VIRTUAL ROOMS RECREATION FOR WAVE FIELD SYNTHESIS

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ABSTRACT

Advanced multichannel sound systems such as Wave Field Synthesis (WFS) allow to recreate spatial wide sound scenes of sources. The recreation of the illusion of a 3D natural and realistic sound scene can be achieved by means of virtual rooms where the wave field is simulated. Such wave field is used as a source of information for the convolution of WFS sound sources with extrapolated impulsive responses in these virtual rooms. To obtain the needed plane waves for auralization, a complete description of the sound field is needed, including an accurate knowledge of the particle velocity. In this paper, virtual rooms are simulated by means of Finite-Differences Time Domain method. This method provides a complete solution of the sound field variables in a wide frequency band and can be used to produce both the impulsive responses of pressure and particle velocity for plane wave decomposition, prior to auralization. To illustrate its applicability, a set of rooms consisting of a typical auditorium room, a cinema and a perfect cube are shown and evaluated.

1. INTRODUCTION

There is a pronounced technological demand for auditory scene analysis. For the modeling of a sound auditory, it is often advisable to start with a scene analysis, for example, in systems for analysis in architectural acoustics or in systems for quality assessment of speech and product sounds.

The acoustic properties of a real enclosed space can be defined by measuring the room Impulse Response (IR) at a specific listening point. This is done for an input signal applied at a given sound source location. In order to obtain simulation environment responses, two methods have traditionally used for obtaining a IR based on a description of the room geometry, named as ray tracing, the image source method and hybrids models based on these techniques. These methods present several limitations being common to both that they are not valid for the low frequency region. At these frequencies, where the wave behaviour of sound and the effects of room modes are more noticeable, these methods are not appropriate.

Besides, in multichannel reproduction systems, dry sources are convolved with these impulse responses to achieve a certain hall sensation. Although, they are usually reproduced through headphones as binaural reproduction, with the "inside-head" effect drawback, or using near field loudspeakers and thus, limiting to a reduced listen area. Wave Field Synthesis can involve a great number of listeners in a wide listening area and reproduces a high number of sources at a time. For this reason, the process of recreating the acoustics of a hall in another room, known as auralization, is a powerful tool for considering such number of elements.

Wave Field Synthesis (WFS) is a sound reproduction method that, by analogy to holography and in base to the Huygens principle, reproduces an acoustic field inside a volume from the stored signals recorded in a given surface. Huygens principle tells that the wave front radiated by a source behaves like a distribution of sources that are in the wave front, named secondary sources. WFS was first proposed with application to 3D sound by Berkhout [1]. The synthetic wave front is created by loudspeaker arrays that substitute the individual loudspeakers. The main advantage of these systems is the great extension of the useful listening area; since all the loudspeakers compose an accurate wave field reproduction zone.

The listening area is determined by the loudspeakers, which are fed with signals that create a volumetric velocity proportional to the particle velocity normal component of the original wave front. In Figure 1, a typical WFS configuration is presented, where a virtual sound source is synthesized in the location of the listener by using a loudspeaker array. However, unlike stereo systems, the synthesized field is not only valid in this location, but also in the rest of the room. Apart from the original WFS reproduction sources, it is feasible to reproduce live recordings [2] or to render dry sources with recorded impulsive responses [3].

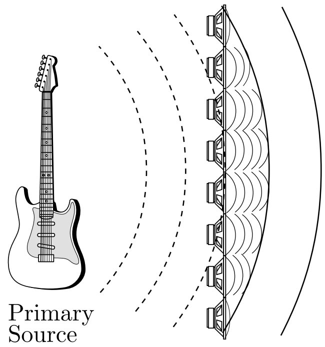

Figure 1: Loudspeaker array technology applied to Wave Field Synthesis rendering systems.

For WFS applications, adding the echoes and reverberation of a virtual hall to the notional sources, which is known as auralization, enhance the source localization and tridimensional sensation. The quality of the auralization is related to the room model accuracy used for calculation of the impulse response, either by means of hall measurements or by computer synthesis of virtual scenes.

