HYBRID ROOM IMPULSE RESPONSE SYNTHESIS IN DIGITAL WAVEGUIDE MESH BASED ROOM ACOUSTICS SIMULATION

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ABSTRACT

The digital waveguide mesh (DWM) and related finite difference time domain techniques offer significant promise for room acoustics simulation problems. However high resolution 3-D DWMs of large spaces remain beyond the capabilities of current desktop based computers, due to prohibitively long run-times and large memory requirements. This paper examines how hybrid room impulse response synthesis might be used to better enable virtual environment simulation through the use of otherwise computationally expensive DWM models. This is facilitated through the introduction of the *RenderAIR* virtual environment simulation system and comparison with both real-world measurements and more established modelling techniques. Results demonstrate good performance against acoustic benchmarks and significant computational savings when a 2-D DWM is used as part of an appropriate hybridization strategy.

1. INTRODUCTION

The Digital Waveguide Mesh (DWM) [1, 2] is a discrete-time numerical simulation method used to model acoustic wave propagation in an enclosed system. First applied to the problem of reverberation simulation and then extensively to physical modelling sound synthesis [3], DWMs have also been shown to be appropriate for virtual acoustic applications through the generation of Room Impulse Responses (RIRs) suitable for auralization purposes [4, 5]. The DWM is especially successful at accurate room acoustics simulation at low frequencies [6], hence providing both an alternative and complementary approach to more traditional geometric acoustics techniques.

DWMs are a subset of the wider family of finite difference time domain (FDTD) numerical approximation schemes that have also been more generally applied to acoustic environment simulation problems [e.g. 7]. Although their origin lies in 1-D digital waveguide based sound synthesis, research to date has been heavily influenced by traditional FDTD methods and techniques [e.g. 1, 8], and continues to be, especially through the development of mixed modelling methods and boundary termination [9, 10, 11]. The DWM potentially offers results that are equally valid to more established techniques such as FEM/BEM/Geometric Acoustics, together with greater flexibility in terms of implementation and realization. For instance, although computationally expensive for large spaces, wave propagation effects, such as diffraction, are an inherent part of the implementation [5], requiring no additional processing load. Additionally, although RIR generation must take place offline, an arbitrary input signal may be processed in real-time using an appropriate convolution scheme, making this approach a realistic proposition for computer music or audio post-production applications requiring accurate synthesis of a virtual environment.

Since the DWM was first applied to the problem of room acoustics simulation by Savioja et al. in 1994 [4], related research has been focused in a number of areas, including 2-D and 3-D implementations, minimisation of dispersion error, boundary termination [9, 10, 11], diffusion modelling [12] and spatial encoding [13], much of which is also discussed in [2]. However despite significant inroads in these areas, the problem still remains that high resolution DWMs required for high sample rate audio bandwidth RIRs take a prohibitively long time to synthesize. Although this will be ameliorated somewhat as desktop computing power continues to increase, with multi-core CPUs now allowing the use of parallelization techniques [14], and even graphics cards being utilized to provide hardware acceleration, large spaces will remain offline only for some time. Hence an alternative approach is required to make these techniques more readily available at the desktop.

This paper examines the use of RIR hybridisation techniques within the framework of DWM virtual environment simulation. A hybrid RIR synthesizes the required acoustic impulse response for a defined source/receiver/space using a combination of multiple room acoustic or reverberation simulation techniques. Such approaches were first proposed in e.g. [6] and have also been explored more recently for DWMs [15] and more generally [16].

This research has been completed within the framework of *RenderAIR*, the next development of the *RoomWeaver* system that was originally proposed for research and development into DWM based room acoustics simulation [17] and this paper is therefore organised as follows. Section 2 briefly covers the background to the DWM and discusses issues relating to mesh implementation. Section 3 gives an overview of *RenderAIR*, discussing aspects of the hybrid modelling engine. Section 4 is a case study where a real-world room is modelled using *RenderAIR* and a geometric acoustics based modelling application, with both compared against actual room impulse response measurements. Finally, Section 5 summarises the work completed to date and indicates future directions for the *RenderAIR* project.

