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DIGITAL SOUND EFFECTS ECHO AND REVERB BASED ON NON-EXPONENTIALLY DECAYING COMB FILTER

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ABSTRACT

This paper presents algorithms of two digital sound effects based on the NEDCF (Non-Exponentially Decaying Comb Filter). The first part of the paper deals with the description of a NEDCF structure and with the algorithm of the digital sound effect of Echo type with easily controllable parameters. The second part describes an extension of the previous algorithm for obtaining a new algorithm of the multi-channel digital reverberation. The reverberation algorithm presented in this paper produces an impulse response with a controllable decay curve, reverberation time and frequency dependent reverberation time. The decay curve can consist of an arbitrary number of increasing or decreasing linear segments which enable the creation of an interesting reverberation effect.

1. INTRODUCTION

Echo and Reverb are the most common and very important digital sound effects, which are frequently used by many of music composers. This paper describes new types of such effects.

The main advantage of the Echo and Reverb algorithms described herein is the possibility to easily set an unconventional shape of the decay curve which can consist of constant, linear, quadratic (and so on) segments. The goal of the introduced reverberation algorithm is not an authentic simulation of acoustic properties of the real auditory space. This reverberation algorithm is to use to the unconventional effecting of musical samples and thus to enhance the possibilities of expression in music composing. The impulse response produced by this algorithm does not have an exponentially decaying envelope, as it is needed in the classical reverberation algorithms, but it consists of an optional number of linear segments with adjustable time duration. The time duration and the number of segments determine the whole reverberation time. In the presented algorithm, an analogy of the frequency dependent reverberation time known from the classical reverberation algorithms is also implemented. However, this implementation has a little different meaning in comparison to the classical algorithms.

2. GENERAL NEDCF STRUCTURE

If we connect two common integrating comb filters (i.e. comb filters with unity feedback gain) into series as depicted in Figure 1, the impulse response of such structure will be linearly increasing (see Figure 2). You can see uniformly spaced (N-1 zeroes) impulses with linearly increasing height.

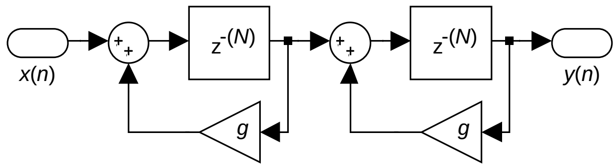

Figure 1 - Two integrating comb filters in series (g = 1)

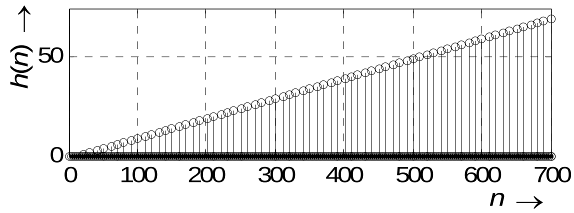

Figure 2 - Impulse response of two integrating comb filters in series

Three comb filters in series produce the same time distribution again, but the height of impulses increases quadratically with time. This is true after the decay of initial conditions of each delay z^{-N}. In this case they have been set to zero. It is clear that the described serial combinations of two or three comb filters are unstable, but it is possible to design a structure that will control the shape of impulse response envelope and so obtain a system called Non-Exponentially Decaying Comb Filter (Figure 3). The main idea is to divide the impulse response in time to get its parts. Therefore the whole impulse response envelope will be piecewise constant, linearly, quadratically (and so on) increasing or decreasing (Figure 4). The shape of impulse response is controlled with the help of a preliminary FIR filter which composes of delay line in series with the coefficients matrix.

Unfortunately, the general NEDCF structure is relatively complicated for higher order of integrating comb filter as well as a general design of matrix **A**, which is beyond the scope of this paper.

3. SIMPLIFIED NEDCF STRUCTURE

The structure of a simplified NEDCF, first presented in [8], consists of two integrating comb filters in series with the preliminary control FIR filter which consists of L delay elements (Figure 5). The envelope of the NEDCF impulse response is thus composed of L linear segments.

