MODELING OF REFLECTIONS AND AIR ABSORPTION IN ACOUSTICAL SPACES — A DIGITAL FILTER DESIGN APPROACH

Jyri Huopaniemi, Lauri Savioja, and Matti Karjalainen

Helsinki University of Technology
Laboratory of Acoustics and Audio Signal Processing
P.O.Box 3000, FIN-02015 HUT, Finland
Jyri.Huopaniemi@hut.fi, Lauri.Savioja@hut.fi, Matti.Karjalainen@hut.fi

ABSTRACT

In this paper, a method is presented for modeling sound propagation in rooms using a signal processing approach. Low order digital filters are designed to match to sound propagation transfer functions calculated from boundary material and air absorption data. The technique is applied to low frequency, finite difference time domain (FDTD) simulation of room acoustics and to real-time image-source based virtual acoustics.

1. INTRODUCTION

Measurement and modeling of acoustic materials is an important part of room acoustical simulation. Another factor, important at high frequencies, is the absorption of sound in the air. Based on such data, computational models of room acoustics may be constructed for non-real-time simulation or real-time auralization.

For real-time computation with varying sound source and receiver positions the image-source method is practically the only one that meets the efficiency requirements. In such a case, the sound propagation paths of early reflections will include the transfer function due to distance, reflections and the air absorption. For accurate non-real-time low-frequency simulation, methods are needed where the room is divided into elements, such as FEM (Finite Element Method), BEM (Boundary Element Method), or FDTD (Finite Difference Time Domain method). The difference method is efficient and straightforward for time domain simulation, requiring an accurate and efficient way for boundary reflection simulation.

In this paper we will present a technique where boundary material characteristics are modeled by low-order minimum-phase digital filters. The filter design takes into account the logarithmic distribution of octave frequencies used for absorption material characterization. Furthermore, the effect of air absorption can be taken into account. This technique is particularly useful in efficient room acoustics simulation and implementation of real-time virtual acoustic environments.

The paper is organized as follows. In Chapter 2, methods for acoustical material characterization are overviewed. The filter design problem and various solutions are discussed in Chapter 3. In Chapters 4 and 5, two applications of the modeling technique are presented; the low-frequency FDTD simulation and the real-time image-source calculation. Finally, in Chapter 6, conclusions and directions for future work are given.

2. CHARACTERIZATION OF ACOUSTIC MATERIALS

The traditional methods for material measurements are the standing wave tube technique [1] and the reverberant room measurement [2]. Methods that need digital signal processing are the intensity measurement technique and the transfer function method [3, 4]. Results may be given in various forms: reflection (impulse) response, reflection transfer function (reflection 'coefficient'), impedance or admittance data, or absorption data. In the literature, absorption coefficients are generally given in octave bands from 125 Hz to 4000 Hz, specifying only the magnitude of reflection. Absorption values are used, e.g., for room impulse response (RIR) prediction by computational methods such as the ray-tracing and the image-source technique [5]. If the material data is given as a measured impulse response or transfer function, the problem is to reduce it for efficient simulation.

The problem of modeling the sound wave reflection from acoustic boundary materials is a complex one. The temporal or spectral behavior of reflected sound as a function of incident angle, the scattering and diffraction phenomena, etc., makes it impossible to use numerical models that are accurate in all aspects. Depending on application, more or less approximation is needed.

In this paper we focus on DSP oriented techniques to simulate sound signal behavior in reflection and propagation. Thus we ignore many important issues, such as the angle dependency of reflection, which could be included and approximated by various methods. We also do not pay attention to the fact that material data measured in diffuse field or in impedance tube should be treated differently.

The most common characterization of acoustic surface materials is based on absorption coefficients, given for octave bands 125, 250, 500, 1000, 2000 and 4000 Hz. In contrary, the DSP-based measurement methods yield the reflection impulse response $r(t)$ or the complex-valued reflection transfer function (reflectance) $R(j\,\omega) = F\{r(t)\}$, where F is the Fourier transform. Since the absorption coefficient is the energy ratio of the absorbed and incident energies, the relation between $\alpha(\omega)$ and $R(j\,\omega)$ is given by

$$\alpha(\omega) = 1 - |R(j\,\omega)|^2 \qquad (1)$$

whereby $|R(j\,\omega)| = \sqrt{1-\alpha(\omega)}$ can be used to obtain the absolute value of the reflectance when absorption coefficients are given[1].