In terms of rendering techniques, each of the L loudspeakers would require an input signal $q[k]$, obtained as a convolution of the N virtual signal $s[k]$ with a filter matrix $W[k]$ of measured or

$$p^{(1)}(\mathbf{r},\omega) = \frac{-j\omega}{4c}\int_0^{2\pi} p(\theta,\omega)\cos\phi H_1^{(1)}\left(\frac{\omega\Delta r}{c}\right)Rd\theta + \frac{-j\omega}{4c}\int_0^{2\pi} j\rho_0 c u_n(\theta,\omega)\cos\phi H_0^{(1)}\left(\frac{\omega\Delta r}{c}\right)Rd\theta, \tag{1}$$

$$p^{(2)}(\mathbf{r},\omega) = \frac{-j\omega}{4c}\int_0^{2\pi} p(\theta,\omega)\cos\phi H_1^{(2)}\left(\frac{\omega\Delta r}{c}\right)Rd\theta + \frac{-j\omega}{4c}\int_0^{2\pi} j\rho_0 c u_n(\theta,\omega)\cos\phi H_0^{(2)}\left(\frac{\omega\Delta r}{c}\right)Rd\theta. \tag{2}$$

synthesized impulsive responses:

$$q[k] = W[k] * s[k]. \tag{3}$$

There are two different approaches to obtain this filter matrix $W[k]$[4]. On the one hand, in the *Model-Based rendering*, point sources and plane waves are commonly used to obtain $W[k]$ coefficients. In this case, the matrix just contains weight and delays coefficients [5]. On the other hand, in the *Data-Based rendering*, the operator $W[k]$ cannot be measured or simulated from a source to a listener position. Impulsive responses require information of traveling wave direction. For this purpose, a special setup of microphone array and a dedicated signal processing is required [6].

The aim of this work is to introduce numerical simulations of virtual rooms and to use them for auralization in WFS systems. For this purpose, physically-based modelling by means of discrete-time models is employed. This concept is developed on the basis of solving numerically partial differential equations that rules wave phenomena. The accuracy of solutions is related to the numerical method considered. Although the latter is mentioned at [7], it has not been profusely analyzed yet.

In addition, since WFS rendering requires traveling wave information, typical direct impulse response measurements of real and/or virtual scenes are insufficient because of the lack of spatial information. To extrapolate plane wave responses, particle velocity recordings are needed. Consequently, the time-discrete model used for that purpose must solve both Euler and Continuity equations and thus the sound field solution of the pressure and particle velocity components can be obtained. The point in this approach is that the approximation of the impulse responses based on geometry simulations do not obtain a complete description of the sound field, as time-discrete models do. The reason is that geometrical methods cannot reproduce the full complexity of the impulse response, specially for low frequencies, and particle velocity components are not computed.

In this paper the feasibility of implementing an auralization algorithm based on Finite-Differences Time Domain (FDTD) with applications to WFS is analyzed. This will create virtual rooms for auralizating custom notional sources of WFS systems. The information obtained form the FDTD simulation will be adapted to obtain data-based rendering filter coefficient matrix and exploit its properties in auralization purposes. FDTD method has been chosen because it provides a systematic complete set of sound field solutions (pressure and particle velocity components) from a description of initial and boundary conditions (frequency dependent and non-dependent boundary conditions). The potentiality of this approach is analyzed in two typical rooms: a cinema and a auditorium hall, and on a perfect cube room, as a counterexample of the room modes in low frequency. These virtual rooms are simulated with the some of the typical absorption conditions in real spaces by means of the surface absorption coefficients. As demonstrated in cite [8], this simulation procedure gives encouraging results in complex virtual scenarios.

2. DATA-BASED RENDERING WAVE FIELD SYNTHESIS USING MICROPHONE ARRAYS

In WFS system, it has been shown that plane waves are a simply and efficient way to accomplish the auralization requirement [9]. Circular microphone array has been demonstrated as the most powerful configuration in order to auralizate sound fields by WFS. In this case, incoming and outgoing impulsive responses of pressure to obtain plane waves are obtained by using cylindrical harmonic decomposition. For generating plane waves for auralization, traveling directions must be obtained and thus, particle velocity information of environment must be stored.

