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An Efficient Auralization of Edge Diffraction

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ABSTRACT

Principles and implementation of efficient auralization of edge diffraction is presented. A calculation principle for the impulse response from an edge is reviewed. This technique is integrated into an acoustic modeling system which is based on the image source method. For auralization purposes a low-order digital filter for each diffracting edge is designed which implements the diffraction phenomenon efficiently and is suitable for parametric auralization. Finally, examples of filter designs are presented.

INTRODUCTION

This paper contributes to the modeling and auralization of diffraction which is one of the most important acoustic phenomenon caused by the wave nature of sound. However, most room acoustic modeling algorithms rely on geometrical acoustic techniques which completely neglect the diffraction. The reason for this is that such methods as ray-tracing [1], beam-tracing [2] and image-source method [3, 4] assume that sound propagates as light, i.e., the wavelength of the sound is much shorter than the dimensions of the reflecting objects. In this paper we show how diffracted sound components can be added to the image-source method and how an efficient auralization of diffracted components can be realized.

Related Work

The edge diffraction phenomenon has been modeled mainly in environmental acoustics when studying noise barriers [5]. In this article our main application area is room acoustic modeling and especially auralization. From the viewpoint of room acoustics only a few studies related to diffraction modeling and auralization have been carried out.

Torres et al. [6, 7, 8] have been modeling diffraction with the same method as presented in this article. They have showed that the applied time domain diffraction modeling together with the image-source method gives almost exactly the same room impulse response in a simple room geometry than can be obtained with scale model measurements. They found out that first order diffracted components were clearly audible, even when the sound source was not in shadow.

Sakurai and Nagata [9] have modeled diffraction in room acoustics using the less accurate Kirchoff approximation. Also Ouis [10] has studied a simple room geometry with one diffracting barrier inside it. However, neither of these studies did not consider auralization despite the fact that their application area is concert hall acoustics.

Tsingos et al. [11, 12, 13] have been studying diffraction with Fresnel-Kirchoff approximation and with Uniform Theory of Diffraction. Their frequency domain solution was to add diffraction to a beam-tracing method. The main interests in their studies have been in modeling and auralizing the diffraction in the shadow zone, i.e., in case of occluded sound source.

Martens et al. [14, 15] have measured and studied the audibility of one occluder with various rendering cases. They also propose simple filter structure to be used in auralization with such an occluder. However, their results do not give any generalizable model for occlusion because their filter structures were not based on accurate diffraction models. Similar approach has been used in the game industry where some guidelines for occlusion and obstruction models have been proposed [16]. Such perceptually relevant, but not accurate, approach are efficient and certainly useful in computer games to realize the overall lowpass effect caused by diffraction. However, for precise room acoustic modeling and auralization more accurate diffraction models are needed.

MODELING OF EDGE DIFFRACTION

The principles of modeling the diffraction impulse response from an infinite edge has been presented already a long time ago by Biot and Tolstoy [17]. Recently, Svensson et al. [18] presented a mathematical solution for how to calculate the impulse response for an edge of a finite length, which is the case in closed spaces such as rooms. This new model yields exactly the correct solution for a rigid finite wedge and it is directly applicable also to curved edges and multiple diffraction. For non-rigid edges no exactly correct method is known. Nevertheless, when the absorption factor of an edge material is low, the presented model could still be reasonable.

The derivation of the impulse response for a finite edge is based on the argument that the local reaction at the edge to an impulsive incident wave is instantaneous [5]. The detailed mathematical derivation [18] of the solution is out of the scope of this article. It leads to the line integral, presented in Eqs. (1) and (2) for diffraction impulse response. The variables can be found in Figs. 1 and 2 where a finite wedge is illustrated. In addition, in Eq. (1) c is the speed of sound, $\nu = \pi/\theta_w$ is the wedge index, m is the "source-to-edge point" distance, and l is the "edge point-to-receiver" distance. The integration range is between the two end points of the finite edge. The line integral shows an integration of impulsive contributions which are delayed by the sound path length $m+l$ for any point z along the edge. Spherical spreading from the source to the edge point, and from the edge point to the receiver, is indicated by the factor $1/(ml)$. The distances m and l are not involved in the β-terms and accordingly, the sum of the β-terms can be seen as a directivity function.

