International Computer Music Conference (ICMC'94), Sept. 12–17, 1994, Aarhus, Denmark

Virtual Instruments in Virtual Rooms— A Real-Time Binaural Room Simulation Environment for Physical Models of Musical Instruments

Jyri Huopaniemi[1], Matti Karjalainen[1,2], Vesa Välimäki[1,2], and Tommi Huotilainen[1]

[1]Helsinki University of Technology
Acoustics Laboratory
Otakaari 5A, FIN-02150 Espoo, Finland

[2]CARTES
(Computer Arts Centre at Espoo)
Ahertajankuja 4, FIN-02100 Espoo, Finland

Jyri.Huopaniemi@hut.fi, Matti_Karjalainen@hut.fi,
vpv@vipunen.hut.fi, Tommi@helmholtz.hut.fi

Abstract

In this paper high-quality model-based sound synthesis of plucked string and woodwind instruments is combined with room simulation and auralization techniques. The result is a real-time virtual three-dimensional room where the listener and multiple virtual instruments can be moved. The system was designed using signal processors in a multiprocessing environment. Sound radiation patterns of musical instruments were measured and direction-dependent filtering was applied to instrument simulation models. Auralization is achieved by measuring and using head-related transfer functions. The binaural three-dimensional audio output of the virtual environment is directed to headphone listening. The environment is controlled by a mouse-operated user interface while the virtual instruments are played via MIDI.

1 Introduction

Digital waveguide and delay line modeling of wave propagation has been an interesting and rewarding field in computer music technology. Artificial reverberation generated by digital delay line networks has been studied for over three decades (see, e.g., [Schroeder, 1962]). Physical modeling of musical instruments using digital waveguides has shown to be one of the most important inventions in sound synthesis in the past two decades [Smith, 1987] [Smith, 1992]. Computer-based simulation of room acoustics has also adopted techniques based on digital delay lines. However, applications that combine different implementations of digital waveguides, such as model-based sound synthesis attached to a virtual acoustical environment, are still quite few.

This paper describes an implementation of a real-time virtual sound and acoustics simulator. As opposed to previously introduced virtual acoustical real-time environments (see e.g., [Foster *et al.*, 1991] [Astheimer, 1993] [Wenzel, 1994]), this paper combines several areas of interest and results in a new way of looking at virtual reality technology.

This environment consists of moving sound sources that are physical models of musical instruments and a freely moving listener who is controlling the system. The models, a virtual acoustic guitar and a virtual flute, are played via MIDI and their movements are controlled from a keyboard. The radiation directivity of the instruments is taken into account by applying direction-dependent filtering.

The listener wearing headphones captures the binaural three-dimensional sound created by the environment. *Auralization* [Kleiner *et al.*, 1993] is achieved by applying binaural digital filters designed from empirically measured *head-related transfer functions* (HRTF) [Blauert, 1983] [Wightman and Kistler, 1989] [Møller, 1992].

The paper is organized as follows. In Section 2, the digital waveguide models for the acoustic guitar and the flute are reviewed. The major improvement over the former models is that now the sound radiation directivity of the instruments is incorporated.

In Section 3, techniques for modeling room acoustics and binaural hearing are discussed. For room simulation we have adopted the *image-source method* [Allen and Berkley, 1979] combined with a method for approximating late reverberation. Measurements of HRTFs and implementations of the binaural digital filters are discussed with emphasis on real-time sound processing.

Section 4 presents the multiprocessor platform that is used for implementing the virtual environment. The QuickC30/C40 DSP programming environment is discussed.

In Section 5, we reveal the secrets behind the designed virtual environment—how and where and why the system is running, and what does it sound like.

The final section summarizes our results and makes suggestions for future work in the field of virtual sound reality.

2 The Physical Models

Sound synthesis based on physical modeling has recently become one of the most active fields of computer music research. The development of *digital waveguide* modeling techniques [Smith, 1992] has led to computationally efficient synthesis algorithms. The foremost goal in applying these methods to musical instrument modeling has been to achieve real-time sound synthesis on modern signal processors [Karjalainen and Laine, 1991]. The research carried out at the Acoustics Laboratory of the Helsinki University of Technology has led to high-quality real-time synthesis algorithms of, e.g., the acoustic guitar [Karjalainen *et al.*, 1993] and the flute [Välimäki *et al.*, 1992].

