DIGITAL WAVEGUIDE MESH MODELLING OF ROOM ACOUSTICS: SURROUND-SOUND, BOUNDARIES AND PLUGIN IMPLEMENTATION

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ABSTRACT

Digital waveguide mesh models have provided an accurate and efficient method of modelling the properties of many resonant structures, including acoustic spaces. 2-D rectilinear and triangular mesh structures have been used extensively in the past to model plates and membranes and as potential analogues to 2-D acoustic spaces. This paper looks at current developments relating to this technique and attempts to highlight potential ways forward for this research. This includes an investigation into the potential suitability of this technique for surround-sound applications and a realtime implementation as a VST plugin.

1. INTRODUCTION

The acoustic properties of a real enclosed space can be uniquely defined by measuring the Room Impulse Response (RIR) at a specific listening point for an input signal applied at a given sound source location. Two methods traditionally used for modelling a RIR based on a description of the room geometry, are ray tracing [1] and the image source method [2]. Hybrid or enhanced models based on these techniques also exist [3]. These methods have individual limitations although common to both is the fact that they are only valid for high frequencies. At low frequencies, or for small, enclosed spaces, where the wave-like behaviour of sound and the effects of room resonances or modes are more noticeable these methods are less appropriate. Digital waveguide mesh models have provided an accurate and efficient method of modelling this physically complex system [4], [5], [6], [7], and it is possible to obtain a RIR as the output from such models.

This simulation of the acoustics of a room with particular emphasis on its reverberant characteristics is a fundamental tool in the field of creative audio processing. The additional demand for surround-sound applications and their associated recording formats has resulted in the requirement for more complex reverberation algorithm design. It is possible to use generic reverberation algorithms adapted for multi-channel use but the accurate modelling of the characteristics of a particular space or environment requires a more detailed approach to which digital waveguide mesh structures provide a potential solution. Using this modelling technique it is possible to measure the response to a given input over a range of points to obtain a multi-channel surround-sound encoded RIR. These surround-sound RIRs can then be convolved with any appropriate audio signal, essentially placing the sound source within the modelled, virtual surround space.

This paper examines results obtained from surround-sound RIR measurements and suggests how these 2-D models could be enhanced to improve the quality of the reverberation effect. Of particular note is the observation of the wave propagation characteristics through the mesh structure at a boundary. Further work discusses the implementation of simple mesh structures in real time using a popular Digital Audio Workstation plugin format.

2. 2-D DIGITAL WAVEGUIDE MESH STRUCTURES

A waveguide is any medium in which wave motion can be characterised by the one-dimensional wave equation. In the lossless case, all solutions can be expressed in terms of left-going and right-going travelling waves and can be simulated using a bi-directional digital delay line. A digital waveguide model is obtained by sampling, both in space and time, the one-directional travelling waves occuring in a system of ideal lossless waveguides [8]. The sampling points in this case are called scattering junctions, and are connected by bi-directional unit-delay digital waveguides [9]. Figure 1 shows the general case of a scattering junction J with N neighbours, i = 1,2,…N.

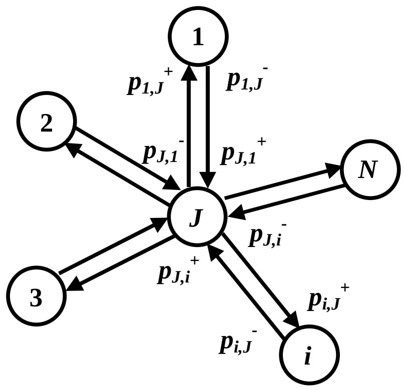

Figure 1: *A general scattering junction J with N connected waveguides for i = 1,2,…,N.*

The sound pressure in a waveguide is represented by p_i, the volume velocity by v_i and the impedance of the waveguide by Z_i. The input to a waveguide is termed p_i^+ and the output p_i^-. The signal $p_{i,J}^+$ therefore represents the incoming signal to junction i

along the waveguide from the opposite junction J. Similarly, the signal $p_{i,J}^-$ represents the outgoing signal from junction i along the waveguide to the opposite junction J. The volume velocity v_i is equal to pressure, p_i, divided by impedance, Z_i. The delay elements are bi-directional and so the sound pressure is defined as the sum of its input and output:

$$p_i = p_i^+ + p_i^- \quad (1)$$

At a lossless scattering junction with N connected waveguides the following conditions must hold:

