ARCHIVES OF ACOUSTICS
21, 2, 131 – 147 (1996)

SYNTHESIS OF ORGAN PIPE SOUND BASED ON SIMPLIFIED PHYSICAL MODELS

A. CZYŻEWSKI, B. KOSTEK and S. ZIELIŃSKI

Sound Engineering Department
Faculty of Electronics, Telecommunications and Informatics
Technical University of Gdańsk
890-952 Gdańsk, ul. Narutowicza 11/12

Problems related to the implementation of a physical model based synthesis of an organ pipe sound are discussed. A new approach to the physial modelling of an organ pipe sound, namely the waveguide synthesis is introduced. The results of some experiments with this kind of synthesis are presented. Specific features of presented methods and corresponding applications are quoted. Examples of a computer analysis of both synthesized and real musical sounds are presented and compared.

1. Introduction

The new generation of digital signal processors offers many possibilities to create synthesized sounds of classic-like instruments. On the basis of the new technology new synthesis algorithms created on the basis of the physics of instruments were proposed. One of the most modern methods that belongs to the category of physical modelling is the "digital waveguide" synthesis [23, 25, 26]. In this method the wave equation is first solved in a general way to obtain the travelling waves in the instrument body that are then simulated in the waveguide model. In the lossless case, a travelling wave between two points in the medium can be simulated using a digital delay line and frequency-dependent losses can be simulated by a FIR digital filter [8, 9, 23]. In the paper also a short review of widely-known traditional synthesis methods is included and their limitations, when applied to the organ pipe sound synthesis, are pointed out.

The tool used for the experiments was the NexTStep Operational System based workstation. This operational system available today with Sun, Hewlett-Packard and IBM Pentium computers is particularly suitable for the development of sound synthesis algorithms, especially because of the advanced software supporting this task.

The SynthBuilder system may serve as an example of such a software application. Within it, it is possible to create and test a new instrument. The unit generator panel contains: an oscillator, buzz, random number generator, simple envelope with attack and decay, filter and mathematical operations. Since inputs and outputs are wired, it is possible to designate parameter fields. Moreover, some available analyzing software packages were designed for musical sounds [2]. The software enables various analyses, especially those more sophisticated, being very useful for the study of musical timbre through the examination of various graphic representations of the data.

Two methods of physical modelling of natural instruments were used in the study. The first one, proposed by Fletcher [11], is based on the numerical solution of the wave equation. The second one, known as the waveguide synthesis, is related to the simulation of travelling waves in the waveguide model of a natural instrument. A couple of experiments with these two methods were made in order to compare their output to results the obtained with some classical methods for digital synthesis and to the sounds of real organ pipes.

2. Traditional sound synthesis methods

Traditional synthesis methods fall into one of the four categories quoted below: additive synthesis, subractive synthesis, nonlinear distortion synthesis and sampling.

Generally, digital synthesis of sound is a function F described by the following equation:

$$s(n)=F(n,p_1,...,p_m), \tag{2.1}$$

where: $s(n)$ — consecutive samples of synthesized sound,

$p_1,...,p_m$ — set of synthesis control functions.

Depending on the selection of the synthesis control functions, one can discern many methods, such as additive synthesis, subtractive synthesis, frequency modulation (FM), phase distortion.

The additive synthesis belongs to the "classical" methods of digital synthesis of sound. It results from the reverse Discrete Fourier Transform of spectral components having variable amplitudes [13]. The additive synthesis demands a big number of stored data. It would be very difficult to relate these data providing input of this algorithm to the way of playing on a pipe organ instrument with regard to musical articulation.

The subtractive synthesis is based on filtration of a wide-band signal in the non-stationary linear circuit. Usually, the subtractive synthesis is used in digital technology as a tool for timbre formation of signals generated digitally in samplers, wavetable synthesizers and other ones. This method provides the most wide-spread technique for the synthesis of pipe organ sounds. However, it does not allow for musical articulation [14, 15] and is not suitable for modelling such phenomena as overtone generation, overblowing and other features accompanying the sound rise in a real pipe because it is technically difficult to obtain sufficiently fast and deep changes of filter parameters.

The frequency modulation (FM), waveshaping and phase distortion fall into the category of nonlinear distortion synthesis. Frequency modulation is well known in radio engineering. J. CHOWNING discovered [5] that by bringing the modulation rate down into the human hearing frequency range (20 Hz – 20 kHz) and controlling the relationship between the carrier frequency, the modulating frequency and the modulation index ratio, one can vary the timbre of the synthesized sound.

In practice, the spectrum shape of the synthesized sound is very difficult to predict and control (the equation may be solved using Bessel functions). Nevertheless, some modifications of this method, which solve the problem partially, were proposed lately [24], especially for the steady-state part of the signal. Independently of the method, the frequency modulation synthesis does not allow to control the spectrum behaviour during the transient states with sufficient, from the technological point of view, effectiveness. This drawback eliminates this method as a practical tool for an efficient synthesis of the pipe organ sound.

The phase distortion synthesis method is another kind of methods belonging to the category of nonlinear distortion synthesis. In this case, samples corresponding to a single period of the cosine wave are stored in the table of the phase distortion synthesizer. In the process of synthesis some samples are read out of the table at a rate different from those of others, which causes phase distortion. As a result various sound waveforms can be obtained.