2.1 The Guitar Model

The block diagram of the guitar model is depicted in Fig. 1. The output of the excitation filter $E(z)$ represents the signal produced by a finger or a plectrum touching a string. This signal is fed into the string model $S(z)$ that is presented in Fig. 2. This version of the string algorithm, derived from a waveguide model, is called the *extended Karplus–Strong model*. The feedforward loop (on the left in Fig. 2) models the effect of the pluck position. The length of the delay line z^{-L} determines the fundamental frequency of the synthetic tone and the loop filter $H_l(z)$ brings about the damping and dispersion effects.

The body model $B(z)$ is in principle a high-order filter that adds the most important resonances and antiresonances of the body to the synthetic sound [Karjalainen *et al.*, 1991]. Due to the commutativity of the blocks depicted in Fig. 1 the impulse response $b(n)$ of the body may be incorporated in the excitation sequence as proposed in [Smith *et al.*, 1991] [Karjalainen *et al.*, 1993] [Smith, 1993].

The filter $R(z,\theta_i)$ is a single-input-multiple-output filter that models the radiation directivity of the guitar to different angles θ_i $(i = 0,1,2,\ldots,K)$. Thus, the output of the guitar model is a vector of signals, that is $\mathbf{y}(n) = y_i(n)$ $(i = 0,1,2,\ldots,K)$. A practical implementation would include a small number of simple digital filters.

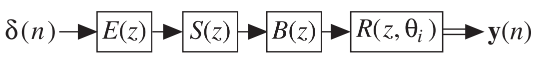

Fig. 1. The block diagram of the guitar model.

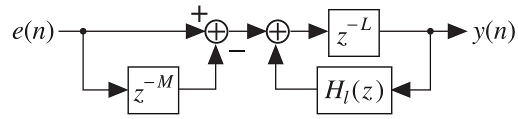

Fig. 2. The Extended Karplus–Strong Model.

2.2 The Flute Model

The waveguide flute model is illustrated in Fig. 3. It can be divided into linear and nonlinear parts. The linear part represents the bore of the instrument and the reflection from the first open tone hole. This is similar to a resonator model of any woodwind instrument. The air jet model simulates the interaction of the excitation and the wave that propagates in the bore. This part of the system includes a nonlinearity which is characteristic of flute instruments or flue organ pipes.

The input signal $e(n)$ is a white noise sequence. The model produces two output signals, $y_1(n)$ and $y_2(n)$. The former corresponds to the noisy sound that radiates from the embouchure hole of the flute. The latter is radiated from the first open tone hole of the flute. For further details see [Välimäki *et al.*, 1992] or [Vuori and Välimäki, 1993].

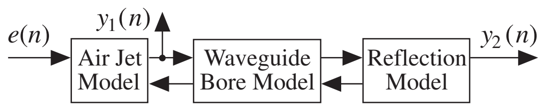

Fig. 3. The waveguide flute model with two outputs.

2.3 Modeling of Sound Radiation Patterns of Musical Instruments

Most musical instruments exhibit complex sound radiation patterns. In string instruments different mode frequencies of the body have their own patterns such as monopoles, dipoles, or quadrupoles, and their combinations. In brass wind instruments the bell has increasing directivity towards high frequencies. In the flute there are inherently two points of radiation, the embouchure hole and the first open finger hole (or the end opening). In other wind instruments most of the radiation originates from a single hole, thus leading to a relatively omnidirectional pattern. Another noticeable factor of directivity is the masking by (and to a less degree the reflection from) the player of the instrument.

Computational modeling of the detailed directivity patterns is out of the capacity of real-time DSP sound synthesis. It is therefore important to find simplified models that are efficient from the signal processing point of view and as good as possible from the perceptual point of view. In the following we consider three strategies.

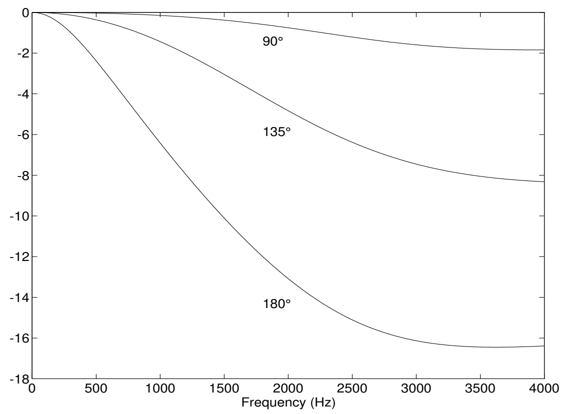

Fig. 4. Direction-dependent radiation of the acoustic guitar modeled with a 2nd order IIR filter. The amplitude scale is in dB.