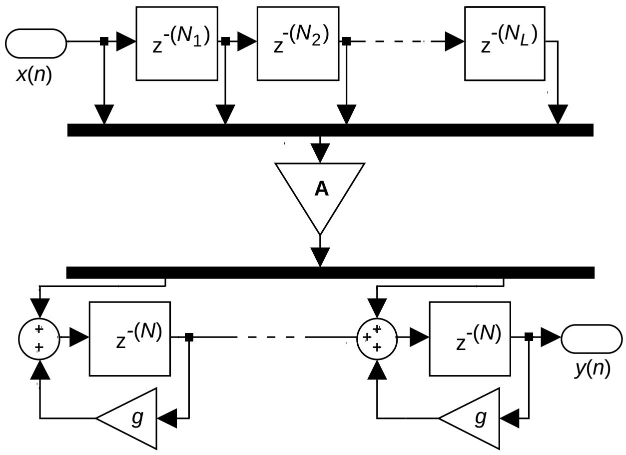

Figure 3 - General NEDCF structure

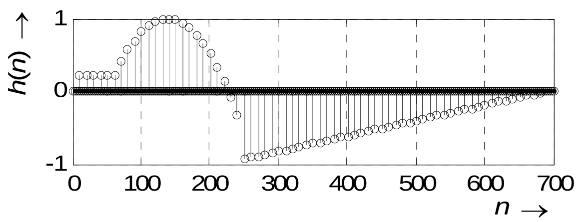

Figure 4 - Possible shape of the third-order general NEDCF structure impulse response

The value of each delay element in the control FIR filter and in each integrating comb filter must be mutually commensurable. Only in this way the stability of the whole system can be achieved. Otherwise impulses delayed by N_1, $N_1 + N_2$,..., $N_1 + N_2$ +,...,+ N_L and weighted by g_1, g_2 , g_3,..., g_{L+1} coefficients do not coincide with impulses circulating in N delaying IIR integrating comb filters, thus they cannot be mutually added up or subtracted from. In this case the output of NEDCF is composed mainly of the impulse response of the second-order integrating IIR comb filter; the system is not stable.

For these reasons it is possible to adjust the time duration of each linear segment of the impulse response only in discrete steps of NT_s size, where T_s is the sampling period.

To compute coefficients g_1, g_2, g_3,..., g_{L+1} it is possible to derive the following equations:

$$g_1 = \frac{\mathbf{e}_1 N}{N_1}, \qquad (1)$$

$$g_n = -\sum_{k=2}^{n} g_{k-1} + \frac{(\mathbf{e}_n - \mathbf{e}_{n-1})N}{N_n} \quad \text{for } n = 2,3,...,L, \qquad (2)$$

$$g_{L+1} = -\sum_{k=1}^{L} g_k , \qquad (3)$$

where L is the desired number of linear segments of impulse response, $\mathbf{e}$ is the vector of envelope values in its breaking points, N is the delay of integrating comb filters, and N_1,..., N_L are the values of the delays in the control FIR filter. The equations given above are valid only in the event that all the impulses of the NEDCF impulse response have the same polarity (the feedback coefficient in integrating comb filters is equal to positive one). If they have not, i.e. in the case of the odd number of impulses between the adjoining breaking points, it is necessary to reverse the polarity of the given feedback coefficient.

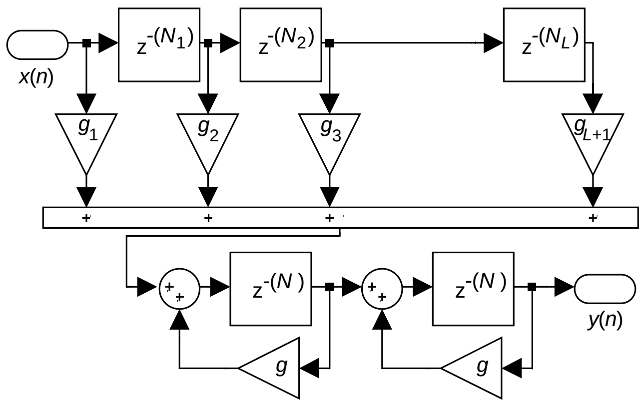

Figure 5 - Simplified NEDCF structure

4. DIGITAL SOUND EFFECT ECHO BASED ON NEDCF

The echo algorithm introduced in this short chapter provides an enhanced and simplified possibility of setting an interesting shape of the impulse response. Its structure is very simple (see Figure 6) and consists of M NEDCFs, M delays and M gains, where M is the number of channels. The delays placed in channels are to provide the possibility of setting a different delay of echoes in each channel and the gains enable setting their loudness.

