SIGNAL MODEL BASED SYNTHESIS OF THE SOUND OF ORGAN PIPES

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ABSTRACT. The paper describes a special digital time-domain additive synthesis technique of the sound of organ pipes. The synthesis applies a periodic signal model that takes into account the basic psychoacoustical parameters of a musical sound: the stationary spectrum, the attack and decay transients, the quasi-steady properties and the external effects. The model is based on the Fourier-expansion of the periodic signals, which is completed with filters and noise-generator implementing the parameters above. The parameters of the model were derivated off-line from original pipe-records by means of signal processing tools. The result of the method is convincing both for laymen and for musicians. Theoretically the introduced signal-model structure and the developed analysis method are also able to model other instruments that has no strong non-linear properties.

1. INTRODUCTION

The high-fidelity synthesis of the characteristics of the musical instruments is a long-standing problem for both the musicians and the acoustical engineers. The pipe-organ – with its building and maintenance costs – exceeds from the musical instruments. Accordingly the organ is also in the centre of the synthesis research for a long while, so there are a lot of synthesis methods, which have been developed.

The well known – and also easy measureable – property of a musical instrument is the stationary spectrum [1]. The organ pipe-registers have different characters, and the spectrum is the function of the pipes' physical parameters (so the other sound-parameters). E. g. the pipes made of wood have more noise and less harmonics than the pipes made of organ-metal, or the odd overtones of a closed pipe have much smaller amplitude than that of the opened one.

However, only the synthesis of the main components of the spectrum is not enough for a good-quality reproduction. The attack and decay transients of an instrument and the modulations on the harmonics, or other quasi-steady processes are also important. Some examinations prove that without the attack and decay transients some instrument is no more identificable [2], and in some cases only the specific modulations of an instrument on a sine signal are enough to recognize the instrument itself [3]. Hence, a good synthesis has to implement both the transient and the quasi-steady processes.

In the related articles the last mentioned property of a musical instrument is the effect of the ambience of the sound-source. The organ normally sounds in a great church or in a hall, far away from the listeners. Closer to the pipes (or without this effect) the organ has unknown sound for laymen, it sounds like a flute or other wind-instrument. Another external effect is the sometimes observable coupling mechanism of two pipes [4]. The localization of the sound-sources and the 'feeling of the size of an instrument' fall also under this category . Both of these last parameters originate from the different position of the pipes. A 'high-fidelity' synthesis also has to take into account these effects.

None of the existing organ-sound synthesis techniques takes into consideration all of these parameters. The Hammond-organ and the analog techniques have problems with the frequency-dependent spectrum [5], [6]. The prevalent sampling method (PCM) cannot make random effects in the stationary state [7]. The drawback of the so-called Physical Modeling (PM) is that it is hard to find the transformation between the discussed psychoacoustical and the model parameters [8].

The signal model introduced in the paper tries to take into account most of these parameters. Section 2 recalls the signal model and introduces the necessary parameters used for synthesis of the organ sound. In section 3 the off-line derivation of the model's parameters from original pipe-records is discussed. Section 4 shows some simulation results, and the paper is closed with a short conclusion.

2. THE SIGNAL MODEL

The concept of the synthesis is the periodic signal model that has been already applied in other technical applications [9]. This model – a conceptual signal generator – is based on the Fourier-expansion of the periodic signals. According to the sampling-theorem, such a generator can generate a band-limited periodic signal, consisting of N complex components. In sound synthesis it realizes directly the discrete spectrum components of the instrument.

In this concept the attack and decay transients have effect only on the harmonics. The adjustable parameters of each harmonics are the magnitude, the frequency and the relative phase. At this moment only the magnitude-changing was examined.

The organ pipes – and the most of the other wind instruments – have a special characteristic, the so-called wind-noise. To make a more realistic model, this feature should be realized. It is a broadband component of the sound, with specific peaks in it (see in Fig. 2). To integrate it into the signal model, the periodic generator has to be completed with a special noise-generator. During the transients the discussed magnitude-changing has to be applied also in this generator.