Directional Filtering

A direction-dependent digital filter may be attached to each path from the source to the listener. Moving and rotating sources can be modeled by changing the filter parameters of the paths in a proper way (e.g., the Leslie effect of a rotating loudspeaker can be simulated). The directional filtering method was studied for two instruments: the trumpet and the acoustic guitar. In the case of the trumpet the radiation response from the bell was measured to different angles by using an impulse-like excitation at the mouthpiece. We came to the conclusion that even first or second-order directivity filters give useful results thus leading to an efficient implementation.

Figure 4 depicts the modeling of direction-dependent radiation of the acoustic guitar (in the horizontal plane) relative to the main axis radiation. Shown in the figure are magnitude spectra for second-order IIR filters at azimuth angles 90°, 135°, and 180°. The reference magnitude spectrum at 0° is assumed to be $H(z)=1$. The lowpass characteristic is noticeably increased as the relative angle is greater. The measurement was carried out by exciting the bridge of the instrument by an impulse hammer and by registering the reference response at 0° and the related response in various directions. The measured reference and the directional response were fitted separately with first or second-order AR models. A simple division of the models was performed to get the pole-zero directivity filter.

The upper part in Fig. 5 depicts the impulse responses measured from azimuth angles 0° and 180°. The lower part shows the response from 0° filtered with a 1st order IIR directivity filter and the actual measured response from 180° azimuth. Note that the spectral slopes are nearly the same, as was expected. In this example the directional filter is

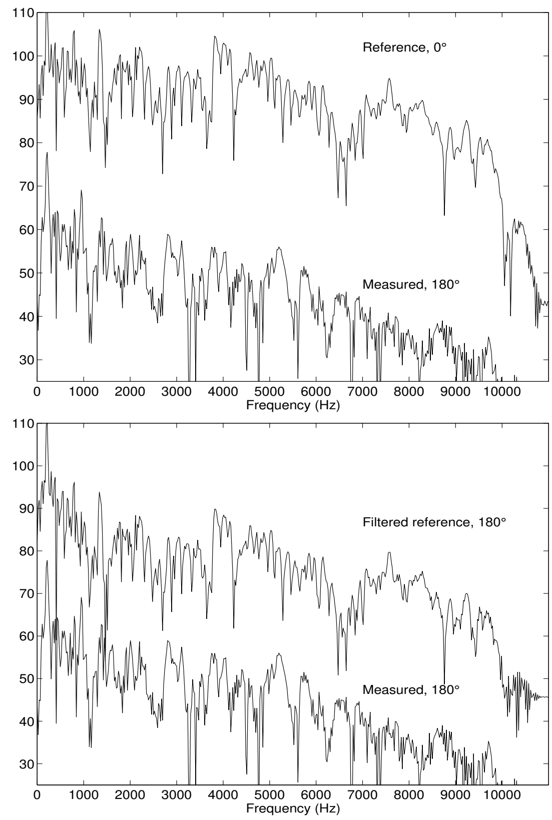

Fig. 5. Modeling of guitar sound radiation with a 1st order IIR filter. The relative amplitude scale is in dB.

$$R(z) = \frac{0.2041 - 0.1431z^{-1}}{1 - 0.4389z^{-1}} \qquad (1)$$

When zooming to the details of the lowest resonance modes we notice, as described in [Fletcher and Rossing, 1991], that the individual modes behave differently. To model such details a relatively high-order directional filter is needed. It is important to notice that due to the critical band frequency resolution an *auditory smoothing* may be applied to the responses before directional filter estimation. This helps to reduce the order of the filter.

Set of Elementary Sources

In this method the radiation pattern of an instrument is approximated by a small number of elementary sources such as monopoles or dipoles. We applied this approach to the flute where there are inherently two point sources of sound radiation, the embouchure hole and the first open tone hole. In general the method is computationally expensive if a large number of paths to the receiver is needed since each new elementary source adds a new set of path filters.

Direction-Dependent Excitation

In the case of commutative excitation (e.g., a plucked string instrument model) the directivity filtering may be included in the composite excitation in a way similar to the inclusion of the (early) room response [Smith, 1993]. The problem with this method is that as many instrument or string model instances are to be run in parallel as there are directions to be included. This limits the number of simulated directions, e.g., to the six main directions of the Cartesian coordinate system. Even then it is computationally very inefficient.