Therefore, a similar microphone array configuration and signal processing used for auralization of real enclosures is proposed to be used in virtual scenarios, based on *plane wave decomposition* [6]. Considering a circular discretized point array of radius R, centered in $\mathbf{r}'$, both pressure and normal velocity are stored and represented as $p(\theta,t)$ and $u_n(\theta,t)$, pressure and normal particle velocity component at azimuth angle θ, respectively. With this data set, wave field reconstruction can be performed using Kirchhoff-Helmholtz integrals in cylindrical coordinates, as shown in (1) and (2), where $p^{(1)}(\mathbf{r},\omega)$ and $p^{(2)}(\mathbf{r},\omega)$ are the inverse and forward extrapolated sound fields in frequency domain and $p(\mathbf{r},\omega) = p^{(1)}(\mathbf{r},\omega) + p^{(2)}(\mathbf{r},\omega)$. $H_n^{(1)}$ and $H_n^{(2)}$ are the Hankel functions of the first and second kind, respectively. See Figure 2 for geometrical details.

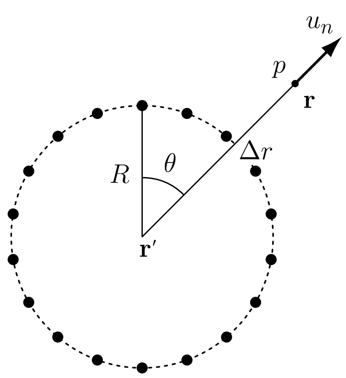

Figure 2: Geometrical details of the circular microphone array configuration.

Observing the Kirchhoff-Helmholtz integrals in the circular arrays case, only the interior of the circle field can be known. In order to obtain any impulsive response, a previous step, based in cylindrical harmonics decomposition, is necessary. Figure 3 shows a generic implementation to feed an auralization processor in WFS reproduction system [10]. In this process, the first step is to obtain a double Fourier transformation of both pressure and normal particle velocity, $P(k_\theta,\omega)$ and $U_n(k_\theta,\omega)$. These are inputs of the

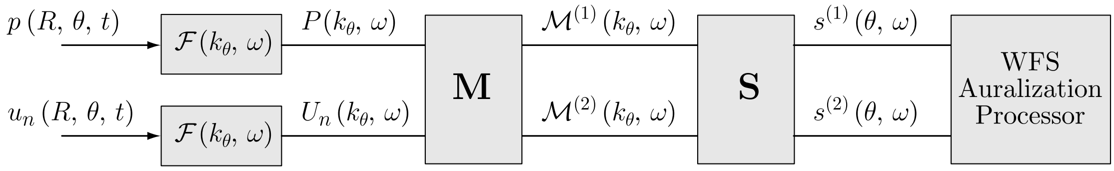

Figure 3: Block-based scheme for obtaining the plane wave decomposition signals for the WFS Auralization processor.

$\mathbf{M}$ matrix filter for obtaining a cylindrical harmonics decomposition ($\mathcal{M}^{(1)}(k_\theta, \omega)$ and $\mathcal{M}^{(2)}(k_\theta, \omega)$). These data set denotes the incoming and outcoming ambisonics representation of the sound field. Although it can be useful to derive the complete sound field, a cylindrical harmonics to plane wave decomposition ($s^{(1)}(\theta, \omega)$ and $s^{(2)}(\theta, \omega)$) allows a easy way to compute these values (matrix filter $\mathbf{S}$). For more details, refer to [6].

One of the main advantages lies on the fact that storing all the impulsive responses is not needed. Taking into account that most of the numerical methods require a great number of spatial discretization points to obtain a proper accuracy, by storing a few impulsive responses (pressure and particle velocity), the complete sound field can be obtained. Moreover, different loudspeaker array can be used since it is possible to obtain the appropriate plane waves.

3. DISCRETE-TIME MODELS FOR ROOM ACOUSTICS SYNTHESIS

3.1. Discrete-Time Based Model Properties

Wave equation is used in mathematics to describe the sound propagation. An impulse response at a given listener location can be obtained by solving this equation. However, this is not an easy task because of the discontinuities present on the wave field. Therefore, it is not possible to obtain the analytic solution of the wave equation, except for simple geometries, and thus, it must be approximated. Briefly speaking, there are two different approaches for the modeling of room acoustics [11]: Geometrical-based methods and wave-based methods.

Geometrical-based methods are based on ray theory in which sound is supposed to act as rays with specular reflections. However, this is only applicable if the wavelength of sound is relatively small according to the area of surfaces in the room and relatively large according to the roughness of surfaces. In ray based methods, a geometric algorithm is used to find ray paths along which sound travels from a source to a receiver. Mathematical models are then used to obtain the filters corresponding to the response of the sound waves traveling along each path. Finally, an impulse response is constructed by combining the filters for each propagation path.

