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Source excitation strategies for obtaining impulse responses in finite difference time domain room acoustics simulation

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ABSTRACT

This paper considers source excitation strategies in finite difference time domain room acoustics simulations for auralization purposes. We demonstrate that FDTD simulations can be conducted to obtain impulse responses based on unit impulse excitation, this being the shortest, simplest and most efficiently implemented signal that might be applied. Single, rather than double, precision accuracy simulations might be implemented where memory use is critical but the consequence is a remarkably increased noise floor. Hard source excitation introduces a discontinuity in the simulated acoustic field resulting in a shift of resonant modes from expected values. Additive sources do not introduce such discontinuities, but instead result in a broadband offset across the frequency spectrum. Transparent sources address both of these issues and with unit impulse excitation the calculation of the compensation filters required to implement transparency is also simplified. However, both transparent and additive source excitation demonstrate solution growth problems for a bounded space. Any of these approaches might be used if the consequences are understood and compensated for, however, for room acoustics simulation the hard source is the least favorable due to the fundamental changes it imparts on the underlying geometry. These methods are further tested through the implementation of a directional sound source based on multiple omnidirectional point sources.

1. Introduction

The finite difference time domain method (FDTD) is a discrete spatio-temporal numerical simulation method that has been shown as being appropriate for modeling acoustic wave propagation in an enclosed system [1,2,3]. Recent developments in frequency dependent absorbing and diffusing boundaries [4,5] offer the potential for a more complete approach to room acoustics simulation. However, full audio bandwidth simulations for even a small room of any acoustic interest are very demanding in terms of both computation time and required memory, and so these techniques are often best used for low-frequency simulation only. The consequence of these computational requirements being that work in this area has usually relied on offline computation of the impulse response (IR) for a given space and source/receiver combination. This IR can then readily be used in a real-time audio convolution scheme suitable for auralization purposes. In addition, these results can be applied to any type of room acoustic analysis that is based on deriving parameters from a suitable IR, including computation of typical room acoustic metrics, such as reverberation time (T_{30}), or clarity ($C_{50/80}$). However, for some other purposes, such as visualization of the sound field, calculation of the whole field based on some form of smoothed excitation signal applied over a longer duration might be a more suitable approach.

Generally, in such room acoustic simulations, a sound source is considered as a time-varying pressure signal applied to a single point on the FDTD grid. A receiver is defined as any other grid point where the numerical response to this source signal is measured, and for auralization purposes, receivers may be grouped individually (mono), as a pair (stereo), or as an array of points (multi-channel surround-sound). In this way, the signal observed at the receiver is analogous to a measurement microphone that acts as a scalar sensor of sound pressure. Based on this concept, in order to obtain an IR, the sound source should ideally propagate omnidirectionally and demonstrate a flat frequency response over a defined bandwidth. To this end, different source signals have been proposed in the literature enabling appropriate control over this required bandwidth (see e.g. [6,7]), with single-point source direc-

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tionality dependent on the propagation characteristics of the FDTD grid itself. Typically however, although usually defined in terms of sound pressure, the discrete-time excitation signal itself can take any form such that it is sampled commensurate with the sampling rate of the FDTD grid used (see e.g. [8]).

The application and implementation of such a defined pressure-like signal has been characterized in [10], either as a *hard source*, *additive source*, also known as a *soft source* [11], or *transparent source*. Each of these source types, hard, soft and transparent, differ in terms of their relative advantages and disadvantages. With a hard source, the pressure at the source grid point is determined by the driving excitation function alone, ideally coupling the signal into the grid, making them the simplest and easiest to implement. However, in doing this they disregard the underlying grid function, and for certain signals, this discontinuity between source function and grid function results in large, low frequency pressure ripples that can be observed at all other grid points [11]. In addition to having little correlation with actual physical sound sources, hard sources also act as signal scatterers for any incident wave. As a consequence they become a discontinuity or perturbation in the medium, or can be considered as a sound radiating, perfectly reflecting boundary node with a size corresponding to the spatial sampling interval [6].

With a soft source, the driving excitation function is added (hence also *additive* source) to the numerical pressure value at the source grid point. The implementation is no more difficult than with a hard source, with the added advantage of no perturbation being added to the problem domain, and hence no additional numerical artefacts or reflected components. In this case, the disadvantage is that the pressure function at the source grid point no longer resembles the applied excitation function. Ideally, as recommended in [11], the response should be measured at the source grid point and used to normalize the output at other grid points, with the suggestion that this is the reason why soft source excitation has not been extensively used in the acoustics literature. In addition, soft sources exhibit solution growth due to source–boundary interaction effects [6], requiring further pre-or post-simulation conditioning to obtain a useful IR, e.g. differentiation of the original pressure-based signal [6], or pre-filtering [8].