2. THE DIGITAL WAVEGUIDE MESH

The digital waveguide mesh (DWM) was first proposed by Van Duyne and Smith [1] as an extension to 1-D digital waveguide sound synthesis appropriate for modelling plates and membranes, potentially leading to full 3-D object modelling (the reader is referred to [2] and [3] for a thorough treatment and discussion of this area). From the basic 1-D digital waveguide model, higher dimension mesh structures are constructed using bi-directional delay line waveguide elements and scattering junctions which act as a regular grid of spatial and temporal sampling points within the modelled domain. The sound pressure in a waveguide element is represented by p_i, the particle velocity by v_i and the impedance of the waveguide element by Z_i where $p_i/v_i = Z_i$. The input to a waveguide is termed p_i^+ and the output p_i^-. The signal $p_{J,i}^+$ therefore represents the incoming signal to junction J along the waveguide element from the neighbouring junction i. Similarly, the signal $p_{J,i}^-$ represents the outgoing signal from junction J along the waveguide to the neighbouring junction i. The total sound pressure p_J in a waveguide element connected to junction J can be defined as the sum of the travelling waves in this element, or alternatively as the sum of the input and output:

$$p_J = p_{J,i}^{\ +} + p_{J,i}^{\ -} \tag{1}$$

For a lossless junction J the sum of the input velocities is equal to the sum of the output velocities, and the sound pressures in all crossing waveguides are equal, and so the sound pressure p_J at junction J for N connected waveguides can be expressed as:

$$p_J = \frac{2\sum_{i=1}^{N} \cdot \frac{p_{J,i}^{\ +}}{Z_i}}{\sum_{i=1}^{N} \frac{1}{Z_i}} \tag{2}$$

As DWM waveguide elements are equivalent to bi-directional unit-delay lines, the input to scattering junction J at time index n, $p_{J,i}^+(n)$, is equal to the output from neighbouring junction i into the connecting waveguide at the previous time step, $p_{i,J}^-(n\text{-}1)$. Expressing this relationship in the z-domain gives:

$$p_{J,i}^{\ +} = z^{-1} \cdot p_{i,J}^{\ -} \tag{3}$$

From (2) junction pressure values are calculated according to input values from immediate neighbours, output values are calculated using (1) and then propagated to neighbours via the bi-directional waveguide elements, becoming inputs at the next iteration according to (3). From (1), (2) and (3), via an appropriate linear transformation, it is possible to derive an equivalent formulation in terms of junction pressure values only:

$$p_J = \frac{2\sum_{i=1}^{N} \frac{p_i}{Z_i} \cdot z^{-1}}{\sum_{i=1}^{N} \frac{1}{Z_i}} - p_J \cdot z^{-2} \tag{4}$$

This expression can also be derived directly from a finite difference time domain (FDTD) formulation of the wave equation [1]. A digital waveguide model generally refers to a representation of acoustic signal propagation via two directional wave components and schemes implemented in this way are termed W-models or W-DWMs. This alternative implementation as a Kirchhoff variable DWM (K-DWM), depends on physical quantities only rather than sampled travelling-waves and is generally equivalent to a FDTD simulation. Mixed modelling where K-DWM and W-DWM approaches have been interfaced combines the computational efficiency of the K-DWM approach in terms of computation time and memory use, with the flexibility of scattering-based boundary termination options for complex geometries through the use of KW-pipe transfer functions, see e.g. [2, 11].

Dispersion error in a DWM, where the velocity of a propagating wave is dependent upon both its frequency and direction of travel, leads to wave propagation errors and a mistuning of the expected resonant modes. Dispersion error is dependent upon mesh topology but its effects, particularly in terms of direction dependence, can be minimised using one of a number of methods including topology implementation, mesh interpolation, frequency warping, or mesh oversampling. In FDTD (K-DWM) modelling, dispersion can also been reduced by using a compact implicit scheme [18]. Note that although accurate synthesis of low frequency modes is required in a room acoustics model, dispersion error is considered less important with increasing frequency as modal density increases, and perception of such variations becomes less critical. The sampling rate f_{update} for a DWM of dimension, D, and spatial sampling distance d is determined by the Courant condition such that:

$$f_{update} = \left(c\sqrt{D}\right)/d \tag{5}$$

where c is the speed of sound. Ultimately f_{update} dictates the quality of RIR output from a DWM, with large sample rates requiring denser meshes, more computer memory and hence taking longer to run, limiting even the most efficient large-scale K-DWMs to offline generation only.

Finally, consideration must also be made as to the most appropriate method for terminating a DWM as part of a room acoustics simulation. In an arbitrary enclosed space it is typical with other geometrical acoustic models to approximate real-world boundary conditions through the use of standardized parameters such as octave-band absorption coefficients and diffusion/scattering coefficients. Rather than attempting to implement a physically derived DWM based solution that more directly models a particular boundary material (with each material similarly requiring an individually modelled solution), it is usually sufficient and appropriate, for the purposes of RIR sound quality, to implement a boundary termination such that its global behaviour approximates these given parameters. Solutions researched and implemented to date are presented in [2, 9, 10, 11].

3. THE RENDERAIR SYSTEM

3.1. Overview

RenderAIR, as shown in Figure 1, is a cross-platform DWM room acoustics modelling application, implemented in the style of an Integrated Development Environment (IDE) with a hierarchical data and file structure. A *RenderAIR* project contains the specification for the dimensions and contents of a particular virtual space, defined as a *geometry*. For each geometry different surface sets can be specified, various source and receiver configurations can be set up and a number of DWMs can be defined.

