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the half of the delay of the fastest functional unit to avoid generating too many extra states.

The performance of the proposed method is compared with the conventional minimum state implementation [3] in Table 1. We can obtain a latency reduction of 29.4% on average through the proposed latency compaction technique.

Conclusion: We propose in this Letter a latency compaction technique which reduces the latency by optimising the system clock period. The proposed method allows non-integer multi-cycling and chaining of operators, resulting in an average performance improvement of 29.4%.

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Modelling and directionally encoding the acoustics of a room

D.T. Murphy and D.M. Howard

Geometrical methods are often used to model the acoustic properties of a room, but are valid only for high frequencies. At low frequencies, diffraction and the effects of room modes cannot be neglected. A method for modelling the two-dimensional propagation of sound within an enclosed room is presented which encompasses both of these particular properties by making use of a digital waveguide model.

Introduction: The processing of audio signals by simulating the acoustics of a room or hall is a widely used technique to enhance recorded and synthesised sounds in the computer music and recording industries. The acoustic properties of a real enclosed space can be uniquely defined by measuring the room impulse response (RIR) at a specific listener location for an input signal applied at a given sound source location. Based on a description of the room geometry, it is possible to model an RIR using either ray tracing [1] or the image source method [2]. Both of 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 modes are more noticeable, these methods are less appropriate.

This problem has been addressed in a number of ways, usually based around a direct time domain model of wave motion within the space, but these methods are often computationally intensive. Digital waveguide models have been used successfully as a partial alternative to time domain models based on the wave equation in the fields of musical acoustics and sound synthesis. They have also been used to model the low frequency behaviour of an enclosed listening space [3]. Digital waveguide models are related both to finite difference time domain (FDTD) techniques [4] and to transmission line models (TLM) that are commonly used in problems involving electromagnetic compatibility [5]. Current digital waveguide models of room acoustics are limited to providing only a partial solution to an accurate RIR, as they are less effective at high frequencies where execution time becomes prohibitive for a full 3D implementation.

This Letter describes the ongoing implementation of a digital waveguide model intended to provide an RIR valid for both low and high frequencies by limiting the model to the horizontal plane only. Results are presented that validate the current model for low frequencies, and demonstrate how audio can be processed using multi-channel RIRs for presentation to a listener in a multi-speaker surround- sound environment.

2D digital waveguide mesh: 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 simple bidirectional digital delay line. A digital waveguide model is obtained by sampling, both in space and time, the one directional travelling waves which occur in a system of ideal lossless waveguides. The sampling points in this case are called scattering junctions, and are connected by bidirectional unit-delay digital waveguides.

To model the propagation of a wave on the horizontal plane within an enclosed space, a 2D rectilinear mesh is constructed using unit delay waveguides and lossless scattering junctions. Each junction will have four similar neighbours, each connected by a unit waveguide. Boundary junctions only have one neighbour. A signal representing acoustic pressure introduced to a waveguide will propagate in either direction along the bidirectional 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. At a boundary, a wave can be reflected with or without phase inversion, and simple absorption can also be modelled.

Spatial encoding of RIRs: It is possible to encode a soundfield by decomposing it into spherical harmonic components. The zero-order pressure component is termed W and is omnidirectional, picking up all sounds from all directions equally. The first-order velocity components are figure-of-eight responses pointing forward, left and up, and are termed X, Y and Z, respectively. The four signals, W, X, Y and Z are known as B-format. Reproducing a B-format signal is possible using an appropriate ambisonic decoding scheme and a multi-speaker array. It is possible to derive horizontal only B-format signals - W, X and Y only - from the mesh, resulting in a B-format multi-channel RIR.

Using a simple convolution scheme, anechoic or synthesised sound is processed with a B-format RIR for playback over a multi-speaker array. Further processing using a set of measured head related transfer functions (HRTFs) allows binaural surround sound monitoring over headphones, removing the necessity for a large multi-speaker array.

Implementation: The sampling rate of the mesh is determined by the spatial resolution and topology of the mesh. With the 2D rectilinear mesh, the update frequency is given by $f_{update} = c\sqrt{2}/d$, where c is the speed of sound, and d is the distance between scattering junctions. To generate an audio rate RIR, a mesh sampling rate of at least 44.1kHz is required. This can be obtained assuming c = 343m/s and d = 0.011m giving f_{update} = 44098Hz. However, the effective sampling rate is well below the value given by f_{update} due to the dispersive nature of the mesh's lattice type structure [6].

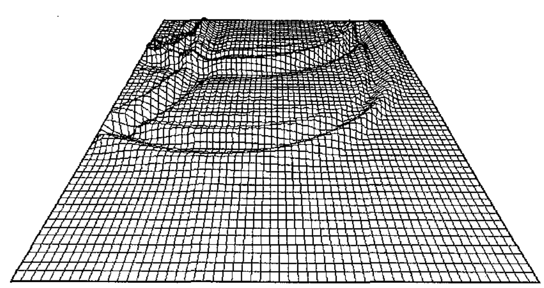

Fig. 1 *Wave propagation along mesh with phase reversing reflections at walls*

Two versions of the waveguide mesh have been implemented for a rectangular room of length 6.6m and width 5.5m. The first is for use at low sampling rates. This enables the user to observe wave propagation on a graphical display with a number of different views and realisations. RIR measurements can be made for various source-listener positions and examined in both the time and frequency domains, and user control is given over variables such as room width, length and mesh density. The second implementation runs without graphical feedback or user interaction, allowing much higher sampling rates to be used.