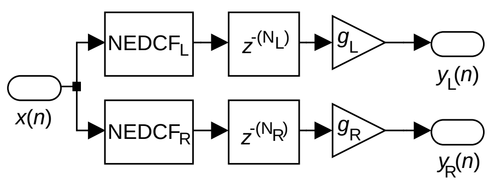

Figure 6 - Structure of dual channel digital sound effect Echo based on NEDCF

5. MULTI-CHANNEL DIGITAL REVERBERATOR WITH NON-EXPONENTIAL DECAY

The structure of multi-channel digital reverberator with non-exponentially decaying impulse response comes from the structure of Moorer's reverberator [1] [3] [7]. Non-exponential decay of impulse response can be achieved by replacing classical comb filters by the comb filters with non-exponentially decaying impulse response (NEDCF) [8].

5.1 Whole structure block diagram

The whole structure block diagram of a single channel reverberator with non-exponential decay is depicted in Figure 7. The multi-channel version can be easily obtained by a simple extension, as shown further in the text.

The output of the reverberator is the sum of three signals. First of them is the direct signal $y_d(n)$ from the input of the reverberator. This direct signal is weighted by the g_d coefficient. The second signal $y_e(n)$ consisting of early reflections is delayed by E samples and weighted by the g_e coefficient. The last signal is a subsequent reflections signal $y_s(n)$ weighted by the g_s coefficient and delayed by S samples.

The weighting coefficients allow setting the desired level of individual signals in their total sum independently of each other. The delay elements then enable the setting of the initial time of early and subsequent reflections.

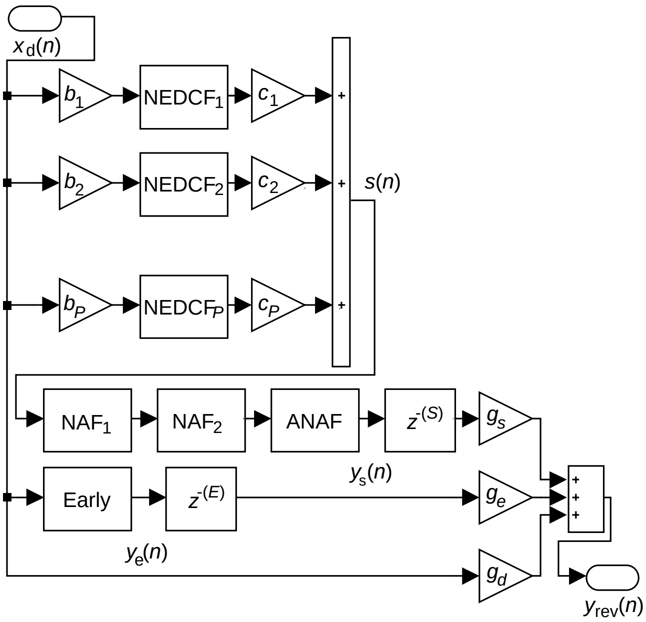

Figure 7 - Whole structure block diagram of the reverberator

5.2 Early reflections block

The general concept of the early reflection simulation proposed by J. A. Moorer [7] is still in use with some improvements only. The early reflections should be frequency-dependent and of proper density. These two parameters are set with the help of the structure depicted in Figure 8. As we can see, the Early Reflections Block consists of a delay line, usually called the "tapped delay line" (TDL), completed with the Nested All-Pass Filter with the frequency dependent output (Absorbent Nested All-Pass Filter - ANAF). The ANAF can be obtained by inserting a filter with the $H_e(z)$ transfer function into the classical Nested All-Pass Filter, as shown in Figure 9.