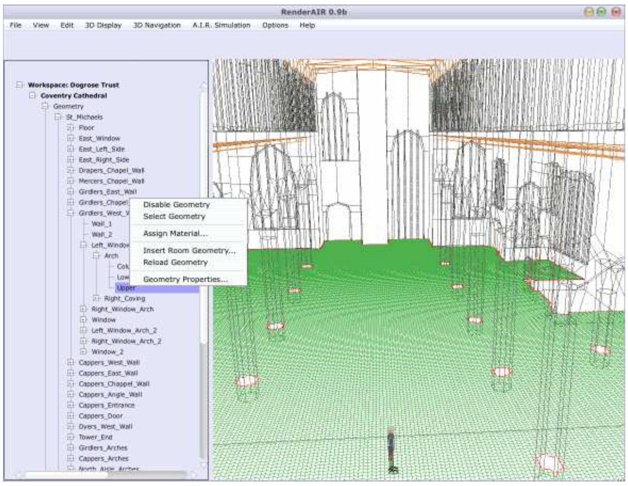

Figure 1: *RenderAIR showing the GUI used for defining and editing properties of a virtual environment.*

3.2. Defining the Space

The size and shape of a room and all the objects in it are specified in terms of a number of planar surfaces together with their reflective/absorptive and diffusive properties. *RenderAIR* uses standard Cartesian co-ordinates and so a surface is defined by specifying the locations of its corners relative to the system origin. By combining multiple surfaces into groups, complex shapes can be created and these can then be stored as *Models* and re-used as required. *RenderAIR* files are written in a simple scripting language that allows variable manipulation, loops and conditional statements, enabling sophisticated models to be defined. Aside from script based definition and manipulation, *RenderAIR* geometries can also be import and exported using the COLLADA 3-D graphics file format. This allows access to buildings and objects generated other 3rd party CAD packages including the popular and freely available *Google Sketchup*.

Materials are assigned to surfaces in the geometry file, but once loaded the materials on a boundary surface may be easily changed and saved as *surface sets*, allowing the same geometry to have several different profiles, and the effect of using different building or furnishing materials to be explored. Sources and receivers are placed according to their co-ordinates. Using the scripting language, groups of transducers can be created along with an associated geometry to produce, for example, a spaced stereo pair, a binaural RIR incorporating a dummy head or a higher order ambisonic receiver. For each source the input signal is specified in a wave or text file.

3.3. Meshing a Complex Arbitrary Space

Once a virtual space has been defined the structure is populated with either a 2-D or 3-D DWM and *RenderAIR* uses a plug-in architecture to allow different DWMs to be investigated both in terms of topology and implementation, which may be as a K-, W-, or hybrid KW-DWM. To simulate the space being tested the room geometry must be filled with a uniform spatial grid of DWM nodes. As the sampling grid arrangement of a DWM will vary according to the topology plug-in used and the arbitrary geometry of the defined virtual space, a generalized, flexible approach to filling the space under test is required. A single *air-node* is placed at a user-defined seed-point within the geometry model and this node reproduces itself by sending out "creepers" along its ports into the surrounding empty space. If no boundaries or existing air-nodes are encountered then a new air-node is created at the end of the creeper. These new nodes will then send out their own creepers, and hence the space begins to fill with basic N-port air-nodes. If a creeper discovers a surface then a *boundary-node* is created at the end of the creeper rather than an air-node. These boundary-nodes are incapable of reproduction and hence do not send out any creepers of their own. In this manner the meshing process will continue until the edge of the modelled space, or object geometry boundaries are encountered. When a boundary-node is created it will inherit the properties of the particular surface associated with its geometrical location.

3.4. Simulation

The simulation process starts by meshing out the space as described above, designing optimal boundary filters for the surface material absorption coefficients given, and then applying input source excitation signals at the specified locations. By updating the state of each node according to the DWM algorithm used one sample at a time these signals propagate through the mesh and this process can be visualised as shown in Figure 2. The RIR at each of the receivers is written to a wave file in a unique directory for each simulation run, to prevent previous simulation data from being overwritten. The simulation is terminated either when the required number of samples have been generated or when the signal level falls below a specified value. Finally, *RenderAIR* allows various RIR post-processing options, including input signal compensation, low-pass filtering to simulate air absorption and higher order ambisonic spatial encoding [13].