[1] The negative value of the square root is possible in theory but practically never happens in practice.

The relation between the normalized impedance $Z(j\,\omega)$ and the reflectance $R(j\,\omega)$ is

$$R(j\,\omega) = \frac{Z(j\,\omega) - 1}{Z(j\,\omega) + 1} \quad (2)$$

which can be used to compute $R(j\,\omega)$ when the material impedance (or admittance, its inverse value) is given. Based on the equations above, material data can be converted to $R(j\,\omega)$ or $r(t)$ for filter approximation.

3. DIGITAL FILTER MODELING OF REFLECTION AND AIR ABSORPTION

We will assume that the sound traveling in the air and reflecting from surfaces behaves linearly and that the system is time invariant. Such a system may be modeled by any linear and time-invariant (LTI) technique. Digital filtering is an especially efficient LTI technique since DSP algorithms have been developed for fast execution. With this approach the problem appears as how to simulate the boundary reflections and air absorption in an efficient, accurate, and physically plausible way. If a transfer function—reflection and/or propagation—is given based on measured impulse response or transfer function data, or an analytical model, the filter design problem is simply to search for an optimal match of data to given filter type and order, and given error criteria. If the data is available in a magnitude–only form, such as absorption coefficients for octave bands, there is an additional task of finding proper phase characteristics as well as interpolation of the sparsely given data in an acoustically meaningful way.

There exist many 'standard' modeling techniques using digital filtering approach based on AR (all-pole), MA (FIR), or ARMA (pole-zero) modeling. E.g., in Matlab [6], such filter design functions as `yulewalk`, `invfreqz`, and `cremez`, are available. The selection of a method depends on the available response data and target criteria of the design. In our experiments we have found the least mean squares fit to sparsely sampled data of absorption coefficients or air absorption using `invfreqz` very useful. The magnitude response is first converted to minimum-phase data, which is the converted to an IIR filter of desired order.

3.1. Reflection coefficients

In Fig. 1, an example of fitting low-order filters to absorption data is shown. In the figure, the magnitude responses of first-order and third-order IIR filters designed to match the corresponding target values are plotted (the target response at Nyquist frequency was approximated from the 4 kHz octave band absorption coefficient). Each set of data is a combination of two materials (second-order reflection): a) plasterboard on frame with 13 mm boards and 100 mm air cavity [7], and glass panel (6+2+10 mm, toughened, acousto-laminated) [8], b) plasterboard (same as in previous) and 3.5-4 mm fibreboard with holes, 25 mm cavity with 25 mm mineral wool [7].

3.2. Air Absorption

The effect of air absorption is an important factor in image-source calculations of large acoustical spaces, such as concert halls where higher order reflections can arrive considerably delayed from the direct sound. For real-time simulation purposes, we have approximated air absorption in the following efficient manner. Analytical expressions for attenuation of sound in air as a function of temperature, humidity and distance have been published by, e.g., [9] and

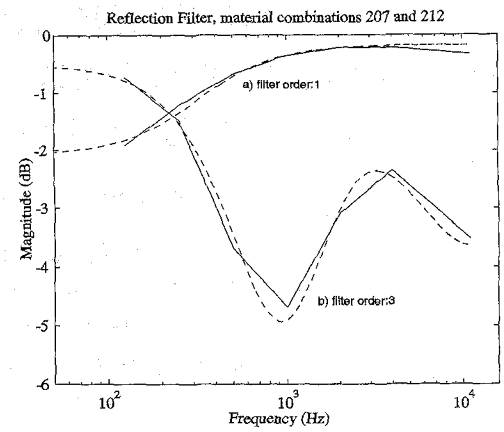

Figure 1: First-order and third-order minimum-phase IIR filters designed to given absorption coefficient data.

discussed in [10]. We computed transfer functions for air absorption at 20°C and 20% humidity for various distances. The resulting magnitude responses were fitted with first-order IIR filters designed using the technique described previously. Results of modeling for four distances from the source to the receiver are illustrated in Fig. 2.

4. LOW-FREQUENCY MODELING OF ROOM ACOUSTICS

Recently, FDTD methods have been shown to be useful for simulating the low frequency behaviour of rooms [11, 12]. In this study we use an FDTD method which is based on the digital waveguide mesh technique [13]. Computations are done in the time domain and the results are impulse responses at analysis positions.

