay curves are desired another comb filter can be added to the structure. A straightforward implementation includes one FIR filter and two comb filters connected in parallel as shown in Figure 4. The upper comb filter has decay parameter g_1 and the FIR filter cancels this after $M{\cdot}K$ samples and initiates the lower comb filter with a different decay parameter g_2. The presented structure requires that $g_1 \neq 0$ and for stability reasons $|g_1|<1$ and $|g_2|<1$.

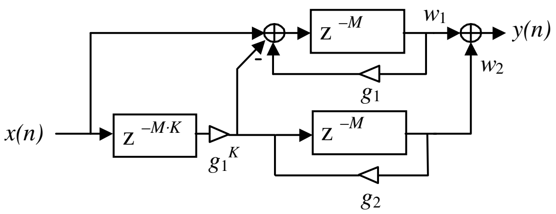

Figure 4: Filter structure with two comb filters.

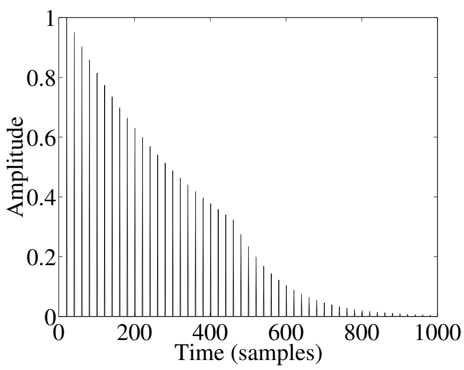

Figure 5: An example output using the filter structure presented in Figure 4. The delay M was 20 samples in both comb filter and the gain coefficients g_1 and g_2 were 0.95 and 0.85 respectively. The coefficient K was 22.

The difference equations for the structure presented in Figure 4 is as follows:

$$y(n) = w_1(n) + w_2(n) \quad (1)$$

$$w_1(n) = x(n-M) - g_1{}^K x(n - MK - M) + g_1 w_1(n-M) \quad (2)$$

$$w_2(n) = g_1{}^K x(n - MK - M) + g_2 w_2(n-M) \quad (3)$$

$$W_1(z) = \frac{X(z)\cdot z^{-M} - g_1{}^K \cdot X(z)\cdot z^{-(M\cdot K+M)}}{1 - g_1 \cdot z^{-M}} \quad (4)$$

$$W_2(z) = \frac{g_1{}^K \cdot X(z)\cdot z^{-(M\cdot K+M)}}{1 - g_2 \cdot z^{-M}} \quad (5)$$

Finally, the transfer function is obtained:

$$H(z) = \frac{z^{-M} - g_1{}^K \cdot z^{-M\cdot K-M}}{1 - g_1 \cdot z^{-M}} + \frac{g_1{}^K \cdot z^{-M\cdot K-M}}{1 - g_2 \cdot z^{-M}} \quad (6)$$

When the feedback parameter of the comb filter is set to unity gain the filter preserves the energy and we get an integrating comb filter. A similar technique has been used in the context of integrating FIR filters [13,14]. Using one, two or three integrators the impulse response is made of step, line and quadratic segments respectively. Figure 6 shows an example second-order integrating filter structure. Two integrators are in cascade with a sparse FIR filter that controls the decay curve. The filter is designed to give a typical ADSR (attack, decay, sustain, release) curve. The corresponding impulse response is presented in Figure 7. Several integrating comb filters (first-order, second-order etc.) can be also connected in parallel. This leads to a generalized polynomial filter. The polynomial filter can be used to generate an arbitrary polynomial curve. The structure can be controlled with an FIR filter so that the polynomial coefficients can be changed freely and a piecewise polynomial impulse response is obtained.

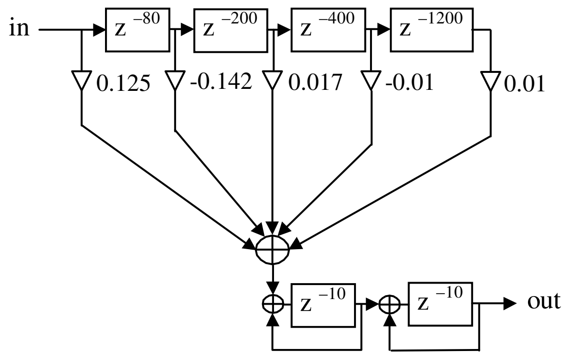

Figure 6: An example structure of second-order integrating comb filter.

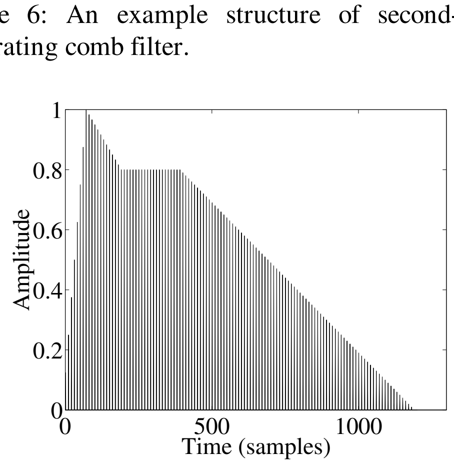

Figure 7: The impulse response of the filter shown in Figure 6.

3 Real-time system

The real-time reverberation system based on integrating comb filters was developed to verify the simulated results. The structure presented in Figure 8 was implemented on a Motorola 56002 DSP. The structure resembles earlier reverberation algorithms presented by Schroeder and Moorer. The second-order integrating comb filter substitutes the conventional comb filter. Using the presented structure the reverberation tail can be modified to non-exponentially decaying shape as can be seen from an example output presented in Figure 9.

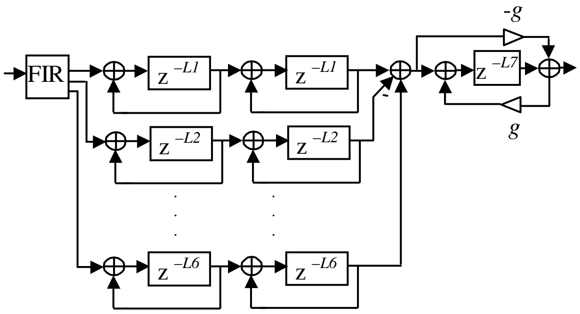

Figure 8: Filter structure used in real-time reverberation system.

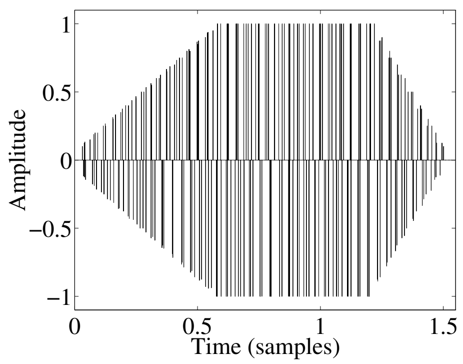

Figure 9: An example impulse response from the real-time system (g=0).

4 Conclusion

We have presented several techniques for non-exponentially decaying reverberation without any nonlinear or time-varying filtering. The proposed techniques appear to be useful for generating a wider range of effects than traditional reverberation. Especially the integrating comb filter concept seems to provide an efficient and flexible way to shape the reverberation tail. We have also presented a real-time system using the Motorola 56002 DSP. The proposed new structures require more detailed analysis of their properties and investigation of their possibilities. These questions will be addressed in future research.

5 Acknowledgment

Part of this work has been financed by the Academy of Finland.

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