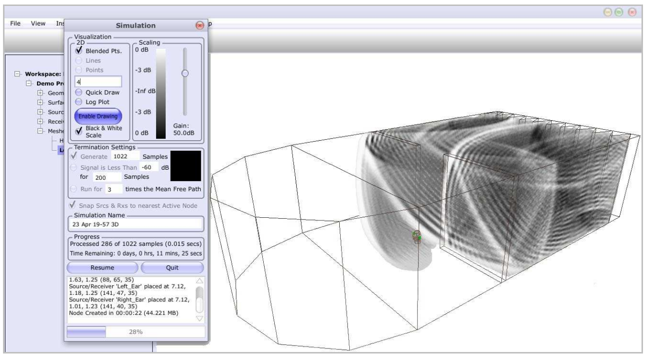

Figure 2: *One of the RenderAIR run-time visualization options demonstrating full 3-D wave propagation using a grayscale plot.*

3.5. Hybrid RIR Synthesis

Full audio bandwidth DWM-based RIR synthesis requires significant, potentially prohibitive, computational resources, especially if an oversampled DWM is used in order to minimize dispersion error. Hence *RenderAIR* adopts a hybrid approach to full room acoustics simulation according to the system resources available and the requirement of the user in terms of final output.

The early-part of a RIR contains much of the perceptually relevant information that gives the listener a sense of the size and shape of the space in which they are placed. Hence this early-part of a RIR should be modelled as accurately as possible for best results. It is also well known that high quality, natural reverberation can be simulated using a more generic approach and that in an ideal space, reverberation is usually not dependent on the exact positioning of source/receiver. Therefore the first hybridisation of the RIR involves an early-part simulation (typically up to 100ms), significantly reducing RIR execution time when compared with a full length high resolution 3-D simulation that will generally be of the order of a few seconds long. The late-part of the total response can then be simulated more generically using another method and in *RenderAIR* this is facilitated through the use of a separate 2-D DWM reverberation model.

A 2-D DWM is by no means a complete or accurate representation of an acoustic space, yet such a simulation will still capture a subset of the important low frequency modal resonances and early reflections in the 2-D plane, giving an acoustic representation that is in part correct. As frequency increases, modal resonances become more densely distributed and individually distinct reflection components become impossible to resolve. With a suitably high sample rate 2-D DWM, such that low frequency modes are accurately captured, and a sufficient density of modal distribution is achieved above the Schroeder frequency for the simulated space this reduction in dimension becomes less critical. High quality reverberation can be achieved despite the modal density increasing only linearly with f_{update} rather than as the square of f_{update} for a similar 3-D simulation [19]. Typically f_{update} is set such that it is at least four times the required bandwidth. Hence *RenderAIR* provides this option for generating RIRs either as an indication as to how a full 3-D space will sound, but in a much reduced timeframe, or as part of a hybrid response. These 2-D results are also appropriate for reverberation processing in their own right, if absolute accuracy in terms of spatial/acoustic perception is not required. This method seems the most appropriate choice as it has been shown to give good results in prior work e.g. [5, 18], takes a fraction of the time to execute over a full 3-D model and makes use of geometrical/virtual environment information already available.

If further computational savings are required, *RenderAIR* allows the sample rate of the ideal 3-D mesh to be reduced, offering very significant additional savings while maintaining the all important accurate low frequency response that is one of the key benefits of this approach [6]. The high frequency component of the total RIR is then modelled and replaced via a basic ray-tracing algorithm [20], valid for frequencies above approximately 1kHz, and being much more efficient to compute than a 2-D or 3-D DWM. It is noted that other approaches can improve the high frequency response of 2-D and 3-D DWM generated RIRs, e.g [18] and offer an alternative solution to a geometric acoustics implementation. However the key point with the approach adopted is that not only does it result in shorter run times but also facilitates a significant reduction in the system memory used, allowing larger and more complex room models to be simulated. *RenderAIR* therefore allows the user to offset objective RIR quality against simulation time and memory use via three independent hybrid RIR modelling approaches:

- 3-D rectilinear DWM with variable sample rate and early-part truncation options.
- 2-D triangular DWM late-part tail generation.
- High frequency ray-tracing.

The user is able to decide how a RIR might be best arrived at based on available system resources, and an estimate for how long it will take to run a complete simulation based on the nature of the DWM algorithm used. Within these constraints an optimal hybrid solution can be arrived at that balances the desire for a high resolution 3-D DWM model against required execution time and memory use, with a 2-D DWM and ray-tracing being used appropriately to compensate for those aspects that cannot be accurately simulated using a bandlimited 3-D DWM. Once the RIR components have been generated *RenderAIR* optimises the fit of the individual responses to deliver a final, complete RIR.