Wave-based methods give the most accurate results because the wave equations are the calculus basis, which best represents the behaviour of sound. The numerical solution of initial-boundaries-PDE of acoustic wave equation is performed with methods that subdivide a geometry into volume or surface elements. Discrete-time models are very useful for wide frequency-band analysis, non linear interactions and impulsive responses treatment. These are based on a discretized physical space where the wave equation is applied to obtain a discrete solution of the sound field variables. In this way, numerical methods provides an accurate and complete solution of sound variables, such as pressure and particle velocity.

Since time-discrete methods try to solve the wave equation as to provide a discrete solution of sound field, a brief overview of wave equation will be introduced. Assuming small perturbations from rest, acoustic wave equations for absorptive medium are given by [12]:

$$\rho(\mathbf{r})\frac{\partial \vec{u}(\mathbf{r},t)}{\partial t} = -\,\vec{\nabla} p(\mathbf{r},t) + f_s(\mathbf{r},t), \tag{4}$$

$$\frac{\partial p(\mathbf{r},t)}{\partial t} = -\,\rho(\mathbf{r})c^2(\mathbf{r})\vec{\nabla}\vec{u}(\mathbf{r},t) + g_s(\mathbf{r},t), \tag{5}$$

where $\vec{u}$ is the vectorial gas particle velocity, p is the deviation from ambient pressure, $c(\mathbf{r})$ is sound velocity and $\rho(\mathbf{r})$ is density of the gas at rest. In homogeneous conditions, $c(\mathbf{r}) = c$ and $\rho(\mathbf{r}) = \rho$. On the other hand, $f_s(\mathbf{r}, t)$ and $g_s(\mathbf{r}, t)$ represents density pressure and particle velocity sources respectively. The complete sound scene is determined as a combination of the Euler (4) and the Continuity (5) equations.

3.2. Finite-Difference Time Domain Method

In this section, Finite Differences Time Domain method is presented and discussed. The FDTD method is based on a second order finite-difference approximation of both space and time derivatives in the wave equation. In this kind of scenarios, two quantities are chosen. In air acoustics, these are sound pressure and the three components of particle velocity.

Wave equation is solved as a recursive equation by using backward differences for space derivatives and forward differences for time derivatives around Yee-unit like-cell [13]. In this equation system, spatial variations of pressure yields in time variations of particle sound velocity and a spatial variation of particle sound velocity distribution yields in a time variation of pressure (this recursive procedure for obtaining the solution is known as *leap-frog scheme*). Supposing a homogeneous medium, the complete system equations for FDTD method is:

$$\begin{aligned}
u_x^{i-\frac{1}{2},j,k,n+\frac{1}{2}} &= u_x^{i-\frac{1}{2},j,k,n-\frac{1}{2}} - \varepsilon_x(p^{i,j,k,n} - p^{i-1,j,k,n}),\\
u_y^{i,j-\frac{1}{2},k,n+\frac{1}{2}} &= u_y^{i,j-\frac{1}{2},k,n-\frac{1}{2}} - \varepsilon_y(p^{i,j,k,n} - p^{i,j-1,k,n}),\\
u_z^{i,j,k-\frac{1}{2},n+\frac{1}{2}} &= u_z^{i,j,k-\frac{1}{2},n-\frac{1}{2}} - \varepsilon_z(p^{i,j,k,n} - p^{i,j,k-1,n}),\\
p^{i,j,k,n+1} &= p^{i,j,k,n}\\
&\quad - \varrho_x(u_x^{i+\frac{1}{2},j,k,n+\frac{1}{2}} - u_x^{i-\frac{1}{2},j,k,n+\frac{1}{2}})\\
&\quad - \varrho_y(u_y^{i,j+\frac{1}{2},k,n+\frac{1}{2}} - u_y^{i,j-\frac{1}{2},k,n+\frac{1}{2}})\\
&\quad - \varrho_z(u_z^{i,j,k+\frac{1}{2},n+\frac{1}{2}} - u_z^{i,j,k-\frac{1}{2},n+\frac{1}{2}}),
\end{aligned} \tag{6}$$

Description	63 Hz	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz
Gypsum	0.024	0.024	0.027	0.030	0.037	0.019	0.034
Wool 15 mm	0.150	0.150	0.700	0.600	0.600	0.750	0.750
Velvet	0.050	0.050	0.120	0.350	0.450	0.380	0.360
Linoleum	0.020	0.020	0.020	0.030	0.040	0.040	0.050
Audience	0.160	0.160	0.240	0.560	0.690	0.810	0.780
Wooden Floor	0.150	0.150	0.110	0.100	0.070	0.060	0.070

