Digital Implementation of Artificial Reverberation Algorithms

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DIGITAL IMPLEMENTATION OF ARTIFICIAL REVERBERATION ALGORITHMS

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Abstract

The paper presents the digital implementation of signal processing algorithms that simulate natural concert hall reverberation. It deals with the complete artificial reverberators proposed by Schroeder, Moorer and Gardner. For each reverberation algorithm the implementation was done in two steps: 1. the algorithm is checked using a model in Matlab Simulink; 2. the Verilog code is written and tested. The results prove that the Verilog design is feasible and can be further developed for acoustic improvements of rooms.

Keywords: artificial reverberators, FPGA

1 INTRODUCTION

Natural reverberation is the combined effect of multiple sound reflections within a room [1], [4]. It might be described by the impulse response of the room, obtained as room's time response to a very short burst. A typical impulse response in a reverberating room is shown in Figure 1.

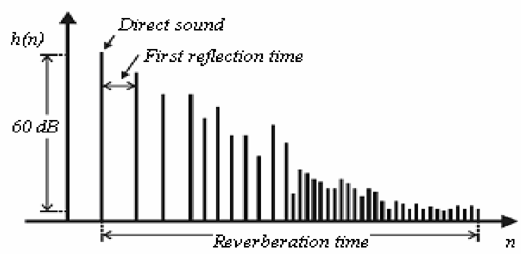

Figure 1 The impulse response of a room

The *reverberation time* is the amount of time it takes for the sound to die away to 60 dB of its original value. The *first reflection time* is the amount of time separating the direct sound from the first reflection. Natural reverberation typically follows a quasi-exponential curve that reaches a peak and decays more or less slowly. Artificial reverberation is generated, manipulated and combined in order to get a stereo high quality sound recording that looks natural. This effect spatializes sounds, thus leading to the illusion of sounds emerging from imaginary environments. From a signal processing point of view, an artificial reverberator is a filter with an impulse response that resembles the impulse response of a concert hall, cathedral, etc. [2].

The digital approach of the reverberation phenomena plays the most important role in the audio signal processors domain [5]. The scope of the digital reverberators is to delay certain frequencies, which will make the sounds reach the listener in different moments of time, adding life and sense of space.

2 ARTIFICIAL REVERBERATORS

The modern technologies bring several solutions for the reverberation problem. Nowadays, the new computers and digital signal processors allow a complicated digital simulation of a natural reverberation.

Schroeder (I) reverberator

The first digital solution of the reverberation was founded by Manfred Schroeder in 1961 and it is based on the comb filter (Figure 2). It consists of a delay cell with an output redirected to the input.

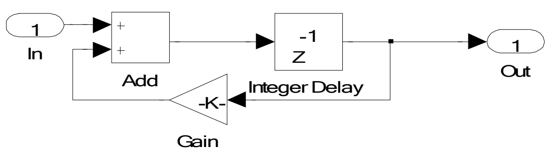

Figure 2 Comb filter

Schroeder noticed that the structure of a comb filter can be modified to get an all-pass filter as shown in Figure 3. Such a filter has a smooth frequency response.

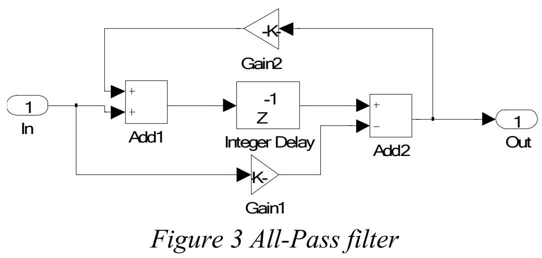

Figure 3 All-Pass filter

Schroeder proposed a reverberator composed of parallel comb filters, series connected with all pass filters, as depicted in Figure 4.

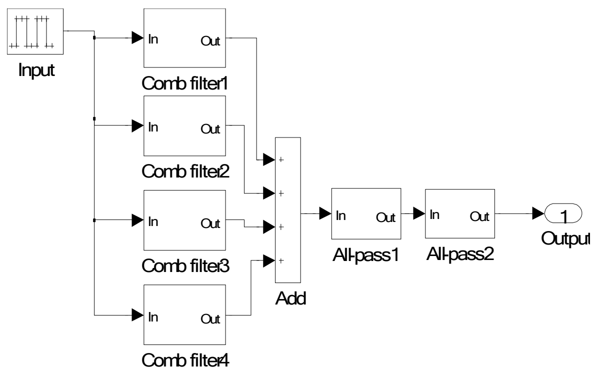

Figure 4 Schroeder first reverberator

The comb filters produce a long reverberant decrease and the all pass filters multiply the number of echoes from the output of the comb filters.