The waveshaping synthesis is similar to the phase distortion method described above. The difference consists in the fact, that the sampled-data of one period of the cosine wave are read out at a constant rate and then processed according to the shaping functions [3]. As the last two methods are based on nonlinear equations, it is difficult to manage all parameters of such synthesis and to control all phases of the created sound.

Many electronic instruments, that allow for a plausible synthesis of natural sounds, utilize the sampling technique. In this method the musical sound is used as a sound source. Subsequently, it is converted into the digital domain, stored in the computer memory, processed according to the chosen algorithm and then converted back to the analogue domain. Although sampling allows for efficient synthesis of natural sounds, a large amount of data must be stored in the computer memory, especially when considering such a big instrument as a pipe organ. The last one of the methods described above is not a system of synthesis in the strict sense.

3. Sound production in an organ flue pipe

An open diapason and a flute pipe are typical organ flue pipes. They are shown schematically in Fig. 1. The main three parts of a flue pipe are: foot, languid, and body. The term "mouth" denotes respectively upper and lower lips, the languid and the ears in the case of a diapason pipe (ears are not shown in Fig. 1). The slit between the languid and the upper lip is called a "flue" or a "windway", therefore organ pipes of this kind of construction are often named "flue pipes".

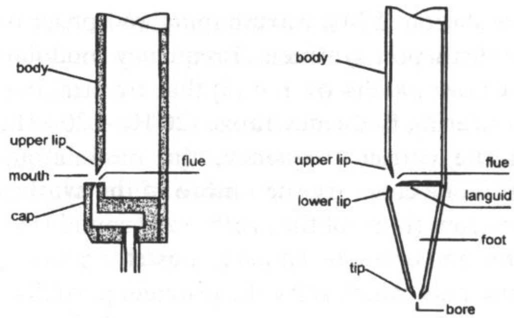


Fig. 1. a. Cross-section of the wooden organ pipe of a "flute" type, b. Cross-section of the open diapason pipe [19].

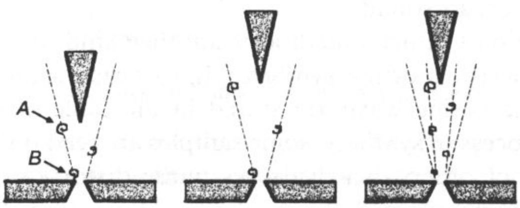


Fig. 2. Mechanism of "edge tone" production [21].

One of the first qualitative explanations of sound production in organ flue pipe was given by RICHARDSON [21, 22]. In his studies a flue pipe was treated as two compound systems in which an "edge-tone" is directly coupled to the column of air of fixed length. The mechanisms of "edge-tone" formation are illustrated in Fig. 2. In the simplest case, a vortex A leaves the outer wall of the orifice as the preceding one on the same side B strikes the edge. The frequency of the "edge-tone" is related to the frequency with which the vortices strike the edge. The wavelength of the "edge tone" equals, or is a small submultiple, of the distance between the vortices in the same row. There is a minimum distance l_0 of the distance l between the slit and the edge for any given velocity of efflux v at which a tone can be produced. In an organ pipe the distance between the slit and the edge of the upper lip is constant, but the velocity of the air-jet can be varied by changing the pressure in the pipe foot. When v is kept constant as l is increased beyond the minimum l_0, the pitch of this "edge-tone" is in accordance with the relation:

$$v/fl_0=\text{constant} \tag{3.1}$$

until l reaches a value equal approximately to the double value of l_0; then the tone, which is now the sub-octave of the original, may suddenly rise an octave.

Richardson concluded that the "jump" of the frequency of the "edge tone" up to the higher frequency occurs when the pipe is supplied with relatively low pressure and, as a consequence of this, the pipe can respond with some kind of overblowing [21]. The air, issuing from a slit in the mouthpiece and striking the upper lip, produces the "edge-tone", the pitch of which would rise continually in proportion to the wind velocity, i.e. to the square root of the pressure in the wind chest. Thus, the organ pipe can be considered as a resonator coupled to an "edge-tone" generator that amplifies its response. These possible modes of vibration in relation to the "edge-tone" are shown in Fig. 3 [20].

As already mentioned, the relation shown in Fig. 3 refers to the case when small power, at low pressure, is turned onto the pipe, before the normal pressure has been attained.

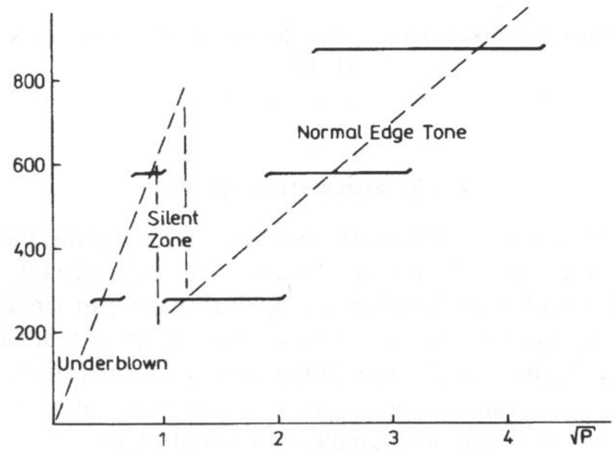


Fig. 3. Organ-pipe tones in relation to edge tones [19].