5.3 Subsequent reverberation block

Another part of the subsequent reverberation block is Nested All-Pass Filter (NAF). It is almost the same structure as in Figure 9 only without the absorbent filter.

The few equations below define the transfer function $H_{ab}(z)$ of the Absorbent Nested All-Pass Filter. The transfer function of a common All-Pass filter is defined as

$$H_a(z)=\frac{-g_1+z^{-N_i}}{1-g_1 z^{-N_i}}. \tag{4}$$

The transfer function $G(z)$ describes all structures in the direct path inside the feedback and feedforward paths:

$$G(z)=H_a(z)z^{-N_o},\ H_e(z)=\frac{G_n(z)}{G_d(z)}, \tag{5}$$

where

$$G_n(z)=\left(-g_1+z^{-N_i}\right)z^{-N_o}H_e(z),\ G_d(z)=1-g_1 z^{-N_i}, \tag{6}$$

thus $H_{ab}(z)$ is given by

$$H_{ab}(z)=\frac{-g_2G_d(z)+G_n(z)}{G_d(z)-g_2G_n(z)}, \tag{7}$$

$$H_{ab}(z)=\frac{-g_2+g_1g_2z^{-N_i}-g_1H_e(z)z^{-N_o}+H_e(z)z^{-(N_i+N_o)}}{H_e(z)-g_1H_e(z)z^{-N_i}+g_1g_2z^{-N_o}-g_2z^{-(N_i+N_o)}}. \tag{8}$$

This configuration of the early reflections block makes possible to control the early reflections density by the g_1, g_2 coefficients in the ANAF structure. The nested filter allows controlling their frequency dependence. The values of the h coefficients of the delay line are given by

$$h_i=(-1)^i\cdot\mathbf{e}_i \quad \text{for } i = 1,2,...,L, \tag{9}$$

where $\mathbf{e}$ is the vector of the envelope values in its L breaking points.

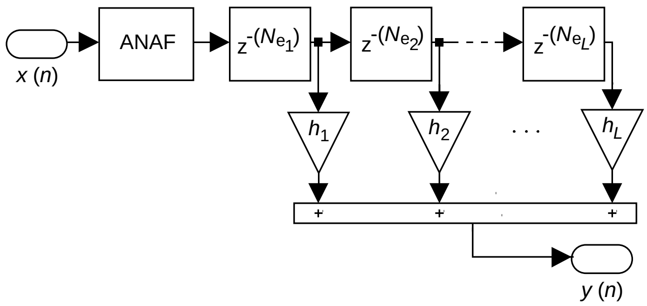

Figure 8 - Early reflections block

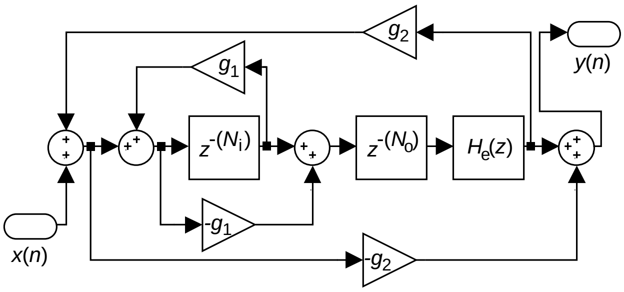

Figure 9 - Structure of Absorbent Nested All-Pass Filter

The basic advantage of Nested All-Pass Filter is, in comparison with the common Schroeder's All-Pass Filter, greater impulse density on the output. Moreover, this impulse density increases with time, similarly as in the real auditory spaces. The purpose of these structures is to substantially increase the density of impulses on the output of the bank of the parallel NEDCFs.

The last part of the subsequent reverberation block is the ANAF structure, which is identical as in the block of early reflections. Besides its further increasing of the impulse density, its function consists in the implementation of the frequency dependent reverberation time in the subsequent reverberation. In this place its implementation shall be discussed.