Schroeder (II) reverberator

This structure consists of a chain of 5 allpass filters with different delays, as depicted in Figure 5.

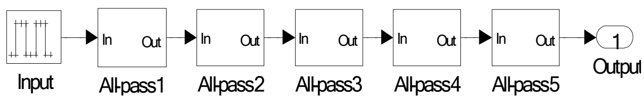

Figure 5 Schroeder second reverberator

Moorer reverberator

Moorer started with the Schroeder reverberator and made serious improvements (Figure 6). The first change was the increase of the comb filters from four to six. It caused longer reverberation times and kept unmodified the modal density. The second change was the introduction of a low pass filter (LPF) in the reaction of the comb filter. The cutoff frequencies of the LPF were based on physical considerations about the air sound absorbtion. By adding a LPF, the reverberation time decreases at high frequencies and the sound looked more real.

Gardner reverberator

It is based on nested all-pass filters, where the delay is replaced by a series connection of a delay and another all-pass filter. Gardner suggested three structures for different size rooms [3]. In Figure 7 the structure of the medium room reverberator is depicted.

3 IMPLEMENTATION

This paper approaches two different implementations: one using the Matlab-Simulink environment and the other using the Verilog hardware description language. The first design represents the preparation step for the second one.

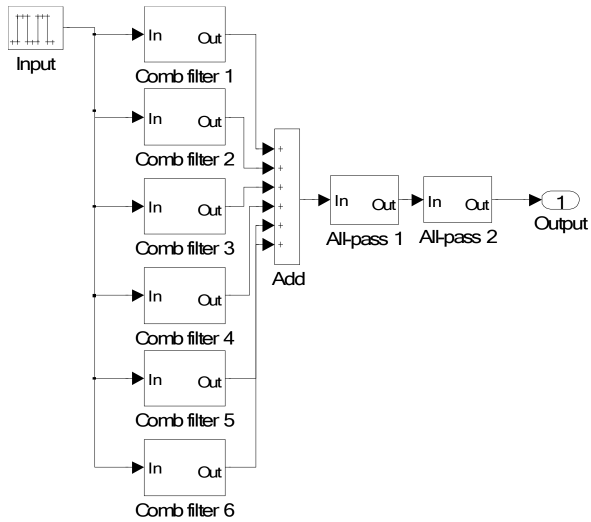

Figure 6 Moorer reverberator

Both implementations use the same input signal: a Matlab modeled impulse. For the Verilog implementation the input data provided by Matlab was transformed into specific files through a special conceived system. These files contain binary data, which characterize the signal at every clock cycle. It must be specified that the input signal was scaled by 1024 because in the digital domain integer calculations were done.

For all the four types of reverberation algorithms, the simulations were run for 4 µs, the necessary time to visualize the results.

The hardware design implied the implementation of the comb, all-pass or low-pass filters with all the specific components at register level. The global signals were: the clock 'clk', working at 50 MHz frequency and the reset 'rst', the system being able to stop in every moment.

For the verification of the conceived system, special Verilog test modules were designed as: a 'clock generator', a 'test bench' (it gives the input signals and simulates different situations that can occur), a 'dump_out' (it takes the output signal and transforms it into a *.data* file) and a 'top_level' (it instantiates all the modules). The compilation and simulation of the test modules were visualized in ModelSim environment.

The main advantage of the digital implementation of these reverberators is that they can be downloaded on development boards having field programmable gate arrays (FPGA) components and used for acoustic signal processing. The FPGAs offer higher processing speed than the common methods involving digital signal processors. Future developments imply the use of floating-point processing types of signals for a superior accuracy.