The digital waveguide mesh is an extension of the one-dimensional digital waveguide technique [14]. This method can be expressed both as a mathematical finite difference (FD) method and as a recursive digital filter structure [15]. For this reason it is possible to state the boundary conditions as digital filters as opposed to impedance or admittance representation common in FD formulations.

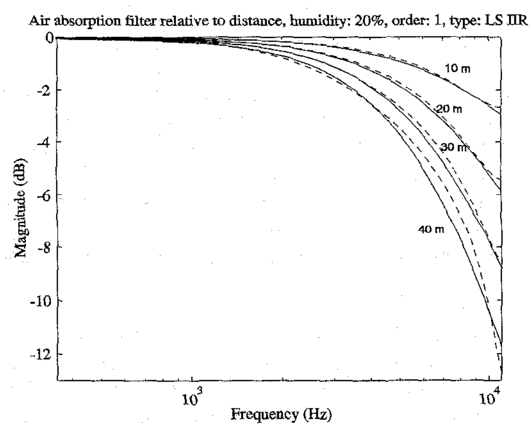

Figure 2: First-order IIR air absorption filters proportional to distance (in meters).

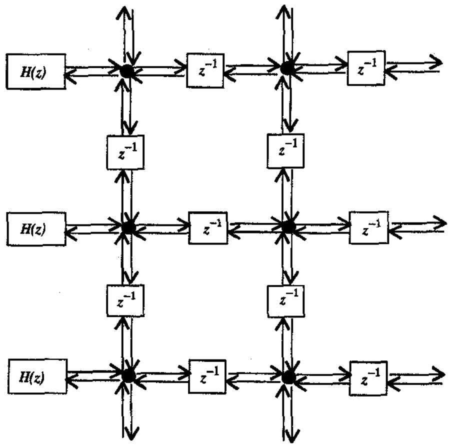

Figure 3: Two-dimensional waveguide mesh structure with boundary filters.

Figure 3 shows the basic structure of a digital waveguide mesh. In the mesh each node is connected to its neighbours by unit delays. Nodes on the boundary (on the left side of the figure) have one connection to boundary filter with transfer function *H(z)*, which represent the acoustical properties of the boundary surface. Each surface material has its own transfer function. Each boundary filter has connections only to one node inside the mesh. This structure simulates locally reacting surfaces, which is usually valid for surface materials in normal acoustic spaces. Although Fig. 3 represents a two-dimensional case this same structure is suitable also to three-dimensional meshes which are used in simulation of room acoustics.

Figure 4 represents numerical simulation results of a 2D waveguide mesh with different boundary filters and incident angles of the reflected plane wave. Figures show the magnitude attenuation of boundary filter and simulation results, also the target response is plotted. In the upper figure the wall is hard and target absorption is low and both simulation results follow the filter response reasonably well. In the lower figure the required material is more absorbing. The simulation result of 90° incident angle is not as good as in the upper figure, but the shape of the attenuation response is correct. Still the simulation of incident angle of 45° is correct. Wave propagation characteristics are direction-dependent in the waveguide mesh. The same applies also to the reflection characteristics. This anisotropic behaviour may be reduced by using an interpolation technique [16].

The sampling frequency of the waveguide mesh is determined by the spacing of the nodes, i.e., the length of the unit delays. For example if a 3D mesh is discretized by 10 cm spacing, the update frequency will be ca. 6 kHz. The simulation results are valid on the frequency band that covers approximately the lowest tenth of the whole band (update frequency). In this case also the boundary filters should be designed to cover the same band.

5. REAL-TIME ROOM ACOUSTICS RENDERING

In real-time rendering of room acoustics, efficient techniques for modeling the direct sound and early reflections have to be implemented. Although the geometrical acoustics approach is limited to high frequencies [10], it is the only possible approach for real-time

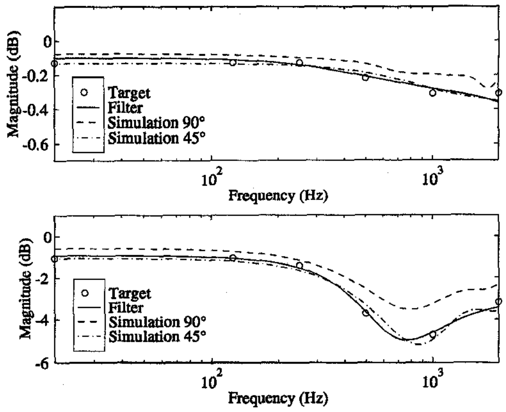

Figure 4: Reflection responses of two 2D FDTD simulations.

dynamic auralization. We have used the methods discussed in Chapter 3 to design reflection and air absorption filters. The DIVA virtual acoustics simulation system [17] uses the image-source method [18, 19] for computing the early reflections in a computer model of an acoustical space such as a concert hall. The hall used in this study is depicted in Fig. 5 [8]. The room acoustics processing is divided into three parts: direct sound, early reflections and late reverberation processing. It is known from room acoustics theory that considerable emphasis should be put on the accurate reproduction of early reflections in order to achieve natural sounding simulation.