Of course the propagation characteristics of a 2-D and 3-D DWM will clearly differ. In a 3-D DWM a signal will travel through a volume of the 3-D virtual space, and interact with 2-D boundaries. In the 2-D case a signal is constrained to a single plane and will travel through this two dimensional representation, interacting with 1-D boundaries. This difference in behaviour will yield inaccurate reverberation times for a 2-D implementation when compared with the full 3-D case. Furthermore, for non-trivial geometries having multiple materials at the boundaries, the 2-D reverberation time will vary depending upon the position of the selected 2-D plane. Adapting the Norris-Eyring equation for reverberation time for theoretically diffuse 3-D and 2-D soundfields [21] obtained from *RenderAIR* geometry and surface data gives two different ideal decay curves for each case which yield the additional gain or attenuation required in each octave band to match the 2-D response to the 3-D response. These gain modifiers are applied to a 2-D RIR using octave-band FIR filters in combination with the (post-process) air absorption compensation filtering. Note finally that the 2-D DWM is selected to intersect with both sources and receivers, although this is not always possible when more that two source/receivers are defined and multiple passes may be required.

3.6. DWM Implementation

In *RenderAIR* an *air-node* is defined as a *N*-port lossless scattering junction that has no surrounding point of intersection with a bounding surface as determined by the modelled geometry. In the 2-D case the DWM used is the commonly exploited 2-D triangular topology based on a triangular tessellation/sampling grid for a 2-D plane and for the 3-D case a rectilinear DWM is used which further exploits the fact that this mesh topology can be partitioned into mutually exclusive subgrids offering significant additional savings in both computation time and system memory [see e.g. 8]. A *boundary-node* is defined as a lossy terminating junction at the boundary of the DWM inheriting acoustical/topological properties according to the defined 3-D space geometry. All boundary-nodes are W-based and use KW-pipe converters on their connecting ports to interface with K-based air-nodes. Frequency dependent absorption at a 1-D terminating boundary-node is simulated and implemented with a minimum phase *n*-th order IIR filter [15]. The transfer function is determined according to the given surface absorption coefficients and the number of connecting air-nodes [11]. A mapping strategy is implemented where reverb time measurements for values of *r* in the simpler non-frequency dependent boundary case are compared with those obtained from theory based on geometry and surface data. The difference in calculated and measured reverb times indicates how given values of *r* should be compensated when used in the boundary filter design, and this in turn helps to compensate for the non-ideal behaviour of a 1-D DWM termination. If a surface is defined as diffusing then a layer of diffusion nodes [12] are placed between the boundary and air-nodes. These nodes are also W-based, and again KW-pipes are included on all connecting ports. Two types of air-node exist and both are K-based such that all inter-nodal communication is based on physical K-variables, beneficial in terms of overall efficiency. An interfacing air-node has to potentially communicate with boundary, air and diffusing layer nodes. A standard K-based air-node only has to communicate with other K-based air-nodes. The *boundary-domain* therefore consists of:

- *M*-port, W-based terminating boundary-node, with $M < 6$, incorporating KW-pipe connecting ports and frequency dependent absorption. *M* varies according to how the mesh fits the required geometry.
- A single 6-port lossless W-based diffusing layer air-node incorporating KW-pipe connecting ports.
- A single K-based 6-port interfacing air-node.

4. RESULTS

4.1. Overview

The main objective of these tests is to demonstrate how *RenderAIR*, and more generally, DWM modelling might be used to successfully synthesize a virtual environment such that the RIRs produced are appropriate for auralization. The secondary objective is to evaluate the computational savings the hybrid approach offers and how they compare to a full 3-D render. These tests will be based on a model of an actual room and *RenderAIR* data will be compared with RIR measurements obtained from within this space and with a purely geometric acoustic simulation produced using the ODEON room acoustics simulation software.

4.2. The Test Room

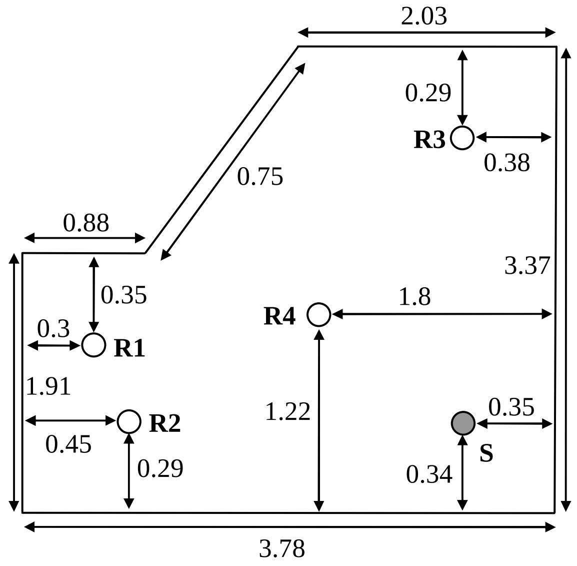

Figure 3: *Plan view of the Music Department test room, with all dimensions given in metres. The sound source is located at S, with receivers at R1-R4. The floor to ceiling height is 2.49 m and the space has an approximate volume of 27 m3. Both source and receiver are set at a height of 1.5 m.*

