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OUTPUT FILTER IMPLEMENTATION FOR DIGITAL WAVEGUIDE CLARINET MODEL

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The modification of a dynamic physical clarinet model based on a digital waveguide is presented in the paper. In particular, the influence of the output filter parameters on the signals generated by the model was investigated. As the output filter, the adaptive filter was designed and implemented with various parameter values for different reference signals. The recorded clarinet sound samples and the signals generated by the clarinet model were compared.

Keywords: physical modeling, waveguide synthesis, clarinet model.

1. Introduction

The paper describes new results obtained during the continuation of the previous study [1] in the field of physical modeling of the woodwind instruments. Waveguide synthesis was used for modeling clarinet properties. A model of a clarinet (Fig. 1) consists of a digital waveguide, a reed function and an output filter. A base for modeling process was the simplified model [2]. Several modifications of this model were performed, especially proposed reed function, which represents reed behavior, instead of a simple reed table. Also an output filter characteristic wasn't specified. The main aim of this paper is to present the output filter implementation. A short description of all blocks is presented below.

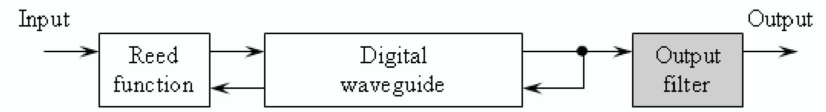

Fig. 1. Waveguide model of a clarinet.

The digital waveguide simulates air propagation inside a clarinet's bore. The reed is responsible for sound generation in the instrument. The main aim of the reed function is to control the air flow. However, some other role of the reed is also considered in this work. The reed function depends on the desired sound loudness and the changes of mouth pressure. For simulations, two different shapes of the reed function are used, and they are adequate for representing a loud and a soft play. They are based on the description of the reed behavior [3]. When playing softly reed function is almost sinusoidal. During loud play behavior of the reed starts to be nonlinear and even a clipping effect may occur (the air flow is zero for part of the cycle).

The output filter determines final properties of sound generated by the clarinet model. In waveguide models with tone hole modeling the filter represents the properties of the output bell of a clarinet. There is popular statement that clarinet emits sound mainly through tone holes. This assumption was verified experimentally. Sound generated by clarinet was recorded using two directional microphones. One was placed in front of bore and tone holes in the distance of 70 cm. The second one was located in front of the bell in the distance of 70 cm. Comparison of this two recordings showed that there are notes for which dominates sound coming from tone holes, and there are ones for which sound emitted from bell is more significant. In the designed model the output filter influences on the emitted sound and cannot by omitted. The main aim of this study was the implementation of the output filter with the use of adaptive filtering.

2. Adaptive filter

The adaptive filter is used to match a waveform of a signal at the input of the filter with a reference signal. The block diagram of the adaptive filter is presented in Fig. 2, where $p(k)$ is the input signal, $w_1, w_2, ..., w_N$ are adjustable weights, and *bias* is a constant component. The proposed filter consists of a tapped delay line, one-layer neural

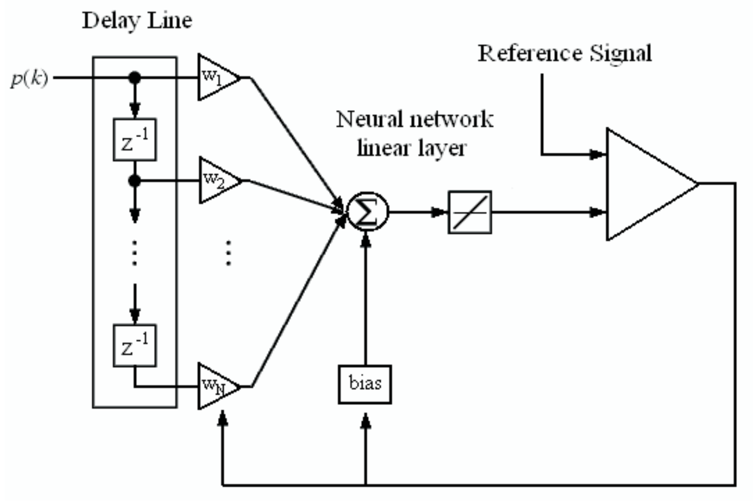

Fig. 2. Block diagram of an adaptive filter.

network and a differential amplifier [4]. A neuron uses a linear transfer function. The delay line consists of various number of digital delay blocks (with transmittance function equal z^{-1}). The number of delay blocks is also an adaptive filter order (called further "delays"). During the adapting process of a specified number of iterations, weights and bias are adjusted. The set of weights and bias values are the final results of the adapting processes. Final values are used as parameters of the output filter.