The considerations above as well as our experiments have shown that the directional filtering technique is normally the most practical one. A first or second-order filter approximation is often a satisfactory solution in a real-time implementation. In some cases such as the flute a set of elementary sources may be attractive.

3 Auralization and Modeling of Room Acoustics

The task of approximating a sound field is the basis of computational simulations of room acoustics. Several mathematical algorithms to accomplish this have been developed, most common being *ray-tracing*, *image-source* methods, and their hybrids. The goal in ray-tracing algorithms is to model the sound waves emitted from a sound source and follow their propagation and reflections from boundary elements; the walls, the ceiling and the floor. In the case of rectangular rooms it is, however, more convenient from the computational point of view to use the image-source method.

3.1 Image-Source Method

The image-source method is based on the well-studied theory of using phantom images of the desired source position in a bounded space. The *M*th-order reflections of a sound source can be calculated by adding *N* images of the room and the source and calculating the length of the path between the two points.

The sound radiating from the source and its images is attenuated due to three reasons: 1) absorption of the boundary materials, 2) the distance between the points of interest, and 3) the frequency-dependent air absorption. These factors have to be taken into account in accurate simulation of room acoustics. In DSP terms frequency-dependent filtering and attenuation can be accomplished with low-order digital filters. The image-source method that we have implemented assumes the modeled room to be rectangular and all the reflections to be specular.

Aspects in Real-Time Room Simulation

One remarkable drawback in the use of the image-source method is the computational explosion resulting from the calculation of higher-order reflections. For example, in a moderate-sized room (i.e., $L \times W \times H = 5 \times 3 \times 3\ m^3$) an impulse response of 300 ms would need images of order M = 20 resulting in $N = n \cdot (n-1)^{M-1} = 6 \cdot 5^{19} = 1.1 \cdot 10^{14}$ separate image sources, where n is the number of boundaries. When a limited time is allowed for the computation of the room impulse response, most of the described simulation methods become unavailable. As in our case, a realistic task even for a fast signal processor would be to calculate and update up to second or third-order reflections from a simple room using the image-source method. Thus, recursive methods for designing natural yet perceptually accurate reverberation have to be implemented. Luckily it has been shown [Barron, 1971] that about 80 ms of the first reflections is sufficient to create a spatial impression of reverberation. The late reverberation can be modeled artificially as is discussed in the following section.

3.2 Artificial Reverberation Using Recursive Methods

A framework for artificial reverberation was built by Schroeder using parallel comb filters and series allpass filters [Schroeder, 1962]. Most of the modern reverberation implementations exhibit complex combinations of both comb and allpass filter structures [Moorer, 1979] [Moore, 1983] [Jot and Chaigne, 1991]. We have studied different aspects in reverberation modeling and ended up with a new structure, which is computationally efficient yet gains good results.

A novel implementation that clearly follows the guidelines from earlier reverberator designs has been implemented. The combined image-source and late reverberation filter is depicted in Fig. 6. The fundamental idea is 1) to use the direct sound and image-source method for accurate modeling of the first reflections, and 2) to use outputs from the direct and imaged sources as inputs to the late reverberation filter.

The input to the reverberation model is the direction-independent output from the musical instrument model. The upper delay line represents the time delay of the direct and imaged sounds. The direct sound and the imaged sounds are filtered with low-pass filters $G_0(z) - G_N(z)$ that include air absorption, boundary material absorption, and the radiation directivity of the musical instrument models.

The architecture used in the late reverberation model is similar to the guitar bridge filter model described in [Smith, 1993]. The idea is to feed each

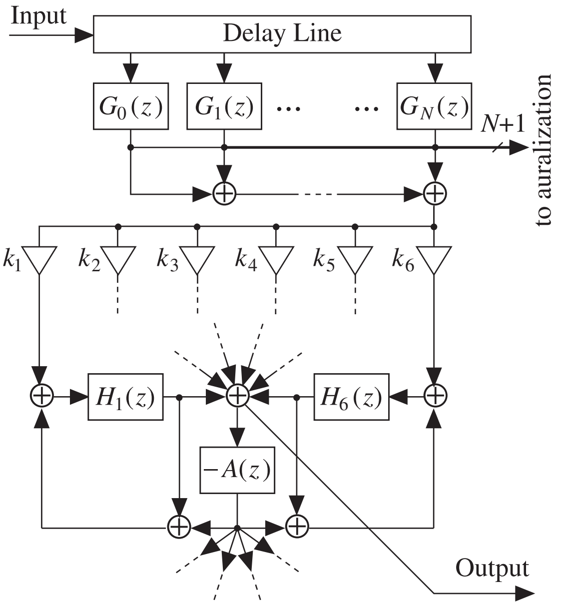

Fig. 6. A reverberation model using combined image-source method and artificial reverberation.

of the six recursive comb filter loops with the negated sum from the five other loops. The blocks $H_1(z) - H_6(z)$ represent a combination of pure delays and a low-pass structure.