The presented algorithm has been implemented as a free Matlab toolbox [19]. It contains also functions for calculating multiple diffractions. The basic diffraction from one edge (Figs. 1 and 2) is calculated as follows. First, the Cartesian coordinates of the source S, the receiver R, and the end points of the edge z_1 and z_2 are translated to cylindrical coordinates. To simplify computation the edge is moved to the origin thus the starting point of the integral z_1 is 0 and the end point z_2 is z. An impulse response of the diffracting edge is calculated by dividing the edge into the small pieces and calculating the contribution of all these pieces according to the Eqs. (1) and (2).

Efficient methods and algorithms for finding edges (together with image sources) are already published earlier by Pulkki et al. [20]. In this solution, edges are considered as diffraction image sources. The procedure for finding edges and creating diffraction image sources consist of three stages. First, a diffractive edge is searched by checking all polygon pairs that are connected with each other. Then this edge is mirrored with all the surfaces of the model and finally the edge is replaced with a point-like image source and a visibility check for this point is performed. The point-like diffraction image source is considered to emit an impulse response which is calculated as explained above. This approximation neglects the fact that the impulse response is not really emitted from one point, it comes from all the points along the edge. However, the point-like treatment is necessary so that the following

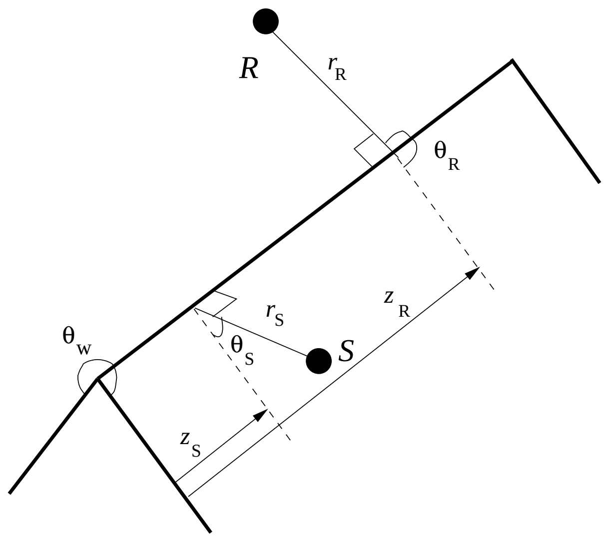

Fig. 1: Geometry of a finite wedge. The positions of the source S and the receiver R are indicated in cylindrical coordinates.

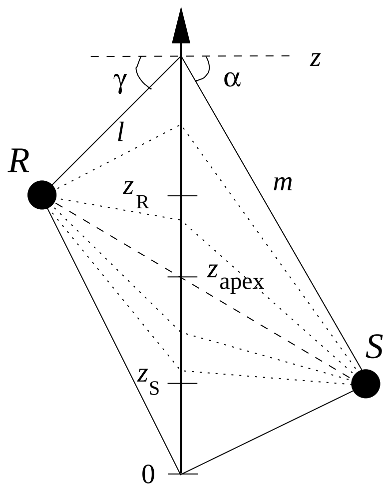

Fig. 2: Top view of the same wedge than in Fig. 1. Sound paths via wedge end points 0 and z are indicated by the solid line, the least-time sound path via the apex point z_{apex} is depicted with dashed line and some other sound paths are illustrated with dotted lines.

$$h_{diffr}(t) = -\frac{\nu}{4\pi}\int_{z_1}^{z_2}\delta\left(t-\frac{m+l}{c}\right)\frac{\beta_{++}+\beta_{+-}+\beta_{-+}+\beta_{--}}{ml}dz \tag{1}$$

$$\beta_{\pm} = \frac{\sin[\nu(\pi\pm\theta_S\pm\theta_R)]}{\cosh\left(\nu\cosh^{-1}\frac{1+\sin\alpha\sin\gamma}{\cos\alpha\cos\gamma}\right)-\cos[\nu(\pi\pm\theta_S\pm\theta_R)]} \tag{2}$$

auralization method is applicable.