Table 1: Absorption coefficients of materials in the simulation scenario described in terms of one-third octave frequency bands.

where the notation $p(x, y, z, t) = p(i\Delta x, j\Delta y, k\Delta z, n\Delta t) = p^{i,j,k,n}$, $\varepsilon_{(x,y,z)} = \Delta t/(\rho\Delta_{(x,y,z)})$ and $\varrho_{(x,y,z)} = \rho c^2 \Delta t/\Delta_{(x,y,z)}$ is used. Sampling intervals must be chosen to assure that numerical stability is accomplished. This relation can be found using Von Neumman criteria. In tridimensional case, this relation must assure the so-called Courant criteria:

$$c\Delta t \leq \frac{1}{\sqrt{\left(\frac{1}{\Delta x}\right)^2 + \left(\frac{1}{\Delta y}\right)^2 + \left(\frac{1}{\Delta z}\right)^2}}. \qquad (7)$$

One of the most important downsides of this inequality is that FDTD requires an elevated number of discretization cells in order to obtain some accuracy. Furthermore, it is also necessary to store data of pressure and particle velocity for each point and each time step.

Although usual shape-cell has a cubic-form, Botteldoren [14] proposes a more accurately different shape-cell, known as *Voronoi cells*. It combines a remained computational effort, compared to the classical Yee-like cell, and the accuracy of the problem definition.

3.3. Boundary conditions in FDTD method

To simulate walls that consider real absorption processes within virtual room simulations, time-domain impedances must be included. The classical approach to the solution of this space-time-varying problem is the conversion of differential/integral time dependent problem to a parametric frequency dependent ω through a Fourier transform. This conversion reduces space-time-initial-boundary value problem to a boundary problem with parametric dependency ω. In the acoustic case, boundaries are defined as the rate between pressure $P(\mathbf{r}, \omega)$ and the boundary normal component of particle velocity $\vec{U}(\mathbf{r}, \omega)$, defined as *impedance* $Z(\mathbf{r}, \omega)$. In the time-domain, this rate is expressed as a time convolution,

$$p(\mathbf{r}, t) = \int_{-\infty}^{\infty} \bar{Z}(\mathbf{r}, \tau) \cdot \vec{n} \cdot \vec{u}(\mathbf{r}, t - \tau) d\tau, \qquad (8)$$

where $\vec{n}$ is the vector normal to the boundary condition surface. A direct implementation of equation (8) into FDTD scheme is not an evident task. To solve this problem, several methods have been developed (see [15]). In this study, an approach proposed by Özyörük and Long has been employed [16]. It accounts for a rational representation of measured impedance and conversion of time-domain operators via Z-transform. In this case, filter design is based on a minimum-phase low-order approach of *reflectance* coefficients.

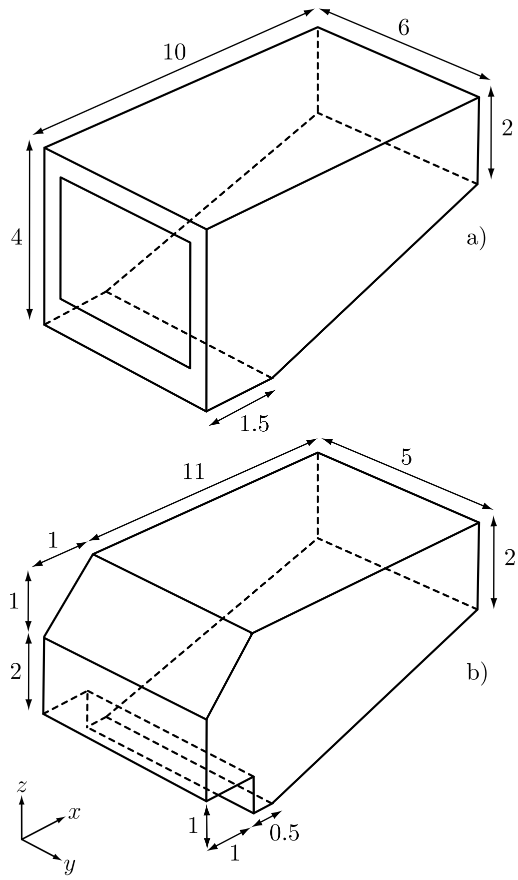

Figure 4: Virtual rooms used for simulation by means of the FDTD method. a) Cinema, b) Auditorium.