Our implementation of the reverberation filter structure includes an allpass section *A(z)*, the output of which is distributed as illustrated in Fig. 6.

3.3 Auralization

Auralization refers to binaural processing of sound signals so that an illusion of three-dimensional sound space is created [Kleiner *et al.*, 1993]. Real-time auralization involves not only digital signal processing but also good knowledge of the mechanisms of human hearing and especially spatial perception of sound. Extensive studies of spatial hearing (see, e.g., [Blauert, 1983]) have posed that most of the spatial localization cues are caused by three facts: 1) *interaural time difference* (ITD), 2) *interaural amplitude difference* (IAD), and 3) the direction-dependent filtering by the outer ear (the pinnae) and the torso. An approximation to the ITD can be found in [Blauert, 1983] and is stated as:

$$\Delta s = \frac{D}{2} \cdot (\varphi + \sin \varphi) \tag{2}$$

where Δs represents the difference in the path length of the arriving sound, D is distance between the ears and φ is the incident angle of the coming wave.

To avoid localization conflicts that often occur in the median plane where ITDs and IADs are at their minimum, accurate modeling of the direction-dependent filtering has to be accomplished. One method that has been successfully mastered in recent years is based on measurements and implementations of *head-related transfer functions* (HRTF). This topic is covered in the next section.

HRTF Measurements

A HRTF represents a free-field transfer function from a fixed point to the entrance of the ear canal. Measured HRTFs have distinctive characteristics showing different torso and pinnae cavity reflections and give information about human sound localization.

HRTF measurements were carried out to provide data for the direction-dependent filtering in the auralization model. The method that was used is described in [Wightman and Kistler, 1989]. The probe microphone transfer function can be described in the frequency domain as

$$Y_1(\omega) = X_1(\omega)L(\omega)F(\omega)M(\omega) \tag{3}$$

where $Y_1(\omega)$ represents the probe microphone's response of the test signal driven to a loudspeaker ($X_1(\omega)$), $F(\omega)$ represents the free-field-to-ear-canal transfer function, $L(\omega)$ the loudspeaker transfer function, and $M(\omega)$ the microphone transfer function. A similar measurement is made using headphones resulting in a transfer function $Y_2(\omega) = X_2(\omega)H(\omega)M(\omega)$, where $H(\omega)$ represents the headphone-to-ear-canal transfer function. When both probe transfer functions are set to equal, the desired filter transfer function is given by

$$T(\omega) = \frac{L(\omega)F(\omega)}{H(\omega)} \tag{4}$$

It is clearly seen that the frequency characteristics of both the loudspeaker and the headphones are uncompensated in this form. From this equation the loudspeaker transfer function remains to be compensated to get the actual HRTF for headphone reproduction. Calculations are performed by deconvolution in the frequency domain.

Measurement Equipment

The equipment used for measurements were especially designed and built for this purpose. The probe microphone system consisted of a Sennheiser KE4-211-8 condenser microphone coupled to a polyethylene probe tube with an outer diameter of 1.77 mm. A set of two microphones with one stereo preamplifier were built to allow measurements from both ears simultaneously.

The measurement software was programmed in Common Lisp using the QuickSig [Karjalainen, 1990] software. The test signal used was a flat-spec-

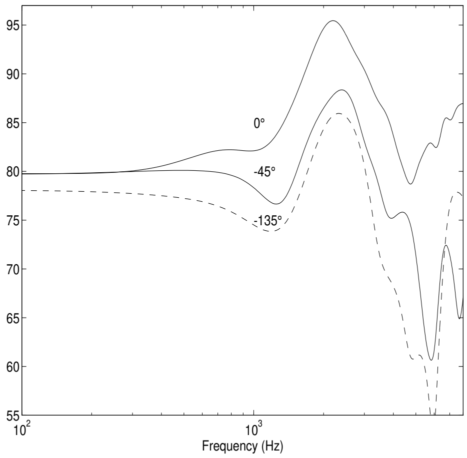

Fig. 7. Measured HRTFs from three azimuth angles. The relative amplitude scale is in dB.

trum minimum peak factor noise burst of length 2048 samples at a sampling rate of 22.05 kHz. The stored *head-related impulse-response* (HRIR) was averaged from 100 measurements. Measurements from 8 different azimuth positions for 10 test persons were carried out. The results are similar to the ones presented in [Wightman and Kistler, 1989].