AN EFFICIENT AURALIZATION METHOD

In the parametric room impulse response rendering [21] all acoustical phenomena, e.g, sound source directivity and air absorption, are implemented with digital filters. This approach enables flexible auralization of direct sound and early reflections calculated with the image-source method. In Fig. 3 the example signal processing chains for one reflection and one diffracted sound component are depicted. The following phenomena are realized for each image source: distance delay and attenuation, sound source directivity, material and air absorption, and binaural filtering. In addition, for one diffraction image source, a diffraction filter $Diffr_M(z)$ is applied and this filter implements the impulse response defined with Eqs. (1) and (2). In this section we will present how such a diffraction filter is designed and how it is applied to both static and dynamic auralization.

The calculated response of an edge is often quite long, depending on the length of the edge. In a typical concert hall case it can be several milliseconds long (thousands of samples at 48 kHz sampling rate), and its direct use for parametric auralization with a FIR filter is computationally too demanding due to the large amount of diffracted components. However, the effect of the diffraction is in most cases lowpass filtering and the frequency response is quite smooth. This leads to the fact that a diffracted response can be implemented efficiently with a low-order recursive filter for auralization purposes.

Many different filter design methods that can be applied to the diffraction filter design problem exist. The implementation presented in this paper is not the only correct one but we have found it accurate enough, robust, and suitable for the purpose. The design procedure is as follows. After the calculation of the exact edge response, the response is multiplied with the least-time distance $m+l$ due to the fact that distance attenuation (according to $1/r$-law) is implemented separately (see Fig. 3). The sign of the diffraction peak is checked and saved for later use. The actual filter design is implemented in frequency domain and therefore FFT is applied for translating diffraction response to the frequency domain. The magnitude response is then smoothed using ERB scale resolution. The actual filter fitting can be done with many filter design methods. After trying out several different methods we chose the Yule-Walker method (function `yulewalk` in Matlab) and the filter fitting to magnitude response data was completed in warped frequency domain. The warped frequency scale used is an approximation of the Bark scale and the warping was done using the conformal bilinear transform [22]. Finally, the designed filter coefficients were stored for auralization process in which warped IIR (WIIR) filters of order 3 have been found good and accurate enough.

The use of warped filters can be argued in many ways but we have found them practical for diffraction modeling. In a modeling case of a typical room geometry thousands of diffracted components exist and an applied filter design method has to provide stable filters for all of them. Another important motive to apply warped filters is that they provide good accuracy at low frequencies where the major part of the diffracted energy is.