4. RESULTS

With the aim of demonstrate the applicability of this method, three rooms were simulated. The FDTD grid parameters were chosen to simulate these three scenarios up to 5512.5 Hz, which means a sampling frequency of 11025 Hz. With this time discretization in a cubic Yee cell, and considering the Courant criteria, a spatial definition of 5 cm was achieved in the three axes.

The latter definitions were employed to simulate three rooms: first, a perfect cube of 5×5×5 m was considered as a counterexample of a proper acoustic design. Secondly, a more convenient

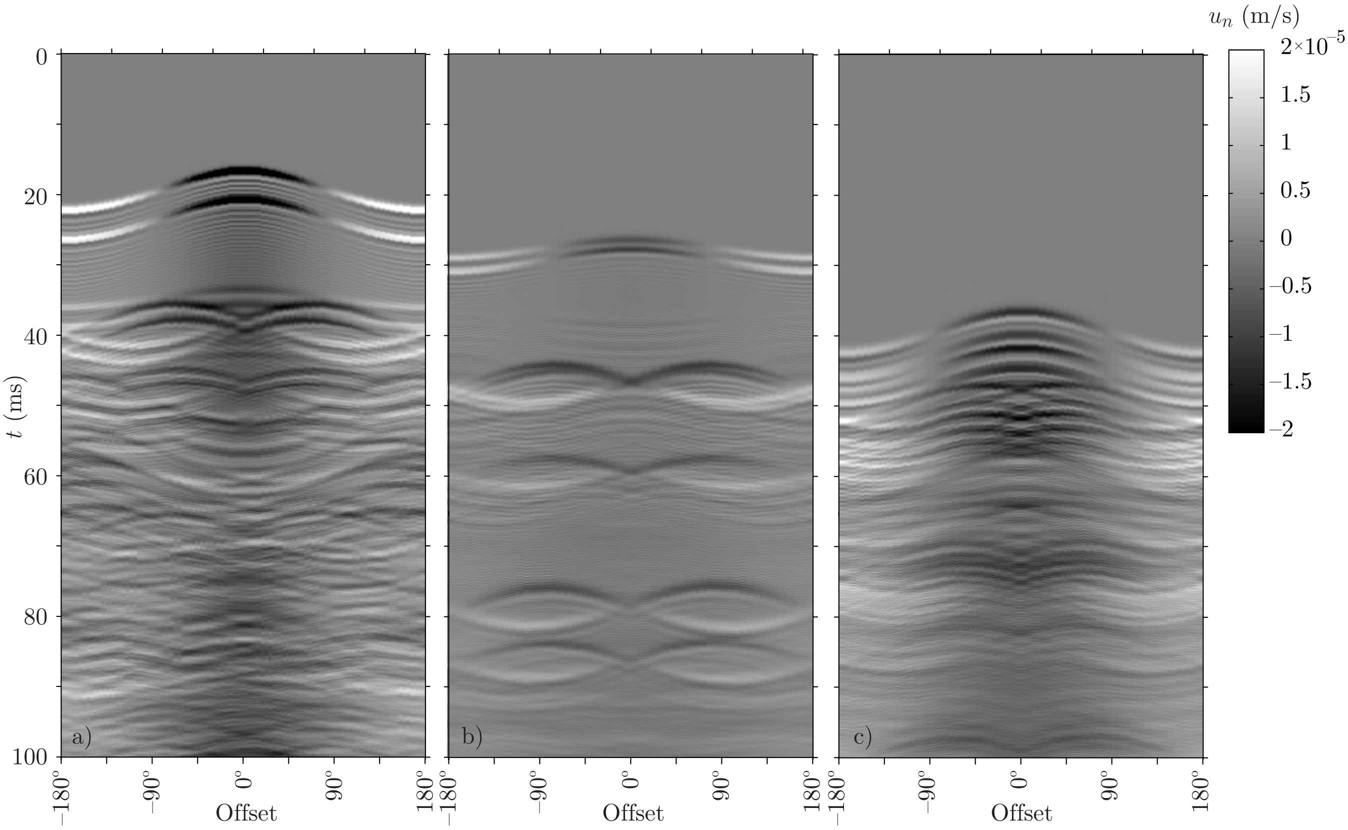

Figure 5: Normal component of particle velocity stored at microphone circular array positions. a) Perfect cube, b) Cinema, c) Auditorium.

room for the real use was created. It is a generic cinema consisting of seating area, surrounding absorptive walls, a reflecting ceiling and a projection screen. Finally, a more complex structure is presented in the form of a many-purposed theater auditorium with a proscenium stage. Figure 4 illustrates both the cinema and the theater virtual rooms together with their boundary dimensions.

