Signal Model Based Synthesis of the Sound of Organ Pipes

Orgonasípok jelmodell alapú szintézise

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Abstract

The paper describes a digital time-domain additive synthesis technique and its application to the synthesis of the sound of organ pipes. The model is based on the Fourier-expansion of the periodic signals. This periodic signal generator is completed with filters and noise-generator implementing the instrument-specific parameters. The novelty of the proposed method is that it utilizes independent IIR filters for each harmonics, which filters are responsible for the transient behavior of the sound. The result is convincing both for laymen and for musicians. For listening to it, there are some original and synthesized sample available at *http://www.mit.bme.hu/~markus/organ*. Additional advantage of the introduced signal-model structure is that it is able to model other instruments that has no strong non-linear properties.

Kivonat

A cikk egy – részleteiben új – digitális szintézistechnikát ismertet, valamint bemutatja az eljárás hatékonyságát orgonasípok hangjának reprodukálására. A felhasznált modell a periodikus jelek Fourier-sorfejtésén alapul, kiegészítve szurokkel és zajgenerátorral, amelyek a hangszer-specifikus jellemzok megvalósítását szolgálják. Az alkalmazott módszer újdonsága az, hogy a hangszer tranziens berezgését az egyes felharmonikusokra alkalmazott független IIR-szurokkel modellezi. Az eredmény mind laikusok, mind pedig hozzáértok számára meggyozo. Az eredmények jobb összehasonlíthatósága érdekében, néhány eredeti és szintetizált hangminta letöltheto a *http://www.mit.bme.hu/~markus/organ* címrol. A bemutatott jelmodell-struktúra további elonye pedig az, hogy egyéb – közel lineáris viselkedésu – hangszerek modellezésére is használható.

1. Introduction

The high-fidelity synthesis of the sound 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 center of the synthesis research for a long while, so there are a lot of synthesis methods, which have been developed. In this article a signal-model based synthesis will be introduced.

In the following the most important properties of the organ sound are reviewed.

It is well known that a significant – and also easy measurable – part of a musical instrument's sound is the stationary spectrum. Accordingly, the organ pipe-registers have also 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 [1].

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 important part of a musical sound, too. Some examinations prove that without the attack and decay transients some instrument is no more identifiable [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 ambiance 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 organpipe-sound is unidentifiable without this reverberation [4]. Another external effect is the sometimes observable coupling mechanism of two pipes [5]. The localization of the sound-sources (originate from the different positions of the pipes) falls also under this category [6].

None of the existing organ-sound synthesis techniques takes into consideration all of these parameters. The prevalent sampling method (PCM) – which is used in the digital organs today – cannot make random effects in the stationary state, because of the special storing mode of the samples, and it does not support also the varying transients. The so-called Physical Modeling (PM) has some drawback, too. Using physical models based on the differential equation of the musical instrument, we have to calculate a lot of redundant parameters, and it needs fast hardware for real-time synthesis [7]. In the case of using semi-physical models, sometimes it is hard to find the transform between the sound- and the model parameters [8]. There is a third method, which is very close to the synthesis to be introduced, the Spectral Modeling Synthesis (SMS). However, the main difference is that the SMS is a frame-oriented method, and it needs in every frame newer parameters [9].

The signal model introduced in the paper tries to take into account most of the sound parameters mentioned above, keep the hardware-demand low. 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 main concept of the synthesis is the periodic signal model that has been already applied in several other technical applications [10]. 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. 3). 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, 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.

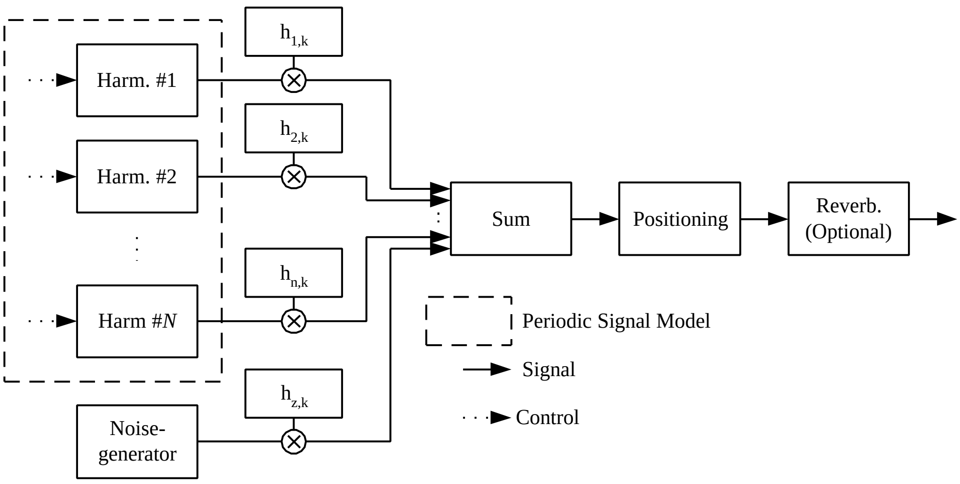

Figure 1: The integrated signal model.