A systematic study of the sounding mechanism of the flute and organ pipe was undertaken by Coltman [7]. He pointed out that the picture of an organ pipe as a resonator that is coupled to the "edge-tone" generator and amplifies its response was oversimplified. He carried out detailed measurements of the acoustic impedance of the air jet. In this study, he measured the acoustic impedance at a given frequency as the function of a jet velocity [6]. A very important observation was made, namely that the interaction between normal resonance modes of the air column within the pipe and the air jet should be treated in terms of a nonlinear interaction [10]. The jet emerging from the narrow slit defined by the edge of the languid and the lower lip of the pipe mouth, is sensitive to the displacement by the fluid motion v through the mouth of the pipe. If the acoustic flow is out of the pipe mouth, the jet is deflected outside the upper lip and the pressure falls, while if the acoustic flow is inwards, the jet is deflected inside the upper lip and raises the pressure in the pipe near the mouth. For a steady velocity, v, in the pipe mouth, the driving force F produced by the jet had a form like that one shown in Fig. 4 [10, 11, 18].

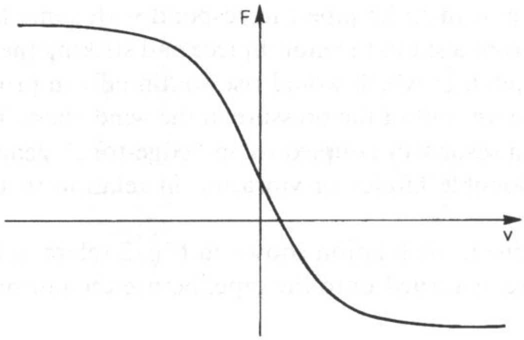


Fig. 4. An exemplary shape of nonlinearities caused by the interaction between the air jet and the pipe [10, 11, 18].

4. The articulation effects

Lottermoser [17] reports that in most cases of baroque organs, the organist is able to influence the transients. When he touches the key quickly (staccato), the higher-frequency tones of short duration arising during the transients appear with full amplitude. When he touches the key slowly (legato), these tones disappear. This phenomenon is very important for presenting well-articulated music.

Significant qualitative differences between organ transient sounds obtained for fast and slow rates of opening of the pallet were found by Caddy and Pollard [4]. Nolle and Finch [19] measured transients of flue organ pipes in relation to the pressure rise time. They found that in the case of short-time transients, an overshoot of the second mode harmonic occurred. In the "slow" regime, not only is there no large overshoot, but modes 2 and 1 can become phase locked well before the final pressure is reached. These effects were also observed by us in our previous studies [14, 15, 16].

5. Physical model based on the wave equation

The organ pipe can be considered as two subsystems, for example, the air column — a very, nearly linear, resonant system — and the jet system including a highly non-linear interaction with the pipe lip. The air column system of the pipe has a series of normal modes with (angular) frequencies n_i. The displacement x_i of the ith mode obeys the equation of the form [11]:

$$\ddot{x}_i + k_i \dot{x}_i + n_i^2 x_i = \lambda_i F(t), \tag{5.1}$$

where: $F(t)$ — external force, k_i — damping coefficients, λ_i — coupling coefficients between the pipe and the air stream.

As far as the air jet is concerned, the driving force F for the static case can be written as a power series expansion:

$$F=c_0+c_1v+c_2v^2+c_3v^3+\dots \tag{5.2}$$

where v is the acoustic velocity of the air stream and c_n are coefficients of the blowing pressure function. Considering the time delay, δ_i, related to the jet travelling across the mouth of the pipe and the dispersive character of the jet, Δ_i/ω_i, equation (5.2) can be written:

$$F(t)=\sum_{m=0}^{\infty} c_m\left[\sum_{i=1}^{\infty} \dot{x}_i\left(t-\delta_i-\frac{\Delta_i}{\omega_i}\right)\right]^m, \tag{5.3}$$

where: δ_i — time necessary for a single vortex to be displaced along the air stream leaving the organ pipe mouth, Δ_i — phase shift.
The set of equations (5.1) and (5.3) can be written in the following form:

$$\ddot{x}+n_i^2x_i=f_i(\dot{x}_i), \tag{5.4}$$

where:

$$f_i(\dot{x}_j)=-k_i\dot{x}_i+\lambda_i\mathrm{F}(\dot{x}_1,\dot{x}_2,\dots). \tag{5.5}$$

This equation can be solved using the method of slowly varying parameters. The delay δ for a jet of length l is:

$$\delta=8\times10^{-4}lp^{-1/2}\alpha^{-1} \tag{5.6}$$

where α — coefficient of air flow velocity.

As the attack transient is strictly related to the pressure rise time and has a crucial influence on the timbre of the resulting sound, the main attention has been paid to modelling the pressure changes in the pipe foot described by the following equation:

$$p(t)=p_0+(p_1-p_0)\exp(^-t/\tau) \tag{5.7}$$

where: p_1 specifies the pressure peak, p_0 is the steady pressure and τ is the decay time from the peak level. It is possible to discern between at least two possibilities. When $p_1 \gg p_0$, the pressure peak is occurring and it may be referred to as the plosive transient attack. When $p_1 \ll p_0$, the transient is slow. A more detailed description of the phenomena related to the sound rise in flue pipes can be found in the literature [1, 11, 15].

In the experiment studies carried out in the Sound Engineering Department of the Technical University of Gdańsk, the method proposed by Fletcher has been implemented in a personal computer [14]. In these studies the frequency dispersion was neglected.