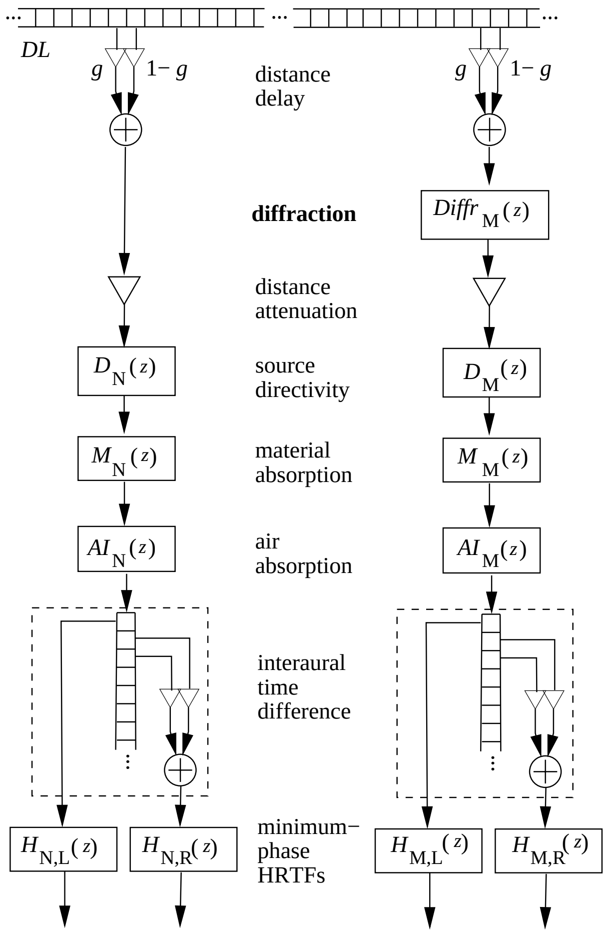

Fig. 3: The signal processing structure for one image source and one diffraction image source. Filter $Diffr_M(z)$ implements the diffraction from an edge. The pick-up point from the delay line *DL* is chosen according to the distance from the source to the listener position. On the left side the subindex *N* is the image source number and on the right side subindex *M* is the number of diffraction image source.

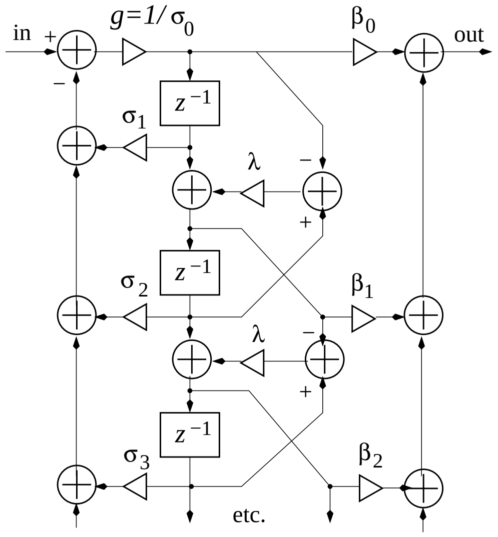

Fig. 4: A realizable WIIR structure [24].

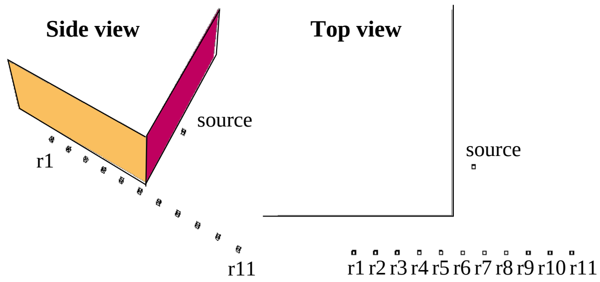

Fig. 5: Example case study: a street corner.

The Implementation of Warped IIR Filters in Auralization
The implementation of WIIR filters is more complex than implementation of normal IIRs. A WIIR filter requires 2 to 2.5 times more computational power that an IIR filter [23]. This means that the applied WIIR filter of order 3 corresponds roughly to an IIR filter of order 7. In some cases warped filters could be implemented with normal IIRs if filter coefficients are unwarped with the inverse bilinear transform [22]. However, unwarping sometimes leads to numerical failures which cannot be avoided despite of the use of double-precision floating point numbers.

The basic idea in implementation of WIIR filter is that all unit delays of an IIR filter are changed to first-order allpass sections. After some rearrangements a realizable form of WIIR filter is obtained [24] and such a solution is depicted in Fig. 4. In this realization the recursive filter coefficients (usually marked with α_i when an IIR is considered) are mapped to coefficients σ_i. An algorithm for such a mapping and its theoretical derivation is presented by Karjalainen et al. [25].

The use of warped filters in auralization of edge diffraction is straightforward once the filter design has been performed. The designed filter coefficients are delivered to the auralization process and they are directly usable. The output of the warped filter is finally multiplied with a stored sign of diffraction impulse response, so that the sign of the diffracted impulse response is correct.

The parametric auralization method enables dynamic sound rendering in which the sound sources or the listener can move. In such a dynamic situation diffraction responses change and new filter coefficients have to be calculated for every single diffraction image source. The update of coefficients of recursive filters $Diffr_M(z)$ have to be done carefully if smooth and continuous output signal is required. In our solution, a smooth output signal is obtained by using two filters for one diffraction image source and cross fading their outputs between updates. The update rate in dynamic rendering is typically from 20 to 50 Hz.