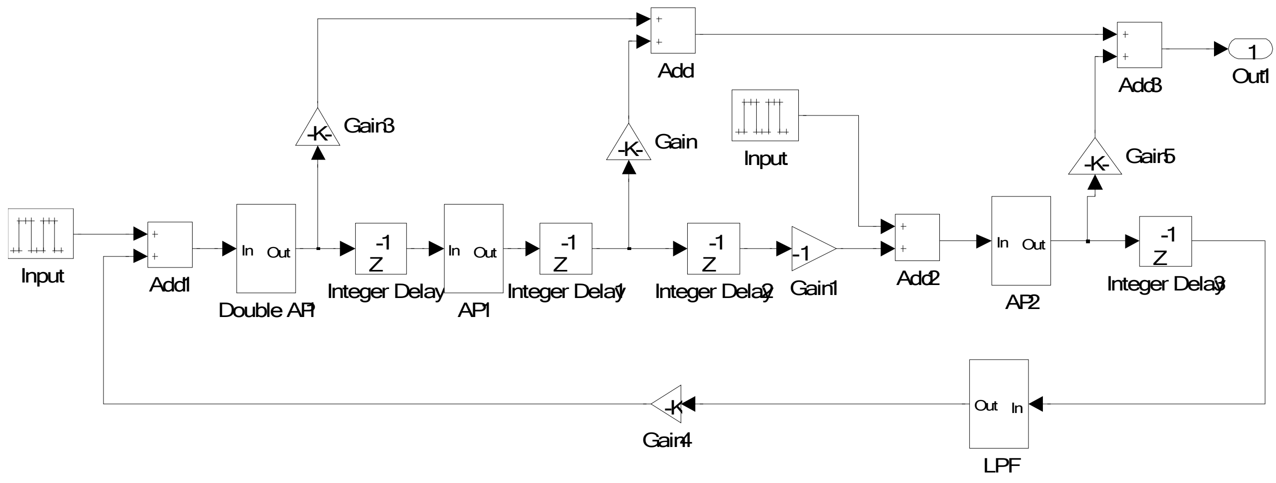

Figure 7 Gardner reberberator

3 EXPERIMENTAL RESULTS

For each artificial reverberator both a Simulink and a Verilog implementations were achieved. The common element is the input signal, a modeled impulse, with the difference that for the digital design it was scaled by 1024 for a better visualisation of the output signal. The digital computations were done with integer numbers, not as in Simulink, where the floating point calculations offered a better precision.

In the Figures 8-11 the output signals obtained for the reverberators in both Simulink and Verilog implementations are presented. The results are almost identical and prove the correctness of the Verilog design.