1. The sum of the input volume velocities, v^+, equals the sum of the output volume velocites, v^-:

$$\sum_{i=1}^{N} v_i^+ = \sum_{i=1}^{N} v_i^- \quad (2)$$

2. The sound pressures in all crossing waveguides are equal at the junction:

$$p_1 = p_2 = \ldots = p_i = \ldots = p_N \quad (3)$$

Using these conditions the sound pressure at a scattering junction can be expressed as:

$$p_J = \frac{2\sum_{i=1}^{N} \frac{p_i^+}{Z_i}}{\sum_{i=1}^{N} \frac{1}{Z_i}} \quad (4)$$

As the waveguides are equivalent to bi-directional unit-delay lines, the input to a scattering junction is equal to the output from a neighbouring junction into the connecting waveguide at the previous time step. This can be expressed as:

$$p_{J,i}^+ = z^{-1} p_{i,J}^- \quad (5)$$

To model the propagation of a wave on the horizontal plane within an enclosed space, 2-D rectilinear and triangular mesh structures can be constructed using unit delay waveguides and lossless scattering junctions with $N = 4$ and $N = 6$ in Equation 4 respectively. A signal representing acoustic pressure introduced to a waveguide will propagate in either direction along the bi-directional delay lines until it comes to a junction. The signal then scatters according to the relative impedances of the connected waveguides. In the current model all impedances are set to be equal.

3. SURROUND-SOUND REVERBERATION RIRS

It is generally given that for a high quality surround reverberation effect, with a sense of depth and envelopment, the reverberant signals at each output should be fully decorrelated [10]. A rectangular room has been modelled using the triangular mesh, 8.0m long and 5.0m wide. The sound source input is situated in the top left corner, 1.0m from each wall. A 5.1 surround-sound RIR is measured around a point situated 6.0m from the top wall and 2.5m from the left wall as shown in Figure 2. Each of the 5.1 RIR measurements has an omnidirectional characteristic with each measuring position being located on the circumference of a circle 1.0m radius from the centre point.

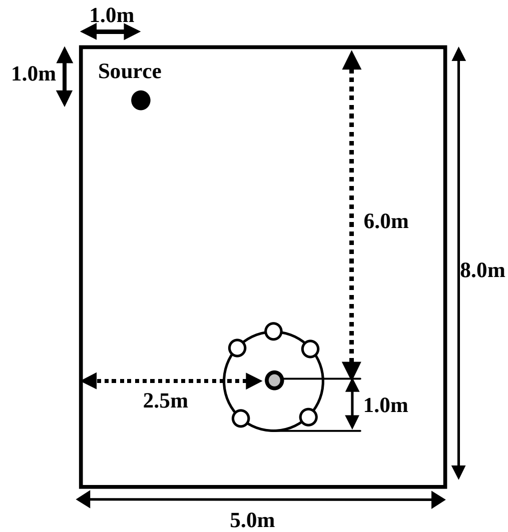

Figure 2: *Plan view of the modelled room showing positions of input sound source and 5.1 RIR measurement points based around a central output position.*

Despite the fact that the geometrical layout of the room is simple and regular, with uniform absorption/reflection conditions along each boundary, there is evidence of significant decorrelation in the measured RIRs. Figure 3(a) shows the normalised autocorrelation function for the Centre RIR. Figure 3(b) shows the normalised cross-correlation between the Centre and Right Surround RIRs, which is typical for the cross-correlation of any pair of RIRs obtained. Clearly this is an encouraging result showing that this technique is appropriate for surround-sound reverberation modelling. By way of comparison Figure 3(c) shows the normalised cross-correlation between the left and right channels of a generic "good quality" filter based reverberation algorithm, showing that the correlation between outputs in this case is even less.

It is conjectured that the surround output RIRs can be decorrelated to a greater extent by improving the quality and complexity of the modelled room. Possibilities include the more accurate modelling of behaviour at the boundaries including frequency dependence and the effect of diffuse reflections, and using a more complex room geometry.