6. Waveguide synthesis

The digital waveguide synthesis is based on a modified approach to the physical model: the wave equation is first solved in a general way to obtain travelling waves in the medium interior. The travelling waves are explicitly simulated in the waveguide model, in contrast to computing a physical variable [12, 23, 26]. In the lossless case, a travelling wave between two points in the medium is simulated using a digital delay line. Frequency-dependent losses and dispersion can be modeled using digital filters.

The general class of solutions to the lossless, one-dimensional, second-order wave equation describing the air column system of the pipe can be expressed as:

$$x(l,t)=x_r(l-ct)+x_l(l+ct), \tag{6.1}$$

where: $x_r(l-ct)$ — right-going travelling waves, $x_l(l-ct)$ — left-going travelling waves, c — speed of propagation, l — position.

Fig. 5 shows the way of modelling such travelling waves using a pair of digital delay lines. The upper delay line simulates the right-going travelling waves, whereas the lower one is responsible for modelling the left-going waves. It is possible to compute the physical air column displacement in any sampling point by simply adding corresponding samples derived from delay lines, as illustrated in Fig. 5.

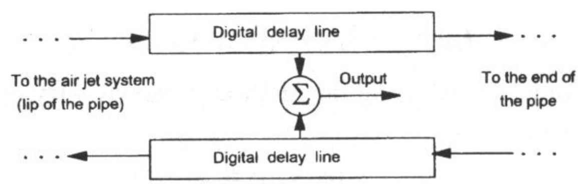


Fig. 5. Scheme of an elementary waveguide.

A more detailed schematic diagram of the semi-physical model of the flue organ pipe is illustrated in Fig. 6a. This model is based on a flute model developed in 1992 by VÄLIMÄKI *et al.* [26]. Experiments with this model were also performed by P.R. COOK [9] who proposed inaudible simplifications. A pair of digital delay lines (each of N/2 sample length) is used for modelling the vibrating air column inside the pipe. Linear filters at the ends of the pipe in this waveguide model simulate losses and the dispersive characteristics of the air column summed up together with the reflections. The lengths of the delay lines correspond to the effective length of the pipe. Since delay lines consist of an integer number of unit delays, the problem related to the proper tuning of the pipe, according to the musical scale, could appear. This problem can be solved using a fractional delay filter based on Lagrange interpolation [27, 28]. Filters F_A and F_B (see Fig. 6a) are responsible for modelling frequency-dependent damping and dispersion in the pipe and reflection properties of the corresponding ends. The oscillation in the pipe is produced by a nonlinear interaction of the air jet and the air column. This interaction can be described in terms of a nonlinear function and implemented by means of a look-up table [10, 11, 19]. Since the air jet has to travel a certain distance from the pipe foot to the lip, it is necessary to model it using an additional delay line (*dl1*). The whole system is excited with the noise shaped by the envelope generator that models changes of pressure in the pipe foot. It is possible to simplify this model by lumping delay lines and the corresponding filters as illustrated in Fig. 6b. This simplification does not cause any audible effect.

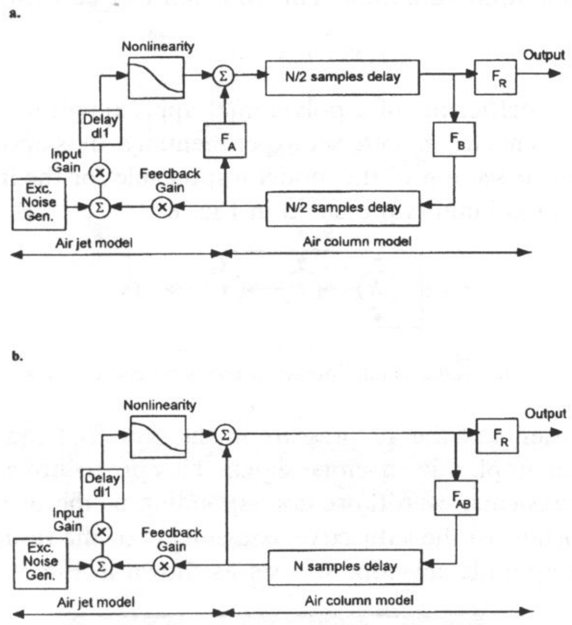


Fig. 6. Block diagram of a flue pipe model:
a. full lay-out,
b. simplified lay-out with elements lumped together.

In the experiments we have carried out, the waveguide model of a flute proposed by VÄLIMÄKI [26] was adopted. In order to simplify the model, the filter F_R was removed (see Fig. 7). This model consists of typical blocks such as delay lines, multipliers, one-pole filter and generators. The transfer function of the one-pole filter is as follows:

$$H(z)=\frac{b_0}{1-a_1z^{-1}}, \tag{6.2}$$

where: b_0 — amplification and a_1 — pole coefficient. The one-pole filter and a delay line (*dl1*) are used to simulate the travelling wave in the air column of the pipe. The jet

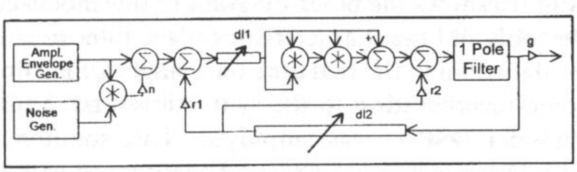


Fig. 7. Waveguide model of a flute pipe used for experiments.

delay is modeled using the second delay line (*dl2*). The amplitude envelope generator and the noise generator are used to model the changes of the air pressure in the pipe foot. In order to ensure efficient DSP calculations, non-linearities are modelled on the basis of a polynomial approximation. This function can be written in the form:

$$y(x) = a_0 x - a_3 x^3, \tag{6.3}$$

where: a_0 and a_3 — coefficients of a polynomial approximation of the 3-rd order. The values of coefficients a_0, a_3 were set experimentally, thus adopted as: $a_0 = 1$ and $a_3 = -1$. The nonlinear section of the model responsible for the interaction between the air jet and the air column is presented in Fig. 8.