3. Derivation of the Parameters

In order to determine the correct parameters of the introduced 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, using a developed analysis process that can be seen in Fig. 2.

First – using magnitude-limits – the stationary and the transient parts (the attack and the decay) were separated in the time-domain. From the stationary part 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 for suppressing the picket-fence effect Blackman window was used.

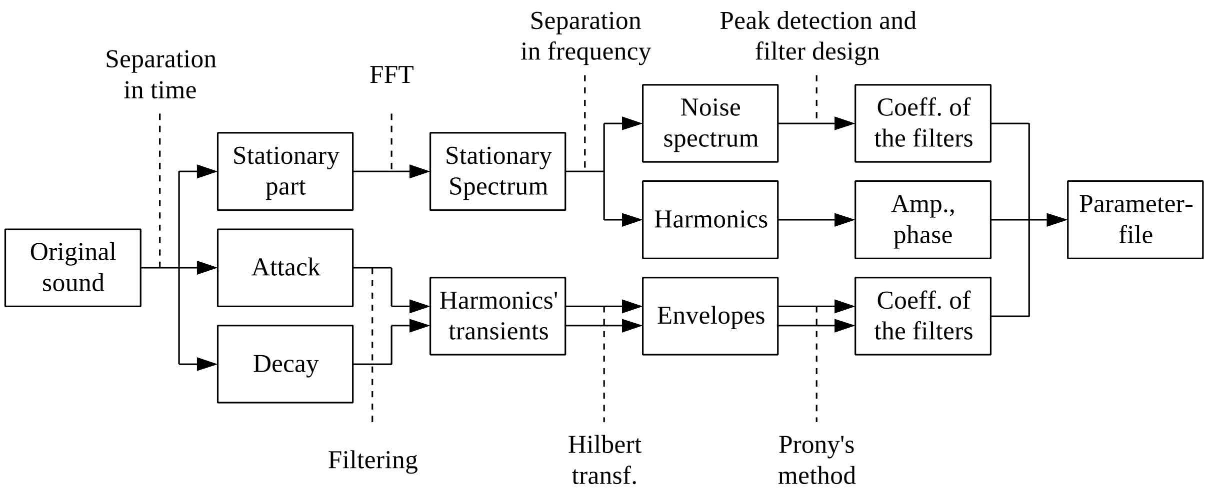

Figure 2: The analysis process.

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

$$x_{i,k} = h_{i,k} A_i \cos(2p\,(i\, {}^{f_0}\!/_{f_s})k + j\,) \qquad i=1..N,$$

where $x_{i,k}$ is the *i*th harmonic component at time step *k*, A_i and j_i are the relative magnitude and phase of the component, f_0 and f_s are the fundamental and the sampling frequency, respectively. Finally, $h_{i,k}$ represents the samples of the step-response of the designed envelope-filter.

These 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 design method [11]. 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 [12]. 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 N^{th} 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. The resulted analog filter consists of 10-14 2nd order resonators, and the filter was converted to digital one by means of the bilinear transform [12].

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 using hall-simulators.

4. Results

To examine the efficiency of the introduced synthesis method, it had been implemented off-line using MatLab, and real-time on a 16 bit fixed-point digital signal processor (DSP). Both of them are controllable by MIDI codes. 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.

The spectrum of two organ pipes and their models can be seen in Fig. 3. 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 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 very good, and the synthesis of the main characteristic of the noise spectrum is also acceptable.