4. BEHAVIOUR AT BOUNDARIES

The most convenient and straightforward way to consider what happens at the boundary of a mesh structure is to set a boundary junction as having only one other neighbour. The effect of a boundary in a real room is to produce a reflection of an incident sound wave, usually with some frequency dependent absorption of the wave energy at the boundary itself. In a digital waveguide structure a reflection is caused by a change in the impedance of

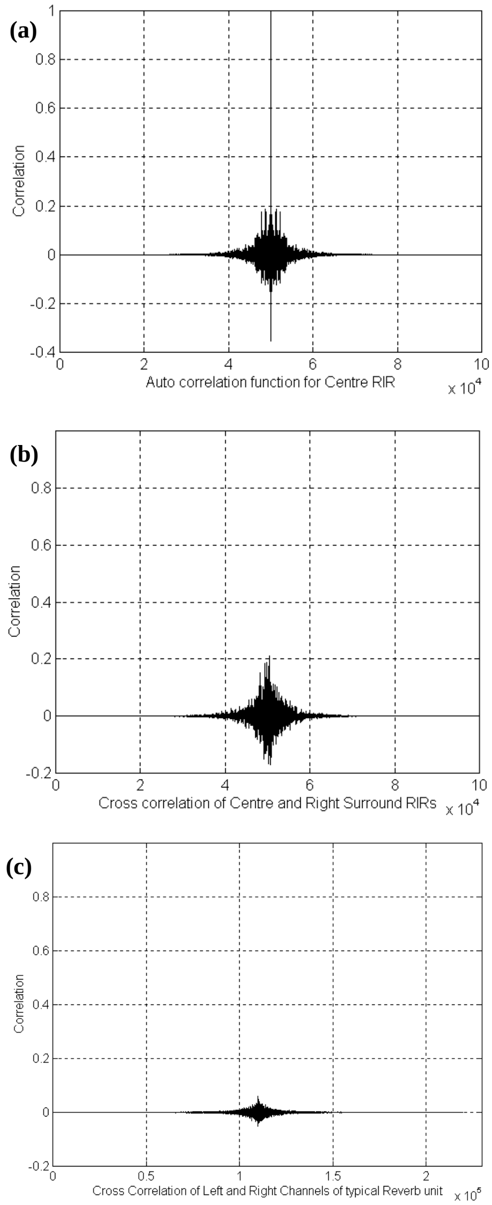

Figure 3: *Cross-correlation functions for surround output RIRs. (a) The normalised autocorrelation function for the Centre RIR; (b) the normalised cross-correlation function for the Centre and Right Surround RIRs; and (c) for comparison the normalised cross-correlation function for the left and right channels of a typical filter based reverberator.*

the waveguide. This can be conceptualised by connecting a dummy junction on the other side of the boundary junction, essentially within the boundary itself, as in Figure 4. The connecting waveguides on either side of the boundary will have different characteristic impedances, Z_1 and Z_2 respectively. If at a boundary the impedance changes from Z_1 to Z_2 the reflection coefficient r is defined as:

$$r = \frac{Z_2 - Z_1}{Z_2 + Z_1} \qquad (6)$$

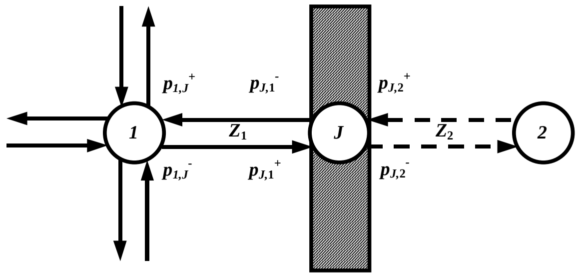

Figure 4: *Termination of a waveguide mesh due to a boundary resulting in a reflection.*

Given that there is no contribution into the boundary junction, J, from the dummy junction 2 and using (4) and (5), the sound pressure for the boundary junction can be calculated as a function of the sound pressures of the incident travelling waves, p_i^+ giving:

$$p_J = (1 + r) \cdot p_{J,1}^+ \qquad (7)$$

In general, simple absorption can be modelled at such a boundary by replacing the junction adjacent to the boundary itself with the equivalent n-port junction (where n = 1,2,…,5), according to the room/boundary geometry that the mesh model has to fit. The amount of energy reflected at the boundary is determined by setting r equal to a value between 0 and 1.