The test room is a rehearsal space in the Music Department at the University of York. It is relatively small, with a volume of approximately 27 m^3, implying that a high sample rate 3-D DWM model can be computed reasonably. It has no soft furnishings, and all items of furniture have been removed for the purposes of this experiment. The main surfaces in the room consist of painted plasterboard, cork floor tiles, standard ceiling tiles and glass windows. Hence despite being small, the room is quite bright sounding and the lack of absorbing materials, together with parallel walls, indicate a highly modal response. A plan view of the room with the single source S and receiver positions R1 – R4 marked is shown in Figure 3. Both source and receivers are set at a height of 1.5m, hence one 2-D DWM is required as this will intersect with all points of interest, with the floor to ceiling height of the room being 2.49 m.

4.3. RIR Measurement

RIR measurements for this test room are obtained for five different cases as follows:

Case 1 - Actual Test Room.

Case 1 is achieved by using a 4-channel Soundfield SPS422B microphone in each of the four receiver positions, R1-R4, to capture the response of the space to a 15s 22 Hz-22 kHz logarithmic sine-sweep excitation, with the loudspeaker positioned at S. Deconvolution of these responses with the inverse of the sine-sweep signal yields the actual RIR. For the purposes of these experiments only the first-order W-channel omnidirectional pressure based RIR is used.

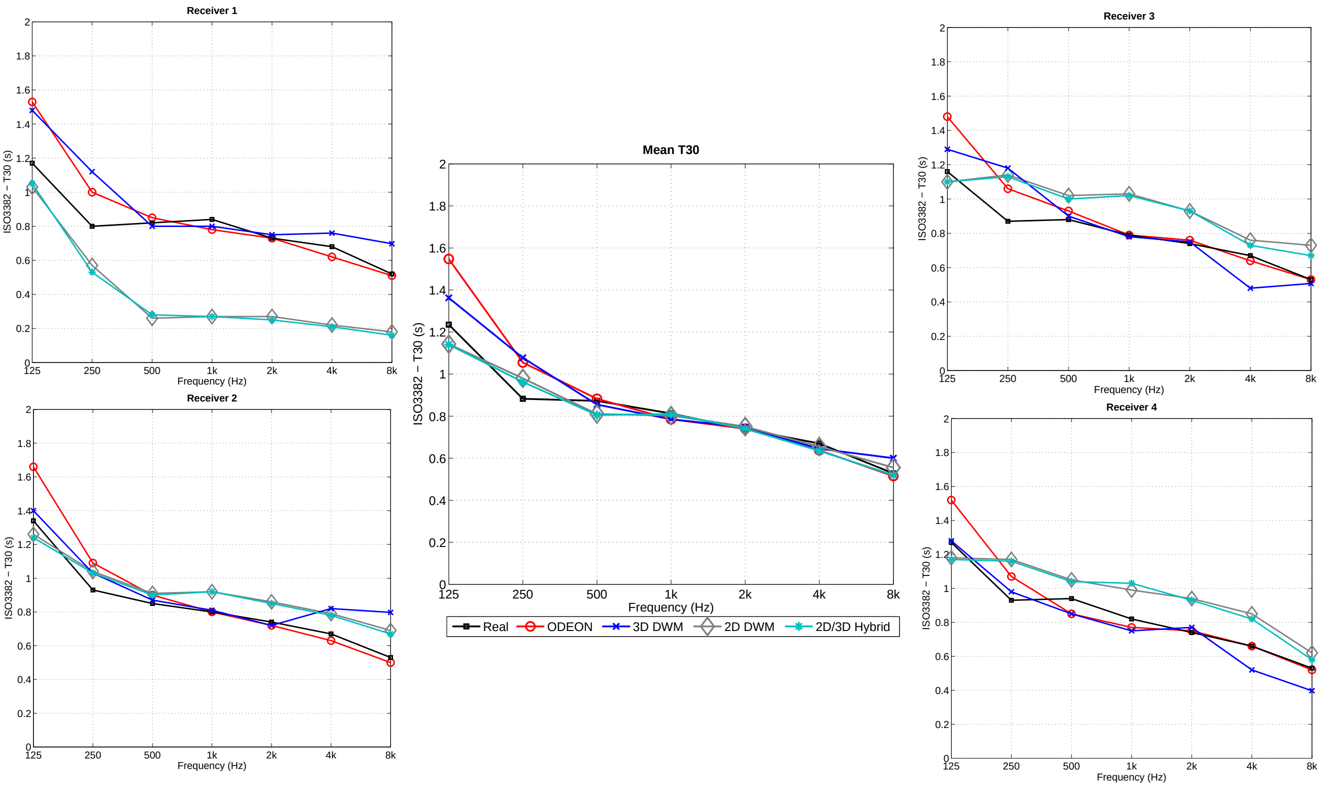

Figure 4: *Octave band ISO3382-T30 values for Receivers 1-4 for each test case as described in Section 4.3, together with mean values averaged across receiver points. The key for all cases is given at the bottom of the central graph.*

Case 2 - Geometric Acoustic Model

Based on physical measurements of the test space. Ideal impulse applied at the input, with outputs based on first-order W-channel omnidirectional RIR for receiver positions R1-R4. 20,000 rays used for the simulation. The RIR is obtained from the energy-time response by adding individual reflections with random phase according to ODEON's own B-format encoding algorithm.