A transparent source, as defined in [10] is one that propagates the same signal as a hard source, but does not act as a signal scatterer (it is *transparent* to an incident wave). It therefore offers the benefits of a hard source in terms of how it couples the excitation signal into grid, but does not result in a perturbation in the problem domain. The disadvantage with such a source being that a compensation filter, required to remove the effect of the fixed grid point, must be computed prior to simulation, and that the excitation function itself becomes more complex – and computationally demanding – to implement.

This paper contains several practical results that affect how FDTD simulations should be conducted with a special emphasis on these three excitation types. We also note that these different source models have different physical interpretations (see e.g. [6,7]) with different practical consequences as a result, and it is this latter point that is the focus of this paper. First, we demonstrate that a unit impulse is sufficient and good choice for an excitation signal, especially with double precision accuracy (Section 2). Similarly, we show by numerical examples that a hard source introduces error in detected modes, soft sources cause an offset in the received level, while a transparent source performs without these artefacts, thus making it an attractive choice (Section 3). Finally, we demonstrate that hard sources have serious problems if they are to be used to create directive sources, instead, soft and transparent sources behave in an ideal manner and reproduce the directivity patterns as expected (Section 4).

2. The finite difference time domain method in room acoustics simulation

There are basically two different FDTD formulations applied in room acoustic simulation: the vector wave equation based model that considers both sound pressure and particle velocity [11], and the scalar wave equation based model that uses only sound pressure [5]. However, they are equivalent in terms of the results they produce, and in this work we use the scalar wave equation model as it is computationally more efficient [9]. The following describes this FDTD method for the 3-D acoustic wave equation:

$$p^{n+1}_{i,j,k} = \lambda^2\left(p^n_{i+1,j,k} + p^n_{i-1,j,k} + p^n_{i,j+1,k} + p^n_{i,j-1,k} + p^n_{i,j,k+1} + p^n_{i,j,k-1}\right) + 3(1-3\lambda^2)p^n_{i,j,k} - p^{n-1}_{i,j,k} \tag{1}$$

The 2-D case is given by:

$$p^{n+1}_{i,j} = \lambda^2\left(p^n_{i+1,j} + p^n_{i-1,j} + p^n_{i,j+1} + p^n_{i,j-1}\right) + 2(1-2\lambda^2)p^n_{i,j} - p^{n-1}_{i,j} \tag{2}$$

where i,j and k denote spatial indices, p^n is the pressure value at time-step n and λ is the Courant number, which is usually set to the limiting value such that for the 3-D case $\lambda = \frac{1}{\sqrt{3}}$, and for the 2-D case $\lambda = \frac{1}{\sqrt{2}}$ thereby simplifying the above expressions further. This approach results in a rectilinear spatio-temporal sampling of connected nodes across the problem space. Boundary conditions are dealt with separately, and can be defined with parameters that determine both frequency dependent absorption [4] and diffusion characteristics [5]. The grid sampling rate f_{update} is related to the spatial sampling distance d according to $f_{update} = \frac{c\sqrt{\lambda}}{d}$ where c is the speed of sound.

It is worth noting that this simple scheme suffers from direction dependent dispersion error, the effects of which can be improved upon by using more advanced schemes with a larger stencil [12,13]. However, we consider this scheme general enough to be applied here with any exceptions to this assumption explicitly noted in the text.

3. Source excitation

A number of methods for applying source excitation have been explored in FDTD and related literature for room acoustics simulation. The aim of any such simulation is to obtain the impulse response $h(n)$ of the system consisting of a (generally) enclosed geometry together with a given source/receiver combination and hence the unit impulse is the ideal source excitation input.

3.1. Choice of source signal and finite precision variable considerations

In real room acoustic measurement a specific analytic excitation signal is used to obtain the impulse response of the space given that a perfect impulse cannot be applied. In FDTD simulations a similar approach is adopted through the use of an appropriate time varying source function (see e.g. [11] for a recent summary) noting that for non-impulse like excitation, the time varying input signal should also be deconvolved from the output to obtain the impulse response. Given that the FDTD equations are linear and time-invariant, as are the three given source types, it is therefore possible to use convolution to obtain the output for any given input signal based only on the measured impulse response. The advantage here being that the unit impulse is the simplest to apply and also the shortest – and hence most efficient in terms of reducing overall simulation run-time, especially when compared to using longer form analytical signals (e.g. exponential sine sweep as commonly used in room acoustic measurement) or direct excitation with an anechoic audio signal. If some other excitation is required, it can be applied post simulation by convolving the desired