3. Reference signals

As reference signals, single notes played on clarinet B were used. Recordings were made in an anechoic chamber and in a recording studio. In the anechoic chamber, sounds were played loudly. In the studio, sounds were played both loudly and softly. The time waveforms of the clarinet sounds were registered using the directional microphone located in front of the bell in the distance of 70 cm. For purposes of this work note middle C (262 Hz) was used.

4. Realization

The adaptive filter and the final clarinet model were implemented in the Matlab environment [5]. For the adapting process, the parts of the waveforms were used. Tests on all single notes were performed, but they resulted in poor adaptation and required a very high processor efficiency. Sounds generated by the clarinet are periodical, but the adapting process for only one period of reference and input signal did not provide good results. Tests of different numbers of periods showed that good results can be obtained for three signal periods.

Different numbers of delays and iterations were tested. Number of delays is an order of designed digital filter (number of delay blocks called "z^{-1}" in Fig. 2). First satisfactory results were achieved starting from 170 delays. But only at about 370 delays, the results were satisfactory for both, soft and loud sounds. The number of iterations which guarantees proper adaptation of the input signal to the reference signal is about 2100. The results obtained for less than 700 iterations were not satisfactory. Filter modifications for more than 4000 iterations were also tested, but the improvement was not significant.

5. Results

The output filter, which represents the clarinet's bell properties, was created using the adaptive FIR filter. Relevant realizations were chosen for three different reference signals. The best results were obtained for 370 delays and 2100 iterations. The assessments of the filter's quality were made by comparing the time waveforms and frequency responses of the signal generated by the whole model, to the recorded original signals. The sounds produced by the clarinet model and the recorded using actual instrument were assessed by listening. It was found that the best subjective similarity of sounds occurred for time waveforms being in good agreement, while the comparison of only frequency responses is not sufficient.

The magnitudes of the filters selected as the best for the sounds played loudly are presented in Fig. 3a – for recordings made in a recording studio, and in 3b – for recordings made in an anechoic chamber. Filter characteristics obtained for sounds played softly in the recording studio are shown in Fig. 3c.