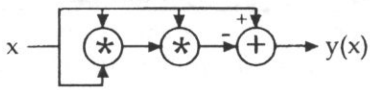


Fig. 8. Nonlinear section of the model simulating the interaction between the air jet and the air column.

The attack transients of the air pressure in the pipe foot can be modeled using various shapes of an amplitude envelope signal. This procedure allows to synthesize various types of transients, even those corresponding to the overblow of the pipe, which is very important to the subjective assessments of the resulting organ sound. The shape of the amplitude envelope was set as shown in Fig. 9.

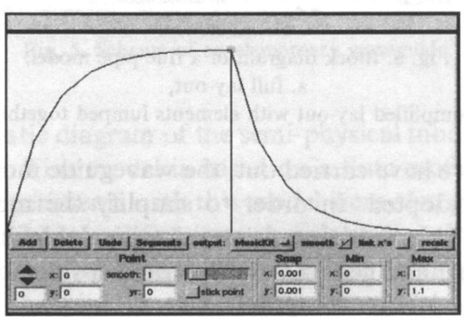


Fig. 9. Shape of an attack transient generated in the amplitude envelope generator.

In the experiments with waveguide synthesis, the SynthBuilder system written for the NeXT computer was used (Software elaborated by Nick Porcaro et al., Stanford University). This application allowed to model the pipe according to the algorithm described before. Fig. 10 shows the block diagram of this model constructed with the use of SynthBuilder tools and the characteristics of the filter used in the model. It was possible to run this algorithm in the real time on a single DSP chip (Motorola 56001). In order to add some reverberation to the synthetic sound, an external sound field processor unit (Yamaha DSP-1) was employed. This solution enables simulating conditions of a big interior, which is in a good accordance with the listeners' habits accustomed to perceiving organ sounds in the acoustical conditions of a church.

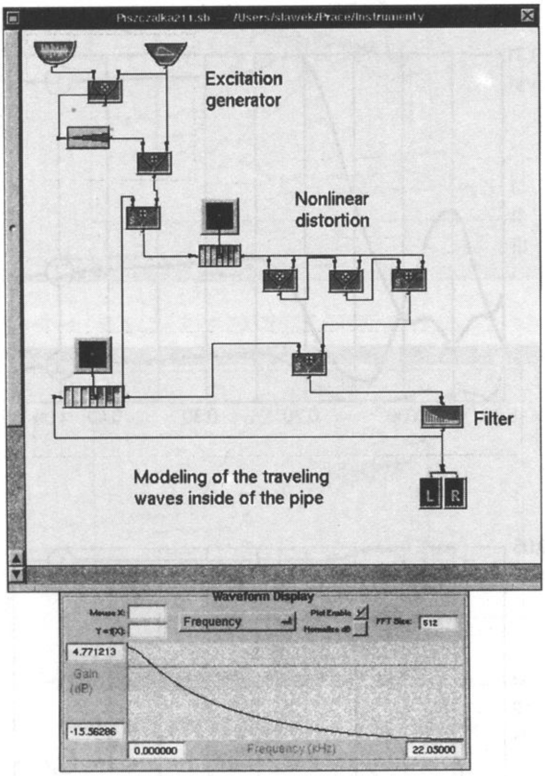


Fig. 10. Waveguide model of the organ pipe implemented using the SynthBuilder software tool and response of the applied low pass filter.

7. Experiments and results

As a result of the carried out experiments, sound of an organ flue pipe were synthesized using the two methods described above, namely the method based on solving differential equations and the waveguide modeling, both describing physical phenomena related to sound bore in organ pipes. Synthetic sounds were then analysed in time- and frequency-domains and compared to the corresponding analyses of natural sounds. Preliminary listening tests were also made in order to compare synthetic and natural sound patterns.

Calculations based on the method proposed by Fletcher have been performed for a wide variety of cases from which only two examples, shown in Fig. 11, have been selected. The plosive attack produces a speech transient in which the second pipe mode is dominant within the first 0.1 s (Fig. 11.a). For the slow attack the speech is delayed, with the fundamental being always predominant (Fig. 11.b). These effects are closely related to musical articulation distinctive features that appear in the sound of a real organ pipe, namely a flue pipe [15]. The harmonic analyses of the corresponding natural sounds are presented in Fig. 12.