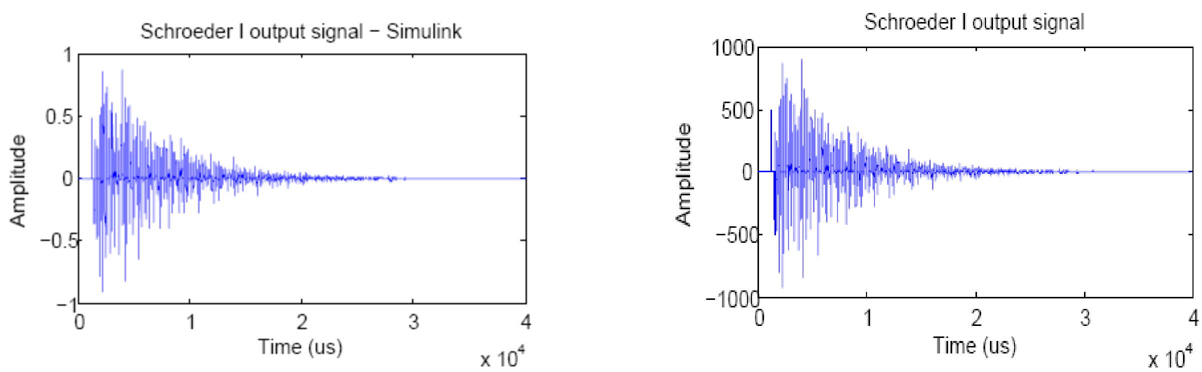

Figure 8 Schroeder I output signal a) Simulink simulation b) Verilog simulation

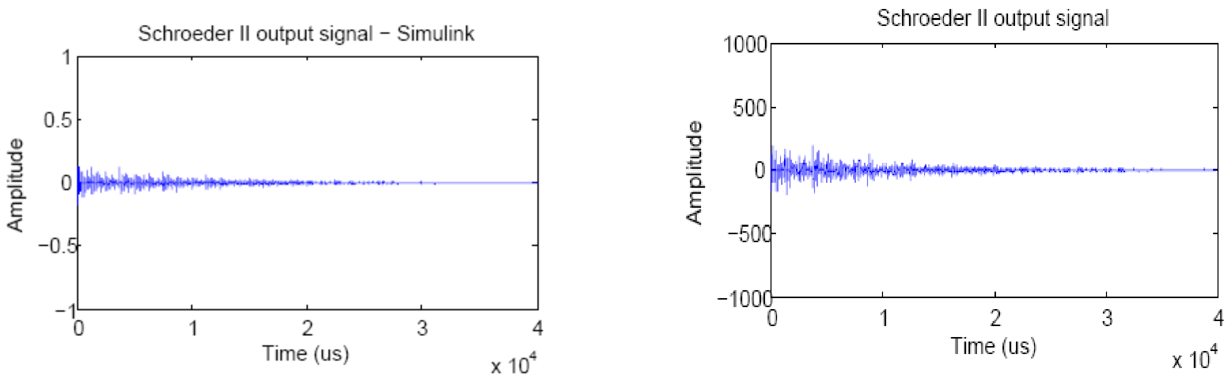

Figure 9 Schroeder II output signal a) Simulink simulation b) Verilog simulation

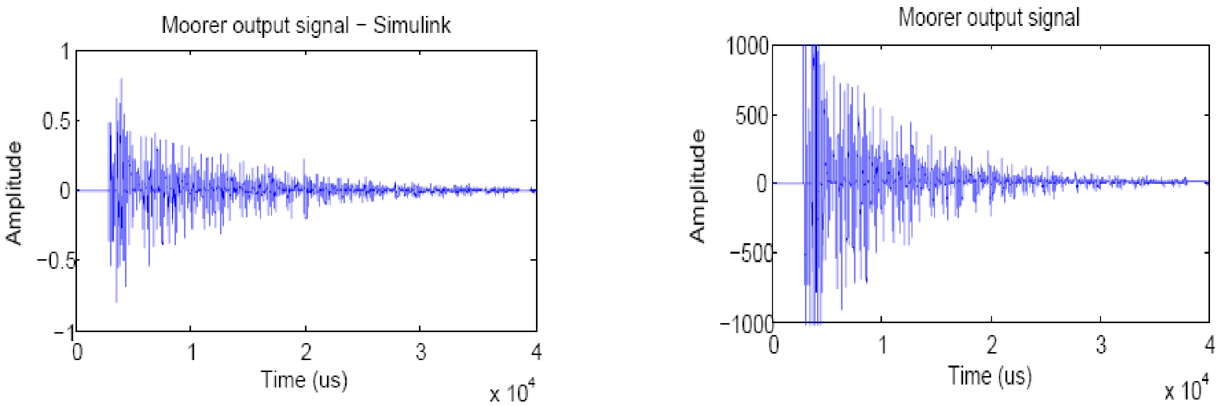

Figure 10 Moorer output signal a) Simulink simulation b) Verilog simulation

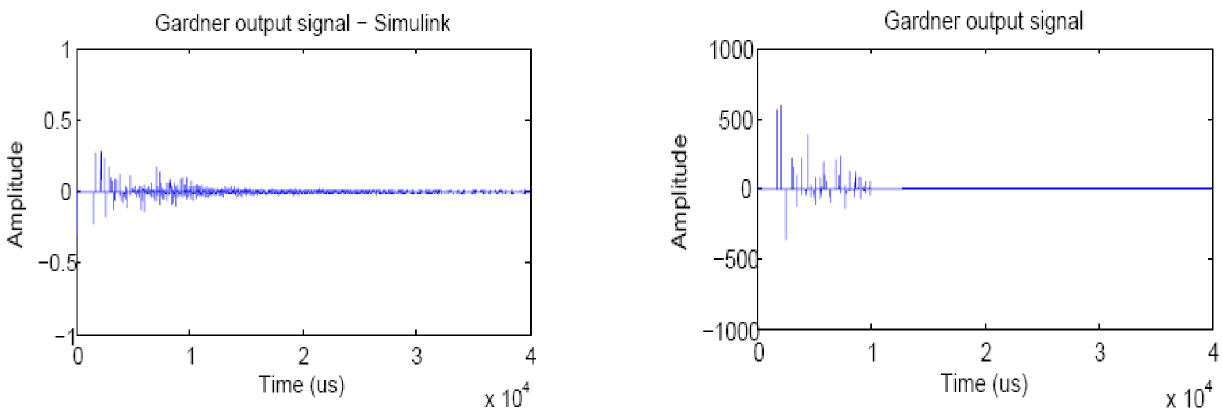

Figure 11 Gardner output signal a) Simulink simulation b) Verilog simulation

4 CONCLUSIONS

The novelty offered by this paper was the digital implementation of four types of artificial reverberators: Schroeder I, Schroeder II, Gardner and Moorer. The first step was the study of the acoustical reverberation phenomen and its parameters. The second step was the description of the blocks in Matlab-Simulink environment, followed by the simulation wih an impulse input. The digital design was implemented on Verilog hardware description language and its verification was realized on ModelSim PE 6.2 Student Edition platform. The same input was applied as in the previous design and the appropriate simulation run. The obtained results in the two different designs were compared with respect to shape and value. The conclusion was that the Verilog implementation is feasible and can be further used for acoustic signal processing for the improvement of acoustic properties of rooms. The code can be the source for a large area of applications because the modern FPGAs contain many resources that support DSP applications, which are optimized for high performance and low power consumption.

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