Case 3 - 3-D DWM Simulation

Based on physical measurements of the test space. Low pass Gaussian input function applied for a 3-D sub-gridded rectilinear mesh and the outputs at each receiver point are taken from a single air-node, equivalent to a W-channel RIR. Mesh sample rate selected such that the final bandwidth is valid to 10 kHz.

Case 4 - 2-D DWM Simulation

Based on physical measurements of the test space. Low pass Gaussian input function applied for a 2-D triangular mesh and the outputs at each receiver point are taken from a single air-node as before. Mesh sample rate selected such that the final bandwidth is valid to 22050 kHz. Reverb time correction applied as described in Section 3.5.

Case 5 - 3-D/2-D Hybrid Simulation

Based on physical measurements of the test space. Low pass Gaussian input function applied and outputs at each receiver point are taken from a single air-node as before. The early part is a 3-D simulation valid to 10 kHz, truncated at 27 ms. The RIR tail uses a 2-D DWM model valid to 18 kHz with reverb time correction applied.

For both the DWM and the geometric acoustic models building materials were initially identified and appropriate absorption/reflection coefficients applied. Unfortunately the results were very non-ideal and resulted in RT60 predictions that were significantly shorter than the actual measurements. Given that the space has little absorption and demonstrates strong modal effects, this is not surprising. Also the difficulty in selecting appropriate boundary filter material conditions for simulating a given real-world space has been highlighted in previous round-robin studies [22]. Hence DWM absorption/reflection coefficients were optimised according to average octave band RT60 values for the actual test room and then applied to all surfaces. Absorption/reflection coefficients were optimized similarly for ODEON. No diffusion modelling has been applied as all surfaces in the real space are hard, flat and smooth.

4.4. Reverberation Time Measurements

RT60 values for each RIR are calculated in octave bands from T30 according to ISO3382 [22]. These values are averaged across all receiver positions to arrive at a final value indicating overall behavior of the space. The results are presented in Figure 4 varying with test case for each individual receiver R1-R4, and then summarized by averaging across receiver points for each test case to give a measure for the room under study.

Note first of all that due to the bandwidth limitations implied by Case 3 (valid up to 10kHz), the 16kHz octave band results are not included. Below 250Hz the validity of RT60 as a metric can be questioned due to the dominance of modal resonances, noting that the Schroeder frequency for this room can be estimated as approximately 270Hz. Hence results are presented for octave bands from 125Hz - 8kHz. All plots are in good general agreement with results from the actual space. In particular the average measurements give a very good match to both real world and geometric acoustics results. However, results for individual receiver points are a little more revealing. In particular the results for R1 are not as good as for R2-R4, with the 2-D DWM and hybrid simulations being somewhat below the general trend. This is possibly due to R1 being in close proximity to a corner leading to an overly modal response and a non-diffuse soundfield at this location (note that R2-R4 will potentially be more significantly influenced by the soundfield in the wider space). The 3-D DWM however, gives a good response for R1, thereby also demonstrating that a 2-D DWM implementation can only ever capture a certain aspect of a complete simulation, with the potential for missing out important acoustic features of the total room response. This point is also evident in the results for R2-R4 although to a lesser extent than for R1 and with T30 values generally greater than those of Cases 1-3. The 3-D DWM results give a much closer match to Cases 1 and 2. The hybrid result follows almost exactly that of the 2-D DWM case. This is a reasonable expectation given that beyond the 3-D early part truncation point the remainder of the hybrid RIR that forms the major part of the total response is synthesized from a 2-D DWM. As a result, the properties of the 2-D response will tend to dominate this hybrid result, especially given that modal resonances causing these longer T30 values will decay over the whole length of the RIR, most of which is obtained from the 2-D simulation.