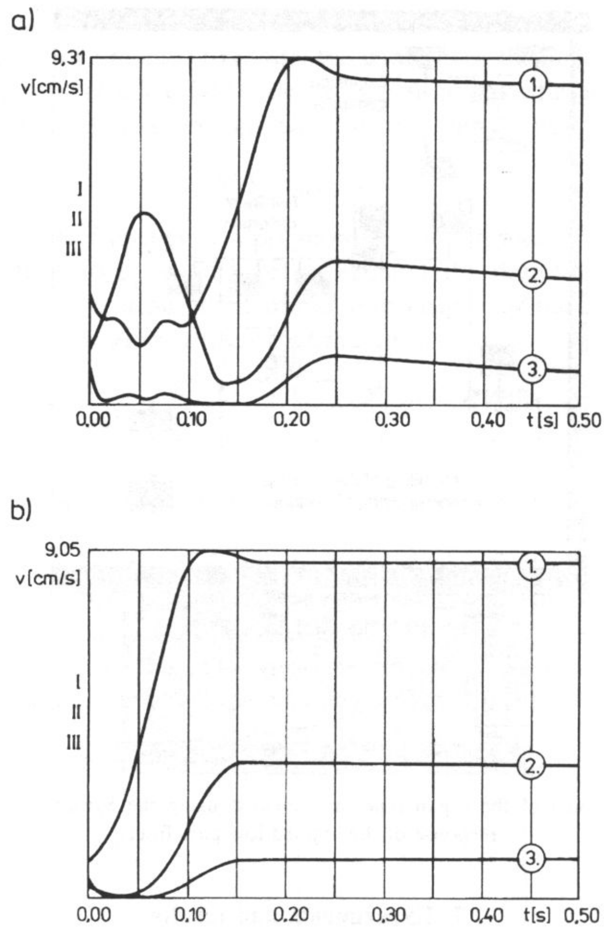
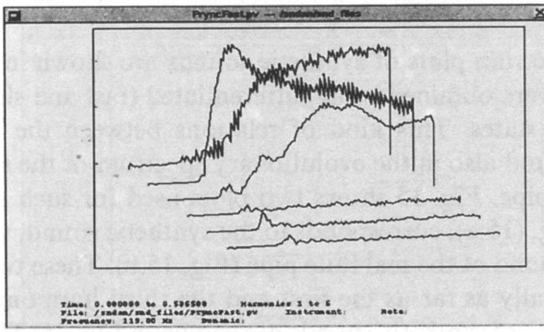


Fig. 11. Calculated pipe transients. In both figures, curves no. 1, 2, 3 refer to the first, second and third pipe modes, respectively:
a. fast attack ($p_0=2$ mbar, $p_1=6$ mbar),
b. slow attack ($p_0=2$ mbar, $p_1=0.5$ mbar).

Similar frequency analyses were made with sounds obtained using the waveguide synthesis [16]. Spectrum plots of the synthetic and natural (steady state) sounds of an organ flue pipe are presented in Fig. 13. There is a good qualitative accordance between them. The content of harmonic components is relatively small. There are only 6 harmonics of significant importance, whereas the higher ones are hidden under the noise components. It is important to note, that the amplitude of the second harmonic is in both cases less than the amplitude of the first and the third ones. Apart from this, the amplitude of the sixth harmonic in the natural sound is greater than the amplitude of the same harmonic in the synthetic one (Fig. 13).

a.



b.

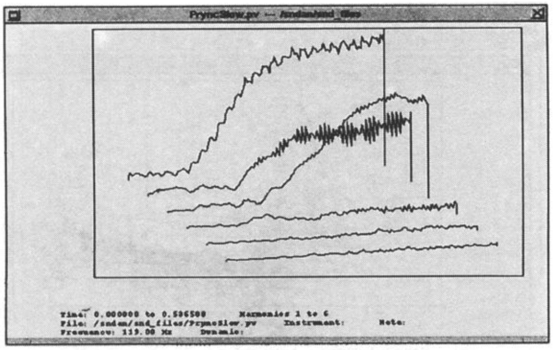


Fig. 12. Evolutionary plot of a spectrum of a Principal organ pipe:
a. fast attack,
b. slow attack
(Analyzing tool: *AnView* — program elaborated by C. Gennaula, C. Goudeseune, J. Beauchamp).

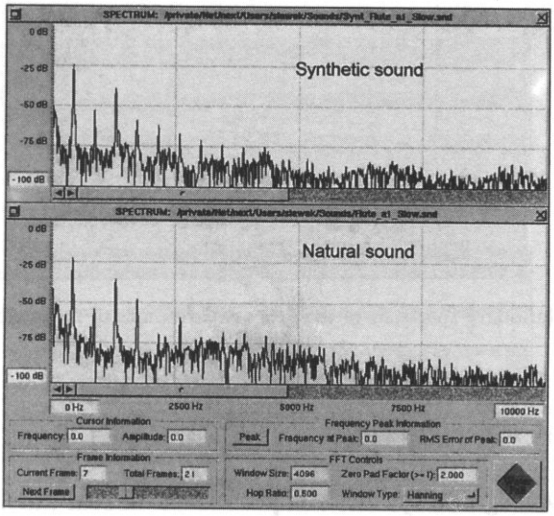
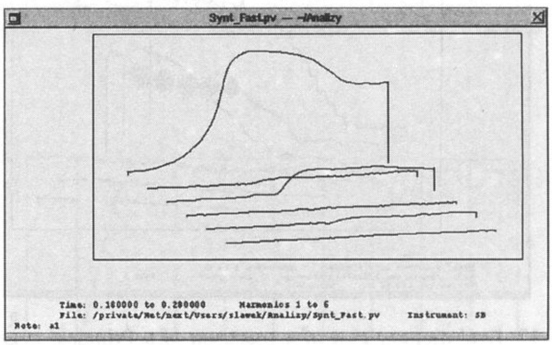


Fig. 13. Spectrum plots of the synthetic and natural sounds (of the flute type).