The applied periodic signal model for sound synthesis can be seen in Fig. 1. The periodic signal generator has two main parameters – the fundamental frequency and the volume –, and each harmonic component has further parameters, these are the relative magnitude and the phase. The noise generator produces filtered white noise which is added to the magnitude-modified outputs of the periodic generator. At the end the summarized output is modified by the external effects.
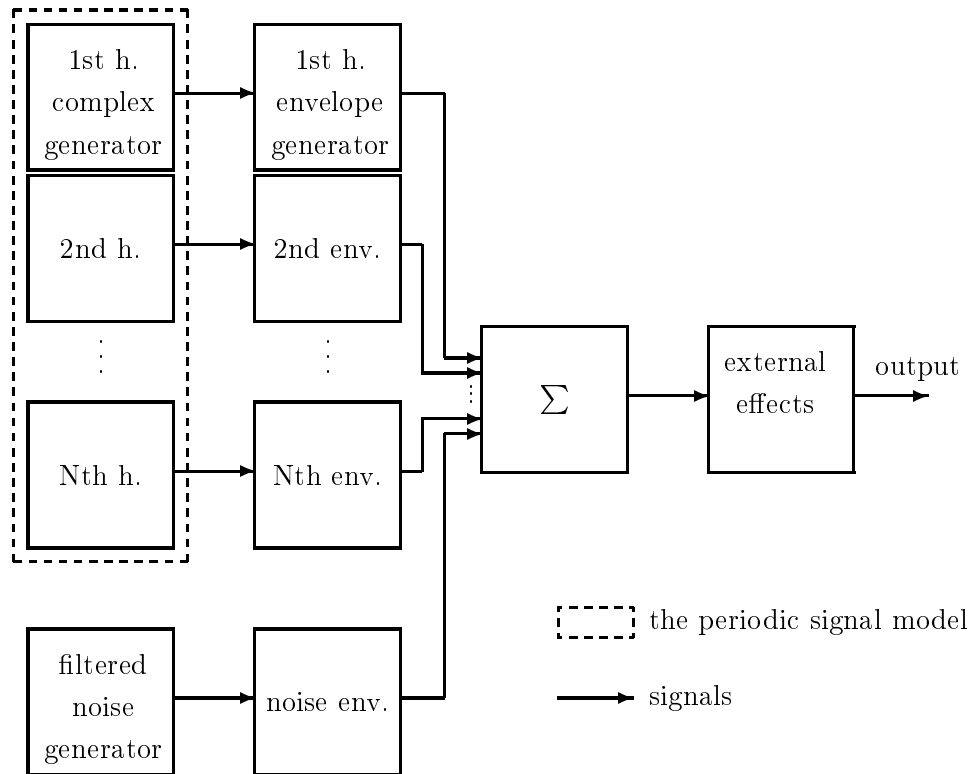

FIGURE 1. The integrated signal model

3. DERIVATION OF THE PARAMETERS

In order to determine the parameters of the signal model, original pipe-sounds were recorded by a measurement system (microphone, A/D converter and a computer) at a sampling rate of 44.100 Hz, with a resolution of 16 bit. The records were processed off-line with MatLab.

The fundamental frequency and the magnitudes of the harmonic components were calculated via the discrete Fourier transform (DFT). Because of the high resolution demand, the spectrum was calculated in 65536 (2^{16}) points, and the picket-fence effect was suppressed using a Blackman window.

For data and computation-time reduction the attack and decay envelopes of the harmonics are implemented as step-responses of IIR-filters. With this method the Nth harmonics at time step k can be computed as

$$x_{N,k} = h_k A_N \cos(2\pi N k f_0 / f_s + \varphi_N),$$

where $x_{N,k}$ is the Nth harmonic component at time step k, A_N and φ_N are the relative magnitude and phase of the component, f_0 and f_s are the fundamental and the sampling frequency, respectively. h_k contains the samples of the step-response of the designed envelope-filter.