Such dynamic rendering can be calculated as batch jobs, when diffraction filters can be designed at each time stamp beforehand. However, for interactive use presented filter design approach is not practical, because the calculation of diffraction impulse response and filter design are computationally laborious.

The other remark relates to the usage of point-like diffraction image source in parametric auralization. As explained in this paper the diffraction filter implements the diffraction phenomenon in time and frequency, but in real life diffracted sound is not point-like. Sound passes the edge through all points along the edge, however, the most of the energy is concentrated to the least-time point of the edge (z_{apex} in Fig. 2). Based on this we could assume that the simplification to apply point-like secondary source is not so severe. In addition, the diffraction image source, being a point source, can be panned to the direction where the least-time point indicates as proposed by Torres et al. [8]. The same principle holds for the sound source directivity, since from the viewpoint of the edge most of the sound energy from the actual source radiates towards the least-time point of the edge. The situations where this simplification could be most audible would be long horizontal edges that are close to the listener.

Filter Design Examples
To clarify the filter design process, one example case of a simple street corner geometry is presented. We calculated impulse responses from one source position to the eleven receiver positions as illustrated in Fig. 5. For simplicity both the sound source and the receivers had omnidirectional characteristics and the air and material absorption was not applied. In other words, only distance attenuation and delay was modeled, in addition to the diffraction modeling. As can be seen in Fig. 5 the sound source is hidden by the corner, i.e., the direct sound is not visible, to the receivers r1-r4. The receivers r5 and r6 capture direct sound and diffracted component, while receivers r7-r11 have also specular reflection from the wall near the sound source. All eleven simulated impulse responses are depicted in Fig. 6.

The filter design is realized for diffracted components. Impulse responses of analytical model and designed filters are compared in Fig. 7. It can be seen that responses of the filters are very close to targets as only small differences can be seen in responses. A frequency domain comparison is more relevant and magnitude responses of all diffracted components and designed filters are depicted in Fig. 8. It can be seen that between 100 and 10,000 Hz the filters fit well to the target responses. The low frequency weighting in design is also clearly seen. This causes inaccuracy at high frequencies but the design is acceptable as high frequencies are perceptually less relevant. The oscillations of the target responses are caused by the length of the finite edge.

Although, the designed filters fit well to target responses both in time and frequency domain the final verification of their accuracy should be done with listening tests.

APPLICATIONS
Commercially available room acoustic prediction softwares such as

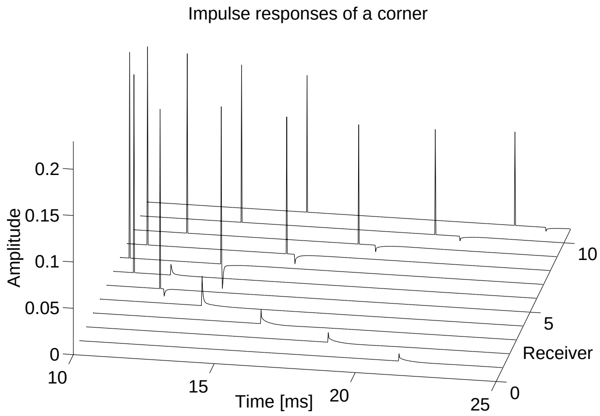

Fig. 6: Impulse responses from the sound source to the eleven receiver points r1-r11 (see Fig. 5).

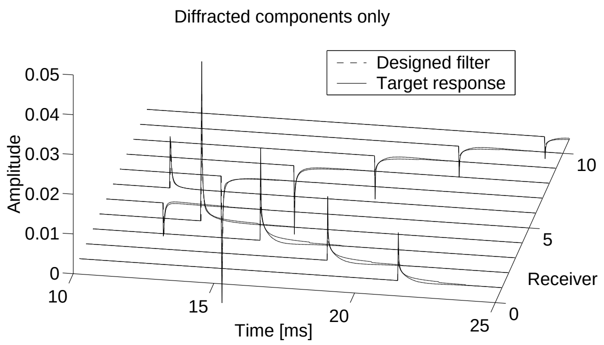

Fig. 7: Impulse responses of diffracted components. Target responses are calculated with Eqs. (1) and (2) and filters designed are WIIRs of order 3.