Although these boundary conditions are satisfactory for most situations they are clearly not ideal. The direction dependent characteristics of the mesh structure due to its topology have an influence on any wave propagation through it, and this property is equally applicable at any boundary as implemented above. Figure 5 compares the frequency response of incident and reflected waves at a boundary with r = +1, that is, total reflection with no absorption, on the triangular mesh. In this case the reflected wave has been measured at an angle of 30^o to the incident direct sound and the input is a gaussian impulse applied over 6 time steps.

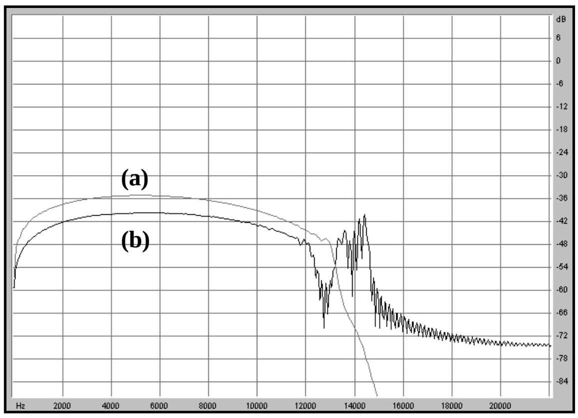

Figure 5: *Frequency response of incident and reflected waves at a boundary on the triangular mesh with r = 1.0. (a) Incident wave; (b) Reflected wave.*

Although some of the high frequency distortion can be attributed to interference effects due to the influence of the direct sound, it should clearly not be as significant as displayed, implying that this is due to the actual reflection characteristic itself. Further, Figure 6 shows the result of a reflection of an impulse from four boundaries, all with r = 0.0, being theoretically anechoic conditions. Despite approximating anechoic conditions, a significant amount of energy is reflected back into the room.

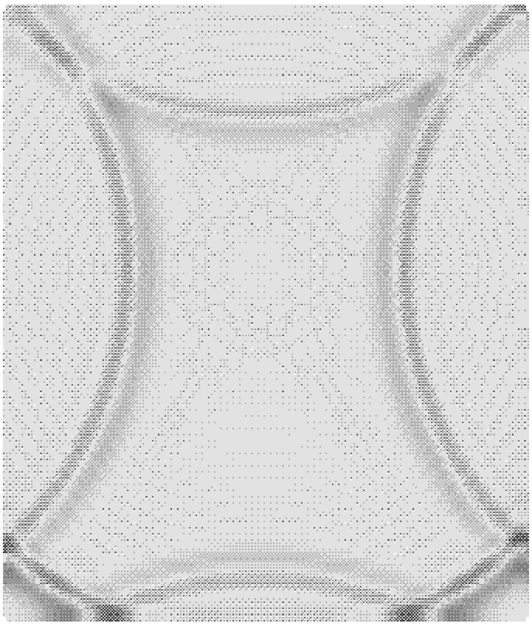

Figure 6: *2-D Rectangular room modelled using the triangular mesh with all boundaries set with r = 0, simulating anechoic conditions.*

Clearly the boundary conditions as currently implemented are an oversimplification of reality even though they are consistent with the mesh construction, and offer a satisfactory solution enabling reflection and absorption to be modelled. By contrast, real acoustic boundaries are both frequency and direction dependent and this has not been considered in the derivation of the boundary conditions. High frequency reflections at anechoic boundaries have been noted in previous waveguide mesh studies [4].

The Transmission Line Matrix (TLM) method [11] is a discretized time domain model generally used for investigating electromagnetic wave propagation, being equivalant to the rectilinear waveguide mesh model. It has also been applied to other disciplines including general acoustics [12]. TLM uses a resistive matched termination to the transmission line segments at the edge of the mesh giving a first order approximation to a free space boundary, effectively the same method as used in the waveguide mesh model. However, TLM results show that waves striking the boundary at non-normal angles of incidence will not see a matched termination, resulting in some reflection. The problem is complicated further by discretized or “stepped” boundaries that are not parallel to the mesh structure yet are attempting to model an ideally continuous and smooth boundary. The ends of each “step”, no matter how high a mesh sampling rate is used, will act to scatter and reflect high frequency waves. These comments seem to be in agreement with observed results observed in 2-D triangular mesh structures.