The envelope-filters were determined in the time-domain. First, each harmonic component was filtered by a narrowband FIR filter designed by means of the windowing FIR filter design method [10]. Then the envelopes of the harmonics were determined as the absolute value of their Hilbert-transform. To get the best time-domain result, the obtained envelopes were averaged, and a step-response of a 3rd order IIR filter was fitted on each of them. The algorithm used the Prony's IIR-filter design method as initial step, then for better curve-fitting the Steiglitz-McBride iteration was used [11]. The described algorithm has found the required filters, but the real-time implementation of them (see in section 4) could be unstable, because of the finite wordlength effect. Therefore the algorithm was modified so that the filters are refreshed only in every Nth time-step. To provide smooth changing between two envelope-filter refreshing, linear interpolation is used.

As mentioned previously, the spectrum of the organ pipe has also noise component. The noise-filter was designed as follows: subtracting the discrete components from the spectrum, 2nd order resonant filters were fitted to the specific peaks in the averaged noise spectrum. They can be designed easily having the center frequency, the gain level and the damping factor of the separated peaks [12]. The resulted analog filter consists of 10-14 2nd order resonators, and it was converted to digital one by means of the bilinear transformation [11].

At this stage of the synthesis the examined external effects are only the reverberation of the hall and the location of the pipes. This latter one can be modeled by intensity and time-delay stereo soundfield, while the reverberation can be simulated by means of a digital effect processor.

4. RESULTS

In this section some synthesis results are described. To examine the efficiency of the introduced synthesis method, it had been implemented off-line with MatLab, and real-time on a 16 bit fixed-point digital signal processor (DSP). Both of them are controllable by MIDI codes. The MatLab program's output is a wave-file, the DSP's is the stereo sound. The DSP-program can compute three sound at the same time, with 8 harmonics for all sound, at a sample rate of $f_s = 22050$ Hz.

In Fig. 2-3 some examples of the implemented synthesis are shown. The spectrum of two organ pipes and their models can be seen in Fig. 2. The first one is a c_4-pipe of a Bourdon register (closed, wood pipe), the second is a Diapason e_4-pipe, which is an opened organ-metal pipe. It can be seen clearly that both the original and model Bourdon pipe have more noise, and
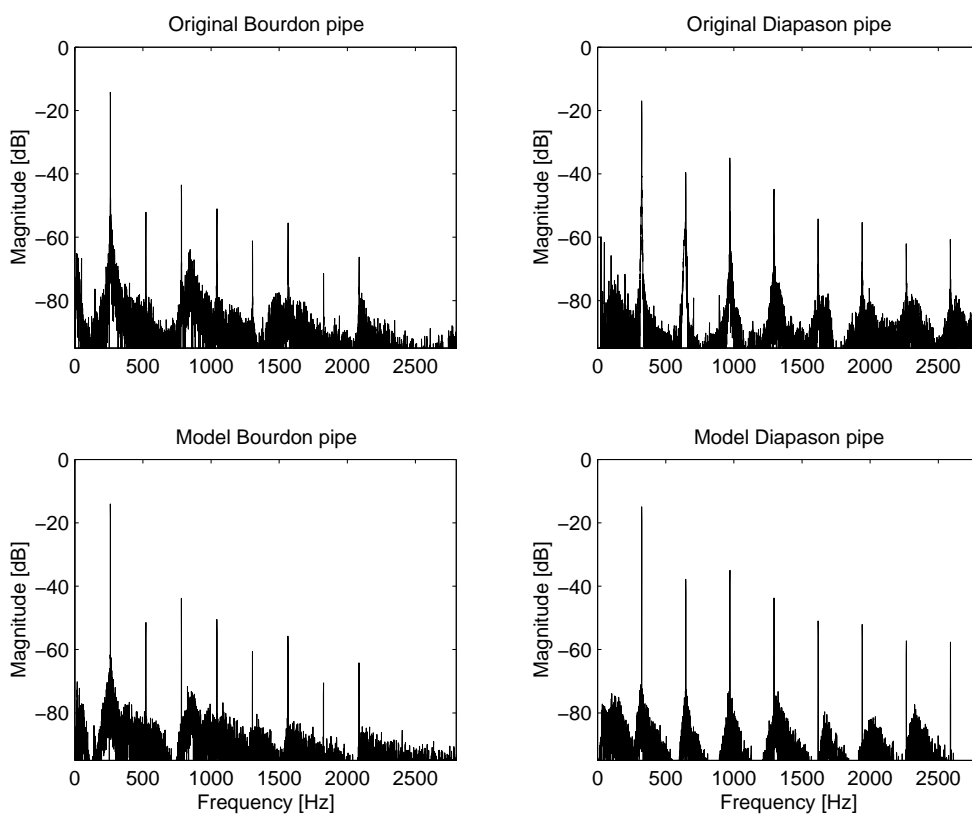
FIGURE 2. The stationary spectrum of two original pipes and their models