An overview of processing for the direct sound and early reflections in the DIVA system is given in the form of a flowgraph in Fig. 6. The inputs to calculation are the room geometry and the positions of the source and the listener. Visibility checking is carried out to determine valid image sources.

The image-source calculation algorithm keeps track of the surface hits for each image, and a first-order IIR approximation is fitted to the given magnitude response data obtained by cascading the appropriate material absorption characteristics. A precalculated set of all the boundary reflection combinations has been stored to enable efficient calculation. The filtering structure is illustrated in Fig. 7. For each image a reflection filter *M(z)* and a distance-dependent air absorption filter *A(z)* is attached (for the direct sound, only air absorption is valid). To reduce the needed calculation, grouping of reflections according to arrival times can be carried out [20]. The

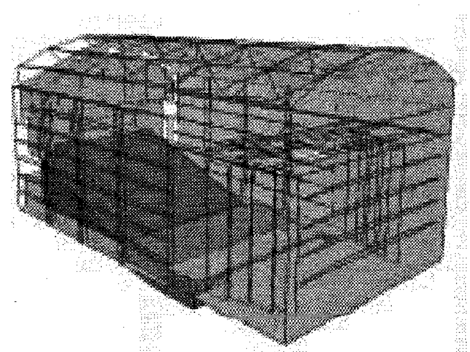

Figure 5: The geometry of the Sigyn hall [8].

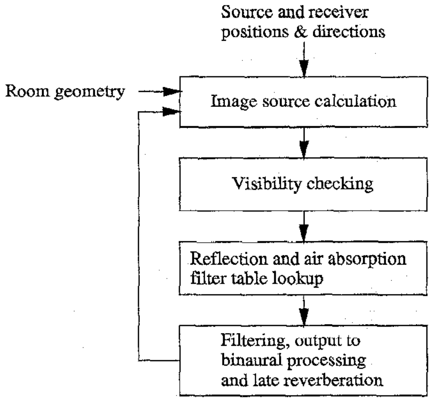

Figure 6: Image-source calculation process in the DIVA system [17].

frequency-independent attenuation of sound is taken into account by a gain factor proportional to distance ($1/r$ law [10]). The output from the early reflections calculation algorithm is directed to binaural processing and late reverberation units.

Generally the cascaded reflection filter magnitude characteristics fall into straightforward low- or highpass structures. There are, however, exceptions in the case of more complex materials such as resonators with air cavities. To enable accurate modeling also in these cases, we introduced an estimation algorithm for required filter order based on least squares error calculation.

6. CONCLUSIONS

In this paper, we have presented a signal processing approach to modeling reflections and air absorption in acoustical spaces. Low order digital filters may be designed to fit air absorption characteristics, the measured absorption coefficient data or real-valued data found from the literature. The methods described are useful in simulation of room acoustics and real-time image-source based virtual acoustics. The approach is currently limited to specular reflections. Extensions to our model could include such effects as diffuse reflections and direction-dependent absorption coefficients [21].

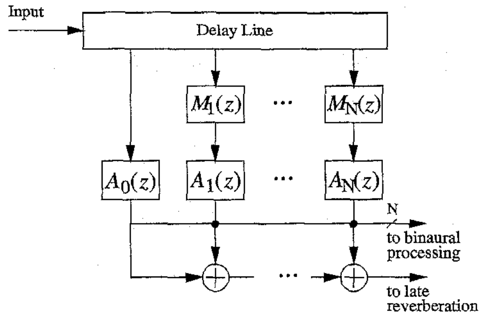

Figure 7: Structure for image-source implementation with reflection and air absorption filters.