Evolutionary spectrum plots of synthetic sounds are shown in Fig. 14.a and Fig. 14.b. These sounds were obtained using differentiated (fast and slow) articulation of the attack transient states. This kind of relations between the fundamentals and harmonics can be found also in the evolutionary spectrum of the sound produced by the real organ flute pipe. Fig. 15 shows two plots used for such a comparison. The upper part of this Fig. (15.a), corresponds to the synthetic sound, while the lower one corresponds to the sound of the real flute pipe (Fig. 15.b). These two plots are similar to each other, especially as far as the first and the third harmonics are considered. Time-domain analyses of synthetic sounds are shown in Fig. 16. The fast attack gives a shorter amplitude envelope rise time than in the case of a slow attack.

a.



b.

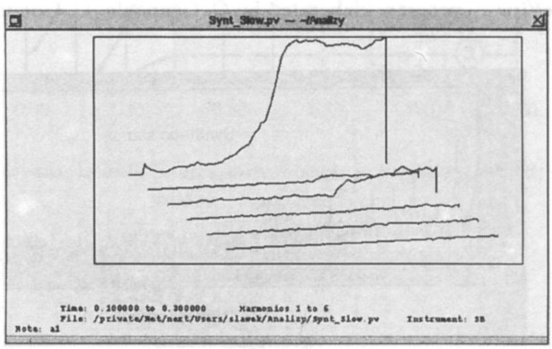


Fig. 14. Evolutionary spectrum of the first six harmonics of the synthetic sound:
a. fast attack,
b. slow attack.

The process of waveguide synthesis is performed in real time. Parameters of the synthesis have been adjusted interactively while listening to the synthetic sound. The parameters specified in Table 1 were found to be optimal for the synthesis of an organ a.

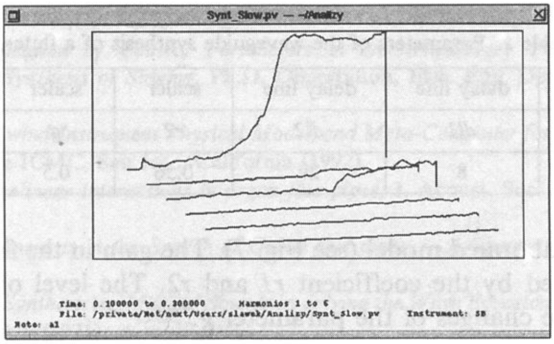


Fig. 15. Evolutionary spectrum of the first six harmonics of the synthetic and natural sound:
(a) synthetic sound,
(b) natural sound.

a.

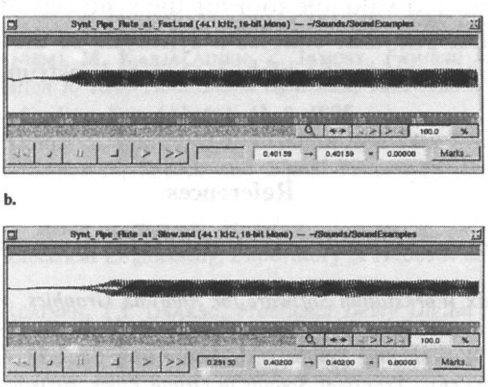


Fig. 16. Time-domain analyses of synthetic sound patterns:
a. fast attack,
b. slow attack.

Table 1. Parameters of the waveguide synthesis of a flute pipe.

scaler	scaler	delay line	delay line	scaler	scaler	filter	filter
n	$r1$	$dl1$	$dl2$	$r2$	g	b_0	a_1
1	0.1	8	29	0.56	0.5	−0.3	−0.8

flute pipe in the elaborated model (see Fig. 7). The gain in the feed-back loop in the model is determined by the coefficient $r1$ and $r2$. The level of the input signal is proportional to the changes of the parameter g.

8. Conclusions

Synthesis of musical sounds by physical modelling becomes feasible due to the increase in the power of the computer equipment. On the basis of a new technology also new synthesis algorithms emerged; the waveguide synthesis that starts to be known as a method created on the basis of physics of an instrument or an instrument family.

The relatively high computing power of the modern DSP circuitry allows such an instrument to run in real-time. The performed analyses and listening tests show that the synthesized sound using the waveguide method is more similar to the natural sound of the organ pipe than sounds synthesized by numerical solution of "classical" equations describing the acoustical behaviour of pipes. What is most important, this observation concerns also the initial stage of a sound rise being critical to the subjective assessment of naturalness of the organ sound produced by pipes excited with various types of air blow. Taking into account that the waveguide method is applicable in real time with the use of popular digital signal processors, it can be considered to be a new and valuable tool for the synthesis of the organ pipe sound. This method is the first one that enables a synthesis of the organ pipe sound with a quality comparable to sounds of widely recognized instruments with a mechanical tracker action.

References

[1] A.H. Benade, *Equivalent Circuits for Conical Waveguides,* J. Acoust. Soc. Amer., **83**, 5, 1764–1769 (1988).