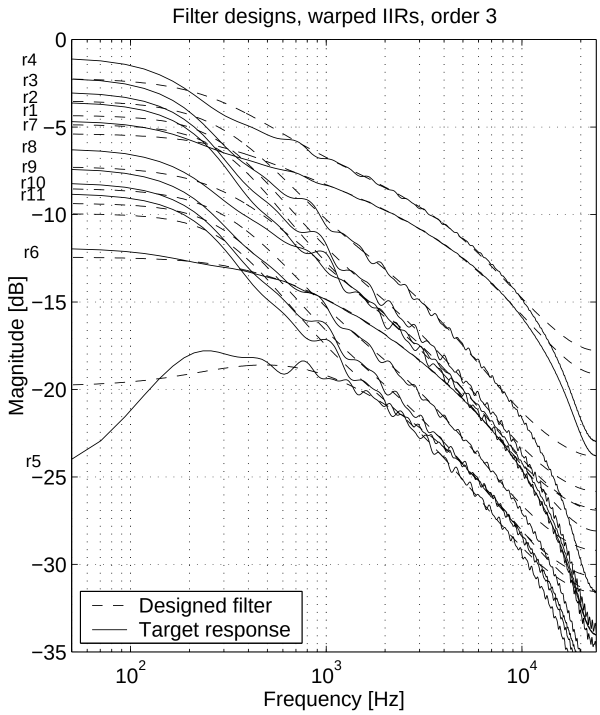

Fig. 8: Frequency responses of diffracted components and designed filters. The receiver points are indicated with r1-r11 on the left. The responses are scaled so that 0 dB corresponds to the level of the direct sound in the receiver positions.

Odeon [26], CATT Acoustics [27], and Ease [28] are based on geometry based modeling methods. Currently, these programs neglect diffraction despite the fact that it is very important in such cases as opera halls, in which the sound sources (the whole orchestra) are not visible to seats on the main floor. All of these programs also have auralization modules, which do the auralization by convolving the simulated binaural impulse responses with anechoic sound signal. Despite the slightly different approach to auralization, the presented efficient auralization of edge diffraction could be applied to these softwares. Although, the presented model is only exact for rigid edges, it could be used for edges (together with image source model that includes absorption) formed by two surfaces with low absorption coefficients.

The other important application area for efficient diffraction modeling and auralization is computer games. In recent sound rendering algorithms for games some simplified diffraction models are implemented [16], but in the future more accurate models could be used. The correct diffraction model is especially important for direct sound when the sound source is occluded.

CONCLUSIONS AND FUTURE WORK

In this paper we have presented an efficient way to implement edge diffraction in parametric auralization. The diffraction model is based on an analytical time domain solution. In the auralization a diffraction is handled with a low-order WIIR filter which is fitted to the analytic diffracted response. The accuracy of the filter design has been demonstrated with examples and filters seem to fit well both in time and frequency domain to the analytical responses.

The presented implementation is not suitable for interactive use, since the calculation of analytical diffracted response and the filter fitting are computationally laborious. However, some other mapping directly from a geometry (positions of source, receiver, and edge) to diffraction filter coefficients is needed. One idea is to use a neural net based solution since a lot of training data can be generated with the method presented in this paper. In addition, applied filter implementation could be optimized computationally, e.g., with multirate techniques, since the major effect of diffraction is at low frequencies.

The other thing to be studied is the assumption that the whole edge response can be implemented with one point-like secondary source. This simplification can be bypassed by cutting the edge into smaller pieces and replacing the edge with a set of point sources along the edge. Naturally, this solution raises the amount of diffraction image sources. Further studies should be done to find out perceptually optimized balance between computational load and the amount of diffraction image sources for one long edge.

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