References

1. Standard ISO 10534-1:1996, Acoustics — Determination of Sound Absorption Coefficient and Impedance in Impedance Tubes — Part 1: Method Using Standing Wave Ratio.
2. Standard ISO 354:1985, Acoustics — Measurement of Sound Absorption in a Reverberation Room.
3. Fahy, F.J., *Sound Intensity*, 2nd Edition, E & FN SPON, 1995.
4. Chung, J.Y. and Blaser, D.A., "Transfer Function Method of Measuring the In-Duct Acoustic Properties. Part I: Theory, and Part II: Experiment" *J. Acoust. Soc. Am.*, Vol. 68, No. 3, 1980.
5. Kuttruff, H., "Sound Field Prediction in Rooms," in *Proc. 15th Int. Congr. Acoust*, Trondheim, Norway, 1995 June, vol. 2, pp. 545-552.
6. MATLAB, Signal Processing Toolbox, MathWorks Inc.
7. Naylor, G. and Rindel, J., *Odeon Room Acoustics Program, Version 2.5, User Manual*, Technical Univ. of Denmark, Acoustics Laboratory, Publication 49, Denmark, 1994.
8. Lahti, T. and Möller, H., "The Sigyn Hall, Turku-A Concert Hall of Glass," in *Proc. 1996 Nordic Acoustical Meeting*, Helsinki, Finland, 1996 June, pp. 43-48.
9. Bass, H., and Bauer, H.-J., "Atmospheric Absorption of Sound: Analytical Expressions," *J. Acoust. Soc. Am.*, Vol. 52, No. 3, 1972, pp. 821-825.
10. Kuttruff, H., *Room Acoustics*, 3rd Edition, Elsevier Science Publishers, Essex, England, 1991.
11. Botteldooren, D., "Finite-difference time-domain simulation of low-frequency room acoustic problems," *J. Acoust. Soc. Am.*, Vol. 98, No. 6, 1995, pp. 3302- 3308.
12. Savioja, L., Järvinen, A., Melkas, K., and Saarinen, K., "Determination of the Low Frequency Behavior of an IEC Listening Room," In *Proc. Nordic Acoustical Meeting (NAM'96)*, Helsinki, Finland, June 12-14, 1996, pp. 55-58.
13. Savioja, L., Rinne, T., and Takala, T., "Simulation of room acoustics with a 3-D finite difference mesh," in *Proc. 1994 Int. Computer Music Conf.*, Århus, Denmark, Sept. 12-17, 1994, pp. 463-466.
14. Van Duyne, S. and Smith, J. O., "The 2-D digital waveguide mesh," in *Proc. 1993 IEEE Workshop on Applications of Sig nal Processing to Audio and Acoustics*, New Paltz, NY, Oct. 17-20, 1993.
15. Savioja, L., Karjalainen, M., and Takala, T., "DSP Formulation of a Finite Difference Method for Room Acoustics Simulation.," In *NORSIG'96 IEEE Nordic Signal Processing Symposium*, Espoo, Finland, Sept. 24-27, 1996, pp. 455-458.
16. Savioja, L., and Välimäki, V., "Improved Discrete-Time Modeling of Multi-Dimensional Wave Propagation Using the Interpolated Digital Waveguide Mesh ," In *Proc. ICASSP'97*, München, April 19-24, 1997, Vol. 1, pp. 459-462.
17. Takala, T., Hänninen, R., Välimäki, V., Savioja, L., Huopaniemi, J., Huotilainen, T., Karjalainen, M. 1996. "An integrated system for virtual audio reality," in *Proc. 100th Audio Engineering Society (AES) Convention*, Preprint no. 4229, Copenhagen, Denmark, May 11-14, 1996.
18. Allen, J., Berkley, D., "Image Method for Efficiently Simulating Small-Room Acoustics," *J. Acoust. Soc. Am.*, Vol. 65, No. 4, 1979, pp. 943-950.
19. Borish, J., "Extension of the Image Model to Arbitrary Polyhedra," *J. Acoust. Soc. Am.*, Vol. 75, No. 6, 1984, pp. 1827-1836.
20. Jot, J.-M., Larcher, V., and Warusfel, O., "Digital Signal Processing Issues in the Context of Binaural and Transaural Stereophony," in *Proc. 98th Audio Engineering Society (AES) Convention*, Preprint no. 3980, Paris, France, February 25-28, 1995.
21. Dalenbäck, B.-I., *A New Model for Room Acoustic Prediction and Auralization*, Dr. Phil. Thesis, Chalmers Univ. of Tech., Göteborg, 1995.