These three simulation scenarios consider real absorptive materials, which are described in the form of their absorption coefficients in one-third octave frequency bands [17], given by the Table 1. For the perfect cube case, all interior surfaces were covered with velvet which has a medium absorptive coefficient for mid and high frequencies. The cinema has a variety of materials: side and rear walls have velvet, the ceiling has gypsum, the screen is made of linoleum, floors are covered with wood and the seating is modeled with the audience absorption. The auditorium has also gypsum as a reflecting material in its rear and side walls and wool for the ceiling. Regarding the stage, the enclose is filled with velvet and its floor has linoleum. Finally, the seating is also modeled with audience and the rest of floor is covered with wood.

To perform a comparison between the three scenarios, a virtual microphone array was located at an appropriate position, which corresponds to the hall center coordinates. So, the cube has the virtual microphone array centered at $\mathbf{r}_m = (2.5, 2.5, 2)$ m, the cinema at $\mathbf{r}_m = (5, 3, 2)$ m and the auditorium at $\mathbf{r}_m = (6, 2.5, 2)$ m. The array comprises 144 microphones on a 0.5 m radius circle in the XY plane, achieving an angular resolution of $\pi/72$ radians.

For the three virtual rooms, the sound source is located at a centered position at front side, which corresponds to a source on the down side of the screen in the cinema ($\mathbf{r}_s = (0.35, 3, 2)$ m) and over the scenario for the auditorium ($\mathbf{r}_s = (0.35, 2.5, 2)$ m), as for the cube. The source generates a pressure gaussian pulse since it presents a suitable bandwidth to the FDTD simulation requirements.

Figure 5 is obtained as a result of the simulation with the above conditions. It represents the impulsive responses of particle velocity at normal direction, which has been stored at the points representing the virtual positions of microphones. Since the potentiality of the presented method respect to geometrical methods is the direct obtaining of the normal velocity, this is the variable to be discussed. In addition, wave fronts and their time arrival can be easily observed, in the same manner as pressure representations.

Apart from the time arrival of the first wave front, which is proportional to the distance between emitter and receiver, the homogeneity and diffusion of the secondary wave fronts can be observed. For the cube, since all surrounding materials presents the same absorption properties, there are strong secondary fronts arriving at the observation point, which would ruin a speech or musical program. With respect to the cinema, the fact that side and rear walls were covered with a high absorptive material may have result in a dead hall. This can be observed in the absence of sound field except for some defined wavefronts. Finally, for the auditorium, a more convenient field arises as a consequence of a proper distribution of absorbing materials and geometry. Note the first wave front preceding by some minor reflections that enrich the harmonic components of the sound program. The diffusion is clearly evident from 80 ms, which would be perceived as a spaciousness sensation. These results show close similarities with real measurements given by the literature [18].

5. CONCLUSIONS

In this paper, a discrete-time modeling auralization for WFS applications, which is an alternative to auralizate with real measurements, is verified through a set of simulation scenarios or virtual rooms. To perform this model, plane waves and their traveling direction information are required. Since the process needs to obtain both pressure and particle velocity, geometrical approaches cannot provide enough information to be implemented in a direct way. Alternatively, numerical methods based on time-discrete domain provide a powerful tool for this purpose.

Among time-discrete methods, FDTD has been chosen because of the complete sound field description of both pressure and particle velocity. In addition, frequency-dependent boundary conditions are also included to generate real absorbing conditions. Under these conditions, three generic environments have been simulated, where a virtual microphone array following the characteristics of a real set-up has been used to record both pressure and normal particle velocity. Results have shown a good agreement between the wave fronts distribution and what was expected by theory. The diffuseness of the sound field was evident when a proper acoustic design was achieved. Overall, FDTD has been demonstrated as an interesting tool for auralization purposes in a WFS reproduction system, as shown by the virtual rooms behaviour.

6. ACKNOWLEDGEMENTS

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