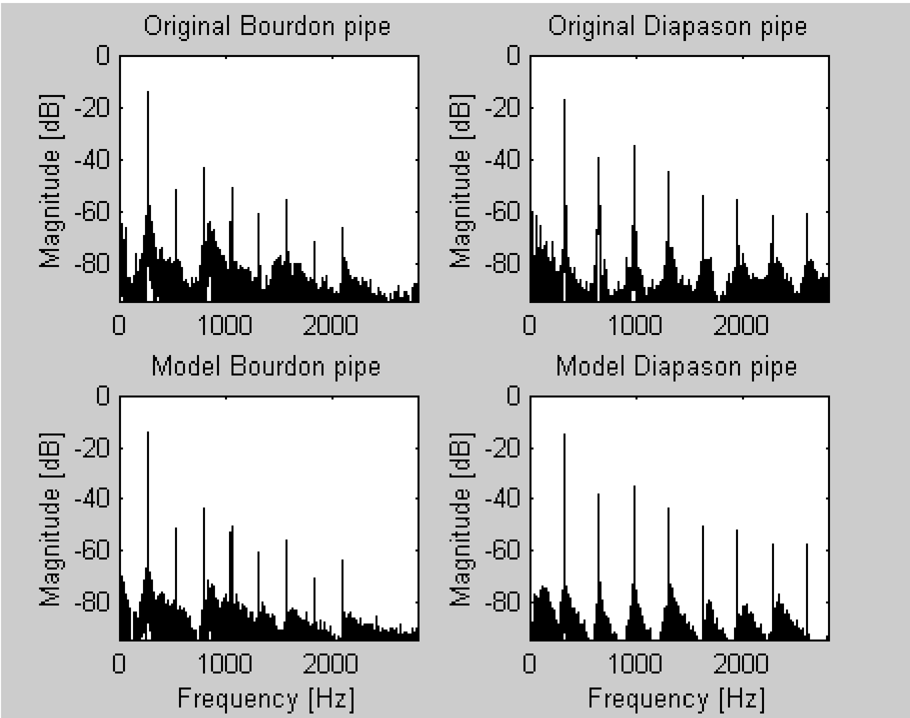

Figure 3: The stationary spectrum of two original pipes and their models.

An example of the averaged original attack transients and the estimated 3rd order IIR filter step-responses can be seen in Fig. 4. The estimation is good for the lower harmonics with good signal to noise ratio (SNR) (see Fig. 3, 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.

Using the MatLab and the DSP-program, we have made some demonstrations. For comparison, we have made original records using the organs we have measured. These original and synthesized samples are available on the Internet, at [13].

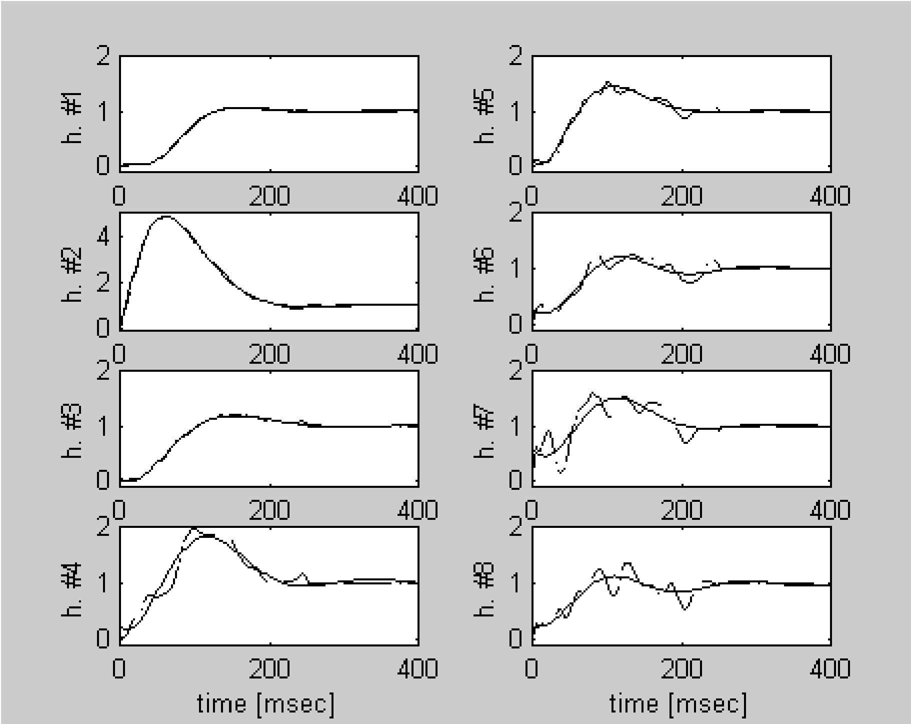

Figure 4: The envelopes of the first 8 harmonics of a Diapason pipe and the fitted step responses.

5. Conclusion

The paper introduced a signal-model based sound synthesis, especially for modeling organ pipe sounds. It is proved that this additive synthesis method is efficient, because its parameters are based on measured sound-parameters. In addition, it was shown that the organ pipes can be modeled using 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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