HRTF Filter Design

A reconstruction of FIR filters from measured HRTFs was performed. A more detailed analysis of the HRTFs will be carried out in near future. An example set of 128-tap FIR filters were calculated. Figure 7 shows the magnitude spectra of the filters from azimuth positions 0°, -45°, and -135°.

4 Multiprocessor DSP Environment

A large amount of computational power is required to achieve real-time performance for binaural room simulation and model-based sound synthesis. The QuickC30/C40 is a multiprocessor DSP environment developed at the Acoustics Laboratory of the Helsinki University of Technology [Karjalainen, 1992]. The system contains several signal processors and allows for complicated real-time applications to be realized.

4.1 Hardware

The QuickC30/C40 multiprocessor system hardware consists of multiple signal processor cards and an Apple Macintosh host computer. The host computer communicates with the processors through a parallel digital I/O interface and a handshaking logic. The multiprocessor system is based on Loughborough Sound Images' MDC40S modules which

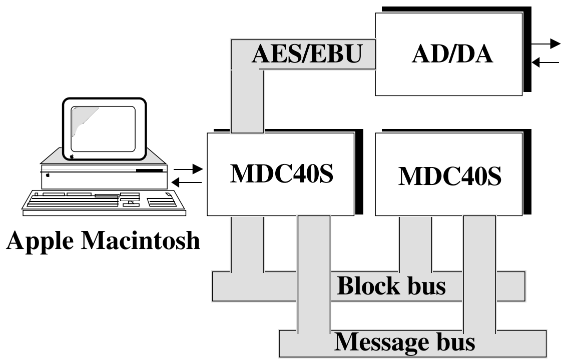

Fig. 8. The QuickC30/C40 multiprocessor system consisting of two TMS320C40 signal processors.

adhere to Texas Instruments TIM-40 specification [Weir, 1992]. The latter specification is in turn based on the Texas Instruments TMS320C40 floating-point signal processor [Texas Instruments, 1991] that has been developed for parallel processing. The processors communicate with each other through communication ports. We have designed special mother cards to simplify the linking of the MDC40S modules. The environment includes 16 or 18-bit AD/DA converters by Crystal Semiconductor.

4.2 Software

The QuickC30/C40 environment has been implemented entirely within the Common Lisp Object System (CLOS) for the Apple Macintosh [Karjalainen, 1992]. The software of the multiprocessor system consists of a message passing protocol based on interrupt routines supporting parallel processing for the communication between the signal processors and the host computer. The system has two separate buses for communication between the C40 processors, one for short messages and another for block transfers as illustrated in Fig. 8.

5 The Virtual Environment

The designed environment was built and programmed to function on a Apple Macintosh Quadra 950 computer with two TMS320C40 signal processors. The room simulation and auralization software as well as the musical instrument models were programmed in Common Lisp using its object-oriented extensions CLOS and the QuickC30/C40. The current sampling rate used is 22.05 kHz.

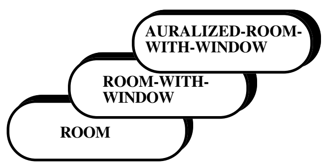

Fig. 9. The virtual environment object hierarchy.

```
(make-instance 'auralized-room-with-window
     :front 0.0
     :back 20.0
     :bottom 0.0
     :top 4.0
     :left 0.0
     :right 10.0
     :sources (make-instance 'guitar
          :x 3.0 :y 3.5 :z 0.0
          :dx 1.0 :dy 0.0 :dz 0.0)
     :receivers (make-instance
        'receiver-item
          :x 10.0 :y 5.0 :z 0.0
          :dx -1.0 :dy 0.0 :dz 0.0))
```

Fig. 10. An example of creating a room simulator object.

5.1 The Building Blocks

The virtual room model consists of three separate building blocks, objects, that can be instantiated separately or together forming the complete system (Fig. 9). The room object contains the basic definitions for the dimensions and for the positions of the listener and the source objects. It also creates the *n* image sources of the designated sound source for accurate modeling of first reflections. The *sources* slot of the object refers to the musical instrument model (guitar or flute), which will be instantiated to run on one of the C40s.