their odd harmonics have smaller magnitude, than those of the Diapason pipes. Furthermore, the metal pipe and its model have much more relevant components than the wood ones'. The modeling of the discrete spectrum is
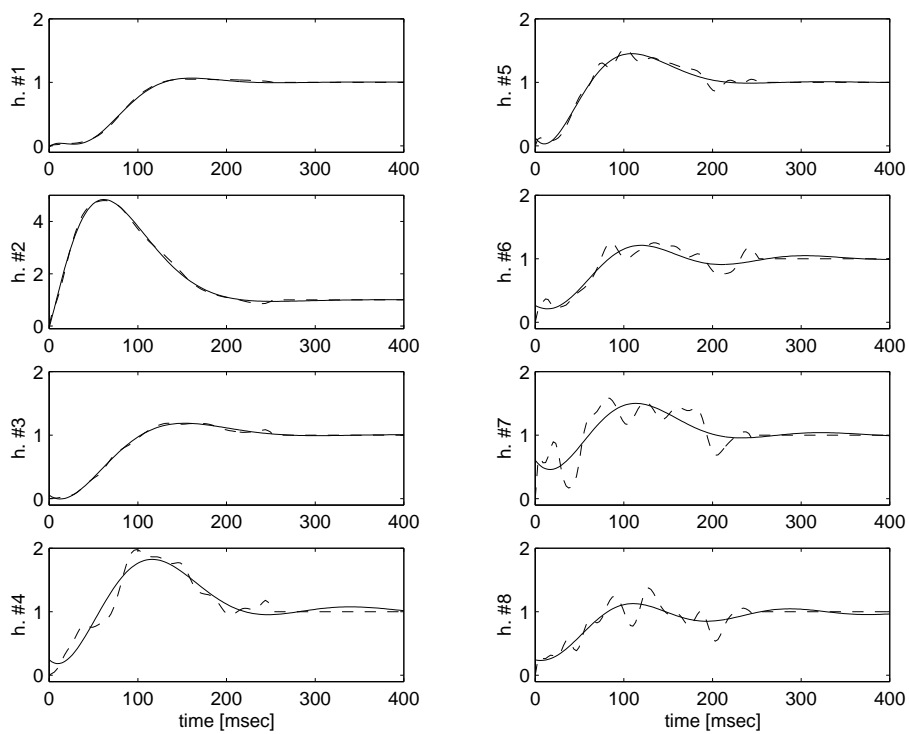

FIGURE 3. The envelopes of the first 8 harmonics of a Diapason pipe (dashed lines), and the fitted step-responses (solid lines)

very good, and the synthesis of the main characteristic of the noise spectrum is also acceptable.

An example of the averaged original attack transients and the estimated 3rd order IIR filter step-responses can be seen in Fig. 3. The estimation is good for the lower harmonics with good signal to noise ratio (SNR) (see Fig. 2, Diapason pipe). The higher the order of the component, the smaller its SNR, this is why the modeling worse for higher order components. Note that their precise synthesis is not required accordingly to their small magnitude.

5. Conclusion

This paper introduced a signal-model based sound synthesis, expecially for modeling organ pipe sounds. It is proved that an additive synthesis method can be efficient, if its parameters are based on good psychoacoustical properties. In addition, it was shown that the organ pipes can be modeled with linear synthesis method. Therefore it should be worth to examine the efficiency of the introduced technique on other (wind) instruments with no strong non-linear properties.

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