4.5. Low Frequency Response

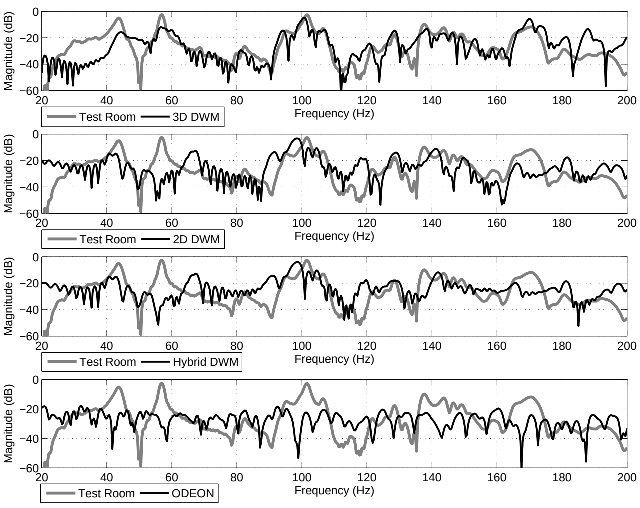

Figure 5: *Low frequency response up to 200 Hz for Receiver Point R2, from top: Test Room compared with 3-D DWM; Test Room compared with 2-D DWM; Test room compare with Hybrid RIR; Test Room compared with geometric acoustic model.*

Figure 5 shows the low frequency response up to 200 Hz obtained from RIRs for R2. The top figure shows the Test Room (Case 1), and the 3-D DWM (Case 3). The next plot shows the results for the Test Room (Case 1) and the 2-D DWM (Case 4) followed by the Test Room (Case 1) and the Hybrid DWM RIR (Case 5). Finally the bottom plot shows the results for the Test Room (Case 1) and the results obtained from ODEON (Case 2). Note in the last example the poor modal response with no distinct resonant peaks. The results for the 3-D DWM provide a very good match to the results obtained from the real room, matching both individual peaks and the overall response profile. The 2-D DWM case gives a good approximation to the overall response, although individual peaks do not match as well as for the 3-D DWM, demonstrating that this is clearly an approximation to real world conditions, with important modal behaviour either missing or inaccurately captured. As demonstrated in Figure 4, and discussed in Section 4.4, the Hybrid DWM follows closely the characteristics of the 2-D DWM. Hence the benefit of this hybridization approach is in the more complete capture of early reflections, particularly from ceiling and floor.

4.6. Performance Benchmarking.

Table 1 presents comparative performance results for each test case 2-5. Note that 0.8s RIRs were generated for each example.

	Case 2	Case 3	Case 4	Case 5
Time (Hrs:Mins)	00:54	14:18	00:37	01:38
Memory Used (Mb)	--	72	4	76

Table 1: *Performance data for each of the 4 simulated cases, showing total time elapsed to produce a 0.8s RIR and the total system memory used.*

Clearly the 2-D (Case 4) and hybrid solutions (Case 5) bring the total RIR synthesis time down to reasonable levels compared with a full 3-D render, which even in this case is only valid to 10 kHz. Note in particular that the 2-D DWM offers a good compromise when computational resources are limited.

5. CONCLUSIONS

This paper has examined how hybrid RIR synthesis might be used to better enable virtual environment simulation through the use of otherwise computationally expensive DWM models. High resolution 3-D DWMs of large spaces are still an impractical option for current desktop based computers, due to long runtimes and large memory requirements. Hybrid solutions have been proposed that can be optimised according to user requirements in terms of absolute accuracy and available computer resources. Some hybrid modelling options have been tested using ISO3382 T30 as a benchmark, and results compared with real-world measurements and more established techniques. The room selected for study has a small regular construction, strongly reflecting surfaces, low levels of absorption and dominant low frequency modes that make this test non-trivial. The 3-D DWM, compensated 2-D DWM and hybrid simulation all produce results that are a good general match to the real room in terms of mean RT60 values. They also produce results comparable to a more established geometrical acoustics technique with a modal analysis indicating that DWM based simulations, and in particular the 3-D case, give the more reliable measure of the acoustic characteristics of the target space. Hybrid solutions offer a good compromise and make significant savings in computational resources although measurements taken from individual receiver points reveal that 2-D based simulations are still some way from the accuracy of a full 3-D simulation. This is obvious given that a 2-D simulation will never accurately capture the behaviour of a full 3-D soundfield, although the results are encouraging enough to suggest that as part of a hybrid response they do have a role to play when it comes to offering computational savings. Work is clearly needed in optimizing the nature of the RIR hybridization, as well as improving the core DWM/boundary-filter implementation. Note also that only two hybrid strategies have been tested in this paper (compensated 2-D; 3-D early part/2-D late part, valid to 10kHz), and there is scope for other possible partitioning strategies of the time-frequency plane. As the development of *RenderAIR* is ongoing, there is significant possibility for capitalizing on recent work relating to boundary formulations [9, 10] that offer improvements over 1-D DWM terminations. Even with such new solutions implemented, boundary simulation requires additional testing based on actual acoustic data and how this might be mapped appropriately to the DWM domain. Once complete more rigorous testing can begin, based on existing room simulation round robin data [22] and these results will then help to form the basis of a series of subjective listening tests.

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