The next class in the hierarchy, *room-with-window*, inherits the properties from the *room* object and adds routines for manipulating the graphical source and listener objects, and for adjusting the room size.

The two previously described object classes are defined and accessed in the host computer's memory. They provide links to the low-level signal processor environment, and the links are established in the following object class.

The *auralized-room-with-window* class inherits the room and graphics properties from previous classes. The communication routines between the two signal processors and the host computer are controlled within this object class. An example of creating a room simulation object is shown in Fig. 10.

Event Controlling

The graphical events are the controllers of the calculations performed in the signal processors. Moving or re-orienting the graphical objects on the screen interrupt the processors by updating the source, source image and listener positions. The needed information for auralized sound processing is: 1) the distance between the source or its image and the listener, 2) the relative elevation angles, and 3) the relative azimuth angles of the listener and the sound source. The MIDI interface is programmed according to [Karjalainen *et al.*, 1993].

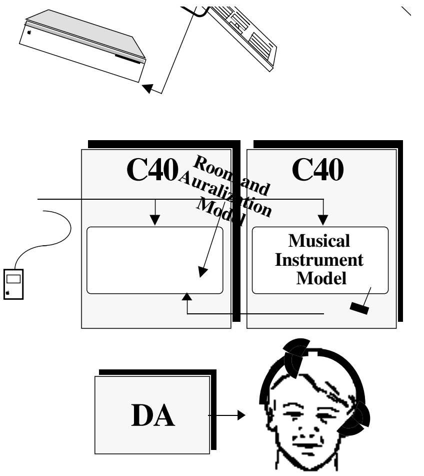

Fig. 11. Overview of the designed environment.

6 Summary and Future Work

A virtual binaural room simulator for physical models of musical instruments was presented. The system produces virtual three-dimensional audio output for stereo headphone listening in real-time. Radiation directivity of the musical instruments has been incorporated by applying direction-dependent low-order digital filters. Room simulation is performed with a combined image-source and late reverberation model. Binaural processing is carried out using digital filter approximations of measured HRTFs.

Future areas of interest in this research project include accurate non-real-time modeling of the environment and a 3-D graphical network interface to the room simulation system. We are also planning further investigations on the modeling of musical instrument radiation patterns.

Acknowledgments

The authors would like to acknowledge the persons taking part in numerous HRTF experiments and measurements. This work has been financed by the Academy of Finland.

References

[Allen and Berkley, 1979] Jont B. Allen and David A. Berkley. Image method for efficiently simulating small-room acoustics. *J. Acoust. Soc. Am.*, **65** (4), pp. 943–950, April 1979.

[Astheimer, 1993] Peter Astheimer. What you see is what you hear—acoustics applied in virtual worlds. In *Proc. IEEE Symposium on Research Frontiers in VR,* San Jose, CA, pp. 100–107, Oct. 25–26, 1993.

[Barron, 1971] M. Barron. The subjective effects of first reflections in concert halls—The need for lateral reflections. *J. Sound Vib.*, **15** (4), pp. 475–

494, 1971.

[Blauert, 1983] Jens Blauert. *Spatial Hearing*. M.I.T. Press, Cambridge, MA, 1983.

[Fletcher and Rossing, 1991]. Neville H. Fletcher and Thomas D. Rossing. *The Physics of Musical Instruments,* Springer Verlag, New York, pp. 223–226, 1991.

[Foster *et al.*, 1991] Scott H. Foster, Elizabeth M. Wenzel, and R. M. Taylor. Real time synthesis of complex acoustic environments. In *Proc. 1991 IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics,* New Paltz, NY, Oct. 20–23, 1991.

[Jot and Chaigne, 1991] Jean-Marc Jot and Antoine Chaigne. Digital delay networks for designing artificial reverberators. Presented at the *90th AES Conv., preprint 3030 (E-2),* Paris, Feb. 19–22, 1991.

[Karjalainen, 1990] Matti Karjalainen. DSP software integration by object-oriented programming: A case study of QuickSig. *IEEE ASSP Magazine*, pp. 21–31, April, 1990.

[Karjalainen, 1992] Matti Karjalainen. Object-oriented programming of DSP processors: A case study of QuickC30. In *Proc. 1992 IEEE Int. Conf. Acoust., Speech, and Signal Processing (ICASSP'92)*, San Francisco, CA, vol. V, pp. 601–604, March 23–26, 1992.