[2] J.W. Beauchamp, *Unix Workstation Software for Analysis, Graphics, Modification, and Synthesis of Musical Sounds,* 94th Audio Eng. Soc. Conv., Preprint 3479, Berlin (1993).

[3] M. Le Brun, *Digital Waveshaping Synthesis,* J. Audio Eng. Soc., **27**, 4, 250–266 (1979).

[4] R.S. Caddy, H.F. Pollard, *Transient Sounds in Organ Pipes,* Acustica, **7**, 277–280 (1957).

[5] J.M. Chowning, *The Synthesis of Complex Audio Spectra by Means of Frequency Modulation,* J. Audio Eng. Soc., **21**, 7, 526–534 (1973).

[6] J.W. Coltman, *Sounding Mechanism of the Flute and Organ Pipe,* J. Acoust. Soc. Amer., **44**, 4, 983–992 (1968).

[7] J.W. Coltman, *Jet drive mechanisms in edge tones and organ pipes,* J. Acoust. Soc. Amer., **60**, 3, 725–733 (1976).

[8] P.R. Cook, *Identification of Control Parameters in an Articulatory Vocal Tract Model with Applications to the Synthesis of Singing*. Ph.D. Dissertation, Elec. Eng. Dept., Stanford University (1990).

[9] P.R. Cook, *A Meta-wind-Instrument Physical Model, and Meta-Controller for Real Time Performance Control*, Proc. of the ICMC, San Jose, California (1992).

[10] N.H. Fletcher, *Nonlinear interactions in organ flue pipes*, J. Acoust. Soc. Amer., **56**, 2, 645–652 (1974).

[11] N.H. Fletcher, *Transients in the Speech of Organ flue pipes — A Theoretical Study*, Acustica, **34**, 224–233 (1976).

[12] L. Hiller, P. Ruiz, *Synthesizing Musical Sounds by solving the Wave Equation for Vibrating Objects*, J. Audio Eng. Soc., **19**, 6 (1971).

[13] P. Kleczkowski, *Group Additive Synthesis*, Computer Music J., **13**, 1, 12–20 (1989).

[14] B. Kostek, A. Czyżewski, *Computer Modelling of the Pipe Organ Valve Action*, 92nd Audio Eng. Soc. Conv., Preprint 3266, Vienna (1992), J. Audio Eng. Soc. (Abstr.), **40**, 5 (1992).

[15] B. Kostek, A. Czyżewski, *Investigation of articulation features in organ pipe sound*, Archives of Acoustics, **18**, 2, 3, 417–434 (1993).

[16] B. Kostek, A. Czyżewski, S. Zieliński, *New Approach to the Synthesis of Organ Pipe Sound*, 98th Audio Eng. Soc. Conv., Preprint 3957, Paris (1995), J. Audio Eng. Soc. (Abstr.), **41**, 5, (1995).

[17] W. Lottermoser, *Acoustical Design of Modern German Organs*, J. Acoust. Soc. Amer., **29**, 6, 682–689, (1957).

[18] M.E. McIntyre, R.T. Schumacher, J. Woodhouse, *On the oscillation of musical instruments*, J. Acoust. Soc. Amer., **74**, 5, 1325–1345 (1983).

[19] A.W. Nolle, T.L. Finch, *Starting transients of flue organ pipes in relation to pressure rise time*, J. Acoust. Soc. Amer., **91**, 4, 2190–2202, (1992).

[20] A. Rakowski, E.G. Richardson, *Eine Analyse des Intonierungsvorganges bei Orgeln*, Gravesaner Blätter, 15, 16, 46–58 (1960).

[21] E.G. Richardson, *Technical Aspects of Sound*, Elsevier Publishing Company, New York, (1953).

[22] E.G. Richardson, *The Transient Tones of Wind Instruments*, J. Acoust. Soc. Amer., **26**, 6, 960–962, (1954).

[23] J.O. Smith III, *Physical Modeling using Digital Waveguides*, Computer Music Journal, special issue on *Physical Modeling of Musical Instruments*, Part I, **16**, 4, 74–91, Winter, (1992).

[24] B.T.G. Tan, S.L. Gan, S.M. Lim, S.H. Tang, *Real-Time Implementation of Double Frequency Modulation (DFM) Synthesis*, J. Audio Eng. Soc., **42**, 11, 918–926 (1994).

[25] V. Välimäki, J. Huopaniemi, M. Karjalainen, Z. Janosy, *Physical Modeling of Plucked String Instruments with Application to Real-Time Sound Synthesis*, 98th Audio Eng. Soc. Conv., Preprint 3956, Paris (1995), J. Audio Eng. Soc. (Abstr.), **41**, 5, 1995.

[26] V. Välimäki, M. Karjalainen, Z. Janosy, U.K. Laine, *A Real-Time DSP Implementation of a Flute model*, Proc. Int. Conf. Acoustics, Speech and Signal Processing (ICASSP'92), San Francisco, California, USA (1992).

[27] V. Välimäki, *Fractional Delay Waveguide Modeling of Acoustic Tubes*, Helsinki University of Technology, Faculty of Electrical Engineering, Laboratory of Acoustic and Audio Signal Processing, Report 34 (1994).

[28] V. Välimäki, M. Karjalainen, T.I. Laakso, *Fractional Delay Digital Filters*, Proc. IEEE Intern. Symposium on Circuits and Systems, Chicago, Illinois (1993).