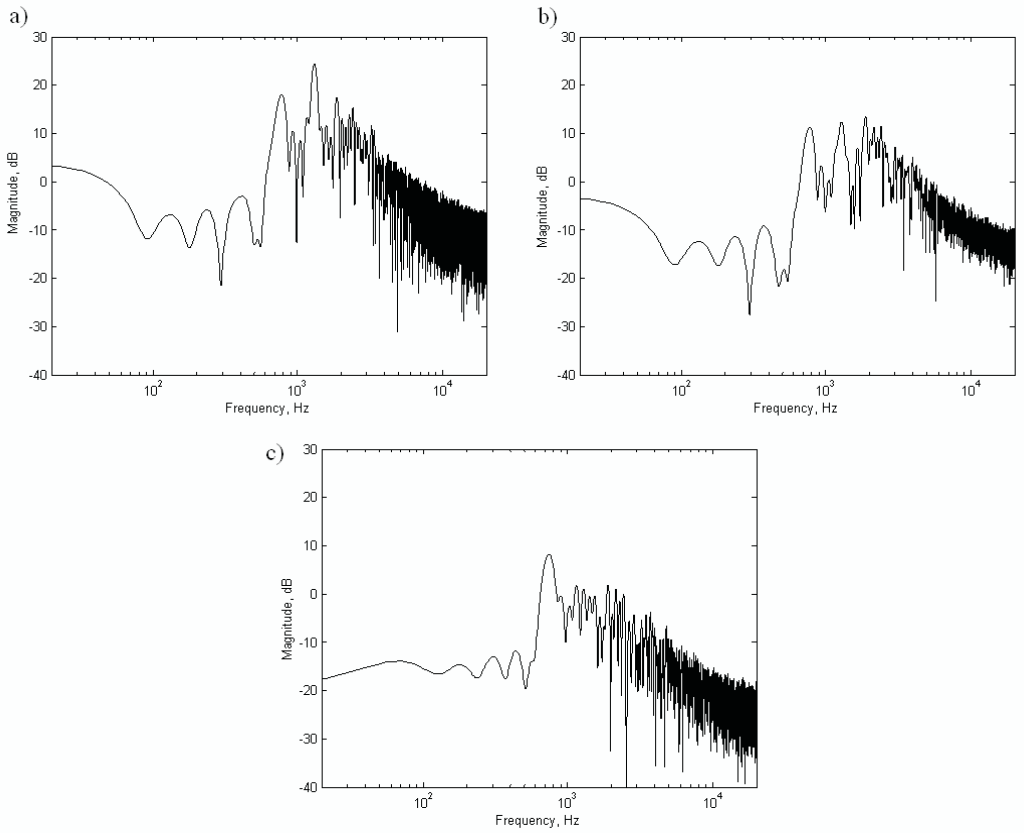

Fig. 3. Filters magnitudes, a) for recording studio-played loudly, b) for anechoic chamber-played loudly, c) for recording studio-played softly.

It may be observed that filter magnitudes achieved for loud sounds are similar, except for higher amplitudes at about 1500 Hz when the waveform registered in a recording studio, is used as a reference signal. The frequency response of the filter designed for sounds played softly is clearly different.

The time waveforms recorded for each considered situation, compared to the sounds generated by the model, are shown in Fig. 4.

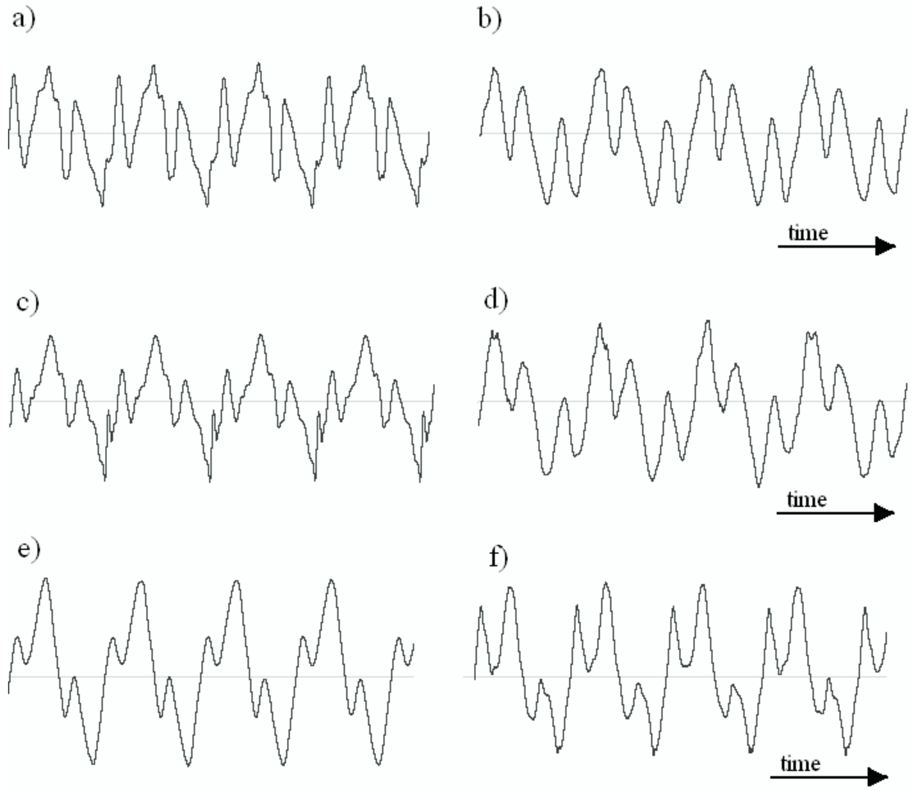

Fig. 4. Time waveforms of the original and simulated sounds: a) original and b) simulated sound for recording studio – played loudly, c) original and d) simulated sound for anechoic chamber – played loudly, e) original and f) simulated sound for recording studio – played softly.

6. Conclusions

Performed recordings show that role of the output bell is significant and the numerical clarinet model should contain output filter which represents propagation properties of the bell. Presented realization of the output filter by means of the adaptive filtering has given good results. For the purposes of the assessment of the modeling results comparison of waveforms of the clarinet sound and the signal generated by the model should be done. Good waveforms agreement takes effect in good listening assessment of modeled sounds. This fact confirms that the use of the adaptive filter (which performs adaptation of time waveforms of the input signal to the reference original signal) for the output filter implementation in the waveguide synthesis of a clarinet is the right decision.

A good result achieved in the realization of the output filter qualifies this part of the model for using it in further clarinet waveguide modeling.

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