[Karjalainen and Laine, 1991] Matti Karjalainen and Unto K. Laine. A model for real-time sound synthesis of guitar on a floating-point signal processor. In *Proc. 1991 IEEE Int. Conf. Acoust., Speech, and Signal Processing (ICASSP'91)*, Toronto, Canada, pp. 3653–3656, May 2–5, 1991.

[Karjalainen *et al.*, 1991] Matti Karjalainen, Unto K. Laine, and Vesa Välimäki. Aspects in modeling and real-time synthesis of the acoustic guitar. In *Proc. 1991 IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics*, New Paltz, New York, Oct. 20–23, 1991.

[Karjalainen *et al.*, 1992] Matti Karjalainen, Vesa Välimäki, Toomas Altosaar, and Seppo Helle. The QuickSig system and its computer music applications. In *Proc. ICMC'92,* San Jose, CA, pp. 390–391, Oct. 14–18, 1992.

[Karjalainen *et al.*, 1993] Matti Karjalainen, Vesa Välimäki, and Zoltán Jánosy. Towards high-quality sound synthesis of the guitar and string instruments. In *Proc. ICMC'93*, Tokyo, Japan, pp. 56–63, Sept. 10–15, 1993.

[Kleiner *et al.*, 1993] Mendel Kleiner, B.-I. Dalenbäck, and P. Svensson. Auralization – an overview. *J. Audio Eng. Soc.* **41**(11), pp. 861–875, 1993.

[Moore, 1983] F. Richard Moore. A general model for spatial processing of sounds. *Computer Music J.*, **7** (3), 1983.

[Moorer, 1979] James A. Moorer. About this reverberation business. *Computer Music J.*, **3** (2), pp. 13–28, 1979.

[Møller, 1992] Henrik Møller. Fundamentals of binaural technology. *Applied Acoustics* **36** (3&4), pp. 171–218, 1992.

[Schroeder, 1962] Manfred R. Schroeder. Natural sounding artificial reverberation. *J. Acoust. Soc. Am.*, **10** (3), 1962.

[Smith, 1985] Julius O. Smith. A new approach to digital reverberation using closed waveguide networks. In *Proc. ICMC'85,* Vancouver, Canada, August 1985. Reprinted in [Smith, 1987].

[Smith, 1987] Julius O. Smith. *Music Applications of Digital Waveguides*. Stanford, CA, Stanford University, Dept. of Music, CCRMA Tech. Report No. STAN–M–39, May 27, 1987.

[Smith, 1992] Julius O. Smith. Physical modeling using digital waveguides. *Computer Music J.*, **16** (4), pp. 75–87, Winter 1992.

[Smith, 1993] Julius O. Smith. Efficient synthesis of stringed musical instruments. In *Proc. ICMC'93*, Tokyo, Japan, pp. 64–71, Sept. 10–15, 1993.

[Smith *et al.*, 1991] Julius O. Smith, Matti Karjalainen, and Vesa Välimäki. Personal Communication. New Paltz, New York, Oct. 1991.

[Texas Instruments, 1991] *TMS320C4X User's Guide.* Texas Instruments Inc., 1991.

[Vuori and Välimäki, 1993] Jarkko Vuori and Vesa Välimäki. Parameter estimation of non-linear physical models by simulated evolution—application to the flute model. In *Proc. ICMC'93*, Tokyo, Japan, pp. 402–404, Sept. 10–15, 1993.

[Välimäki *et al.*, 1992] Vesa Välimäki, Matti Karjalainen, Zoltán Jánosy, and Unto K. Laine. A real-time DSP implementation of a flute model. In *Proc. 1992 IEEE Int. Conf. Acoust., Speech, and Signal Processing (ICASSP'92)*, San Francisco, CA, vol. II, pp. 249–252, March 23–26, 1992.

[Weir, 1992] R. Weir. *TIM–40—TMS320C4X Module Specification*. Texas Instruments Inc., March 1992.

[Wenzel, 1994] Elizabeth M. Wenzel. Spatial sound and sonification. *Auditory Display, Proc. Vol. XVIIII,* Addison-Wesley, 1994.

[Wightman and Kistler, 1989] Frederic L. Wightman and Doris J. Kistler. Headphone simulation of free-field listening I: Stimulus synthesis. *J. Acoust. Soc. Am.*, **85** (2), pp. 858–867, Feb. 1989.