Results: Fig. 1 shows a wave propagating along the mesh. Note the phase reversed reflections from the walls.

Fig. 2 shows a wave propagating along the mesh and passing through gaps in a dividing wall. Notice how the waves spread out into the second half of the room showing that the model is capable of demonstrating diffraction effects.

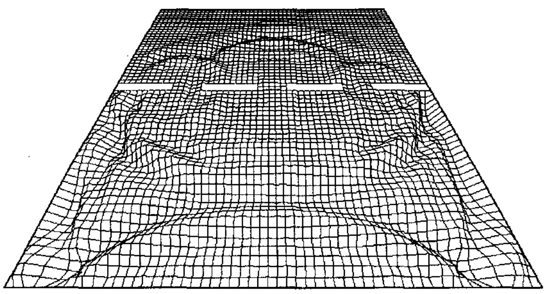

Fig. 2 *Diffraction effects due to gaps in a wall placed in modelled room*

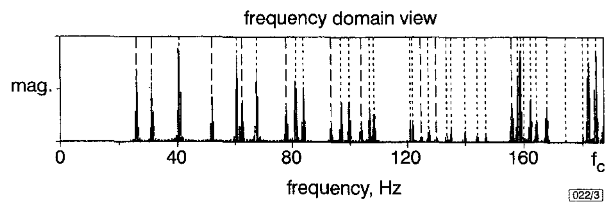

Fig. 3 *Frequency response of modelled room below approximate critical frequency F_c*

— — analytical axial modes
– – – analytical tangential modes

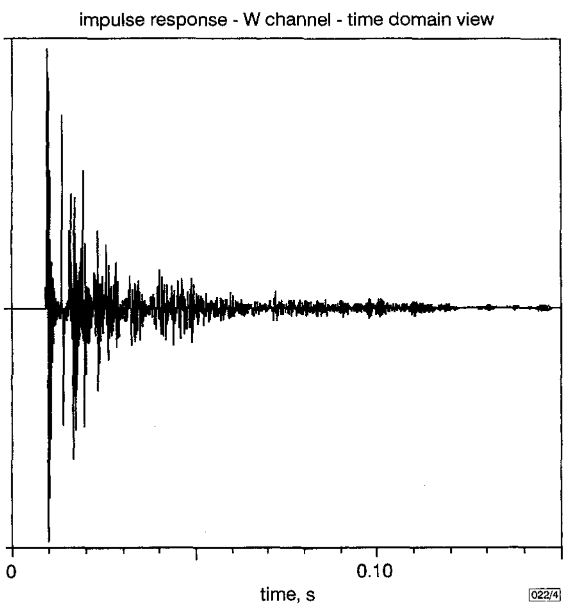

Fig. 4 *W channel RIR from horizontal B-format RIR measurement*

Fig. 3 shows a measured RIR in the frequency domain up to the approximate critical frequency of the room, below which modal behaviour is dominant. A smooth impulse was applied at one corner of the room. Boundary conditions were set to give a phase preserving reflection, so allowing a pressure maximum at the walls, with no absorption. A 2s RIR was measured at the opposite corner of the room with a mesh sampling rate of 44.1kHz. The RIR has the first 0.2s removed to eliminate any transient response, and the remaining 1.8s has an FFT applied to it using a Kaiser-Bessel window.

For comparison, the analytical room modes have been calculated [7] and plotted on the same graph. In this case, only axial (reflections between two surfaces) and tangential (reflections between four surfaces) are valid.

The Figure shows that in this low frequency region there is an exact correlation between the analytical room modes as calculated, and the resonant frequencies highlighted by examining the frequency response of the measured RIR.

Fig. 4 shows the W channel RIR from a horizontal B-format measurement. In this example the walls are phase preserving and are slightly absorptive.

Note that the impulse response is characteristic of a typical real RIR. There is an initial time delay before the sound reaches the listener, a single large impulse representing the *direct sound*, followed by a number of clearly discrete impulses of similar size or lower amplitude generated by *early reflections* from the walls in the room. Finally, there is the slowly decaying *reverberant sound* generated by the many overlapping and delayed reflections from the walls.

Conclusions: This Letter has presented a method of modelling acoustic wave propagation in an enclosed space using a 2D digital waveguide mesh. Initial results show that diffraction effects are clearly visible and that the modal behaviour of a theoretical 2D room is closely approximated at low frequencies. Audio samples, when convolved with a B-format RIR obtained from the mesh, can be presented to a listener over a multi-speaker array. The effect of the B-format RIR is to add a reverberant quality to the anechoic or synthesised audio complete with appropriate audio cues for sound source localisation, as if the sound had originated from within the modelled space.

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10GHz range extremely low-loss surface acoustic wave filter

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A low-loss 10GHz range surface acoustic wave (SAW) filter is investigated. It is fabricated on 128° Y-X $LiNbO_3$ using electron beam direct writing and a lift-off process (aluminium electrode width = 95nm, thickness = 30nm). Experimental results of the ladder type SAW filter show low loss characteristics. A minimum insertion loss of 3.4dB in the 10GHz range is obtained.

Various communication and signal processing systems, such as mobile communication systems, require high frequency devices, and it is expected that frequencies of ~10GHz will be attained in the near future. Surface acoustic wave (SAW) devices are now in the area of 2GHz, and 10GHz range devices using submicron