There are a number of possible improvements that can be implemented to model the boundary of a waveguide mesh more accurately. For anechoic conditions, TLM modelling techniques suggest recalculating the required impedance every few time steps for each point on the boundary in order to take account of the angle of arrival for the incident waves [12]. More generally, the impedance based boundary conditions can be replaced with a digital filter to model the frequency dependent absorption characteristics of real materials. This has been attempted on a 2-D rectilinear mesh [13], with some success although the directional dependent nature of wave propagation on this topology, again being equally applicable to the boundary reflection characteristics, gives results that vary with angle of incidence. It is also possible to model more diffuse reflections at the boundaries, scattering wave energy in every direction regardless of the angle of incidence. This has been attempted by pre-varying the angle of incidence of a wave in a random fashion over time at the boundary junctions on the mesh using circulant matrices. This may prove to be an even more accurate model, particularly as the amount of diffusion can be varied and frequency dependence can also be implemented as an extension to this technique. This method has been applied to 2-D drum membranes based on a triangular mesh topology [14]. Further possible solutions can be sought from the wealth of information available relating to boundary problems in finite difference time domain techniques, again a method that is effectively equivalent to the waveguide mesh model [8].

5. REALTIME WAVEGUIDE MESH PROCESSING

Part of our current research relates to an investigation into the feasibility of using waveguide mesh models for real-time sound synthesis and processing. To this end the Steinberg VST Software Development Kit [15] has been used to develop two audio processing "plugins". A VST plugin is an audio process that takes a stream of samples from a host application, processes them according to the algorithm implemented within the plugin and returns the result back to the VST host. A plugin is not an application in its own right, and from the host application's perspective is a black box with an arbitrary number of audio inputs, outputs and associated parameters. The VST format has become something of standard, being commonly used in digital audio editing, processing and recording applications across both Mac and PC platforms, providing processing and synthesis options for all levels of demand from freeware to professional post production solutions. Dedicated VST DSP hardware acceleration solutions are now also becoming more commonly available.

Two plugins have been developed, a VST instrument that can be triggered via MIDI that is designed to simulate a very simple membrane using the triangular waveguide mesh and a VST resonator that uses a similar mesh to process an audio signal. Figure 7 shows the VST waveguide mesh instrument, and the parameters that the user can control to alter the timbre of the output sound, including input and output position, amount of reflection at the boundaries and in the case of the instrument only, both the level and width of the excitation pulse.

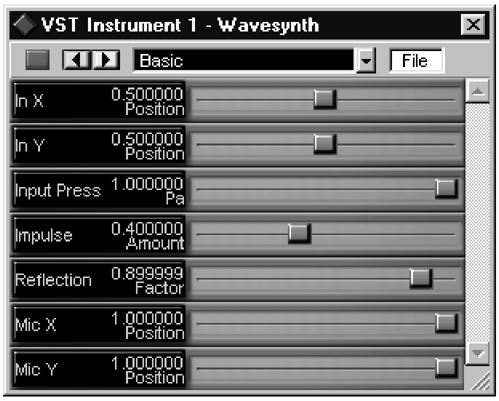

Figure 7: *The Wavesynth VST waveguide mesh instrument.*

Although promising and clearly demonstrating the potential for such modelling synths/processors, the CPU performance required to get even small 2-D membranes working at audio sampling rates is unsurprisingly rather high, given the nature of the modelling technique. For instance, simple benchmarking revealed that a 6x6 mesh uses all of the performance resources of a Pentium III 500Mhz processor, with a more useful 10x10 mesh being achievable on an Athlon 1.3GHz based system. Clearly if the considerable potential of this modelling technique is to be made available to the wider community, research is required to optimise the modelling algorithm to remove some of the redundancy inherent in repeating the same simple calculations over a such large data structure. Hardware DSP acceleration, possibly within the VST (or an alternative) protocol, is another possible solution.

6. CONCLUSIONS

Digital waveguide mesh models have been applied to the creation of RIRs for convolution based surround-sound reverberation effects. Results show that this is a promising technique, incorporating the property of naturally decorrelated reverberation when multi-channel output RIRs are obtained. Work is ongoing to improve the modelling technique for better quality reverberation, with some study being made of the properties of the mesh at a boundary. This will ultimately lead to the accurate implementation of materials based, frequency dependent boundary conditions. Real-time mesh structure audio processing plugins have demonstrated that they are potentially a very promising method of sound synthesis and audio processing. However, for more interesting and useful sound output, dependent upon the implementation of larger mesh structures, improvements in model execution time are clearly required.

7. REFERENCES

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