The definition of the frequency dependent reverberation time of a non-exponentially decayed reverberation is different from the definition of reverberation with exponential decays. This is a synthetic, unnatural type of reverberation for which we cannot find any analogy with the natural reverberations. Therefore we can conceive the frequency dependent reverberation time as we like. However, if we use our experience from the exponential reverberation, then the increasing linear segment of the impulse response envelope must result in the linear increase of the given frequencies and vice versa. The question is, which frequencies should be increasing or decreasing and to what extent. If we try to simulate this concept of frequency dependent reverberation time, the result is absolutely unusable. What is more, the real-time implementation requires the time variant system that presents a significant complication. The proposed technique, though not quite correct, provides the best results as for the listening aspect.

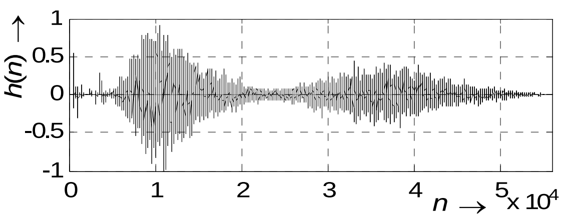

Figure 10 - Possible shape of the overall reverberator impulse response

5.4 Extension of the single-channel reverberator version

In the case of the multi-channel version of this reverberator, the vectors **b**, **c** (see Figure 7) come into matrixes. Let M denote the number of input channels, while K the number of output channels and P the number of NEDCFs in the parallel filter bank. The matrixes **B** and **C** can be expressed as

$$\mathbf{B}=\begin{bmatrix} b_{11} & b_{12} & \cdots & b_{1M} \\ b_{21} & b_{22} & & b_{2M} \\ \vdots & & \ddots & \vdots \\ b_{P1} & b_{P2} & \cdots & b_{PM} \end{bmatrix}, \mathbf{C}=\begin{bmatrix} c_{11} & c_{12} & \cdots & a_{1K} \\ c_{21} & c_{22} & & a_{2K} \\ \vdots & & \ddots & \vdots \\ c_{P1} & c_{P2} & \cdots & a_{PK} \end{bmatrix} \quad (10)$$

The transfer functions matrix of the parallel filter bank of P NEDCF structures will be denoted as $\mathbf{D}(z)$, the vector of signals on the output of the system will be denoted as $s(z)$ and the vector of signals on the input of NEDCF parallel filter bank will be denoted as $x(z)$. Using the Z-transform we can write

$$\mathbf{D}(z)=\begin{bmatrix} H_{\mathrm{NEDCF}_1}(z) & 0 & 0 \\ 0 & \ddots & 0 \\ 0 & 0 & H_{\mathrm{NEDCF}P}(z) \end{bmatrix}, \quad (11)$$

$$s(z)=\begin{bmatrix} s_1(z) \\ s_2(z) \\ \vdots \\ s_K(z) \end{bmatrix}, x(z)=\begin{bmatrix} g_{s_1} x_{d_1}(z) \\ g_{s_2} x_{d_2}(z) \\ \vdots \\ g_{s_M} x_{d_M}(z) \end{bmatrix}, \quad (12)$$

where g_{s1}, g_{s2} ,..., g_{sM} are the input weighting coefficients, and $H_{NEDCF1}(z)$, $H_{NEDCF2}(z)$,...,$H_{NEDCFP}(z)$ are the NEDCF transfer functions. The vector of signals on the input of NEDCF parallel filter bank is then given by

$$s(z)=\mathbf{C}^T\cdot\mathbf{D}(z)\cdot\mathbf{B}\cdot x(z). \quad (13)$$

Each component of the $s(z)$ vector is equal to one signal channel on the input of the first NAF structure, i.e., each channel has its own part for the impulse density increasing and for the implementation of frequency dependent reverberation time.

6. CONCLUSION

The new types of Echo and Reverb digital sound effects algorithms have been designed. All structures were simulated in Simulink.

The presented reverberation algorithm has been tested with many kinds of music samples and it is well suitable for using in music composing. Very interesting results can be obtained by processing of short percussive sounds and of voice. The main disadvantage of the introduced reverberation algorithm is a relatively complicated implementation on the signal processors because of its many input parameters. However, the results show an interesting and good quality reverberation with an acceptable digital system loading.

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