DOI: 10.61782/fa.2025.0729

FORUM ACUSTICUM EURONOISE 2025

REAL-TIME VIRTUAL ENVIRONMENT AND ROOM ACOUSTICS SIMULATOR

Kevin de Haas[1] **Michael Schutte**[1] **Stephan D. Ewert**[1*]

[1] Medizinische Physik and Cluster of Excellence "Hearing4all", Carl von Ossietzky Universität Oldenburg, Germany

ABSTRACT

In our daily life, sound plays an important role for awareness of our environment, communication, and recreation. The ability to predict, simulate and acoustically render real-world acoustics in a computationally efficient way is of key importance for various applications ranging from architectural acoustics to hearing research. Historically, acoustic simulation and artificial reverberation have been developed in seemingly remote areas, either using offline processing with focus on accuracy, or using real-time processing with focus on computational efficiency. Our geometrical acoustics based approach, the Low-latency Interactive Virtual Environment and Room Acoustics Simulator (liveRAZR) aims to bridge both application areas. To account for edge diffraction and sound propagation through finite openings, we combine a geometrical diffraction pathfinder with our universal diffraction filter approximation. Diffuse late reverberation is simulated using a spatially rendered feedback delay network. Sound scattering and a realistic increase of echo density are simulated by distributing scattered energy from each reflection to the diffuse late reverberation module. The proposed model consists of a time-domain implementation for 6-degrees-of-freedom motion of source and receiver based on variable delay lines and an accompanying offline version. We discuss several use cases, ranging from echolocation to augmented acoustic reality for visually impaired.

Keywords: *Virtual acoustics, Geometrical acoustics, Digital filters, Virtual reality, Diffraction*

[*]***Corresponding author***: *stephan.ewert@uol.de.*

1. INTRODUCTION

Acoustic modelling and simulation of environments is concerned with predicting sound propagation and creating an audible experience of existing, planned, or fictional spaces. Applications exist in a variety of areas, ranging from architecture and civil engineering, e.g., [1] to movies, video games and virtual/augmented/extended reality [2], [3]. Models to predict acoustics and sound propagation in the context of architectural planning and noise control typically operate offline, while real-time simulation with interactive update or sound sources and receivers are mandatory for auralization of virtual environments. Historically, at least three different and seemingly remote areas can be identified for acoustic models, each with specific aims and limitations: i) Room acoustics models can reach high accuracy of the simulation, e.g., [4]. They provide impulse responses to predict room acoustic parameters, with auralizations mainly limited to static precomputed conditions. ii) Predictions for sound propagation and protection in traffic, urban, and outdoor environments also operate offline and are partly based on empirically motivated models, e.g., [5]. iii) Real-time virtual acoustics has evolved using concepts either from room acoustics simulations combined with fast convolution of dynamically updated impulse responses, e.g., [6], [7] or using time-domain processing, e.g., [8] often based on perceptually motivated concepts, originally suggested for artificial reverberation in the context of digital audio and music effects. Recently, several approaches aim to incorporate physics-based principles of highly detailed acoustic simulations with real-time suited signal processing from artificial reverberation.

Hereby, geometrical acoustics (GA) has evolved as a de-facto standard, which approximates the propagation of sound as being represented by rays. While neglecting wave properties which play a role at low frequencies where the wavelength is large against the dimensions of geometrical structures, GA offers considerable advantages regarding computational efficiency. In GA, early sound reflections can be efficiently simulated using the concept of image sources [9], [10]. Because computational costs increase fast with reflection order and number of surfaces in the environment, alternative approaches are typically employed for late reverberation, covering the spatial properties and spectro-temporal evolution of the reverberant tail. One option is to use ray tracing [11] to retrieve the energetic decay of reverberation in the environment, which can then be used to spectro-temporally modulate a noise process [12] in order to gain a (binaural) room impulse response, (B)RIR, see, e.g., [7]. The so predicted BRIR is typically applied to the source signal using fast and/or sparse convolution techniques, yet with computational requirements increasing with the duration of the BRIR.

Alternatively, the image source model (ISM) and late reverberation can be applied in the time domain using variable delay lines and recursive filters as well as recursively connected delay lines to model late reverberation. Extending the classical Schroeder reverberator [13], feedback delay networks (FDNs) [14], [15] are most widely used. Several recent works [16]–[19] aimed to connect physical room properties with recursive filter networks for modelling room acoustics, including multi-slope decays. Recent hybrid approaches connecting an ISM and FDN were suggested in [20]–[22] following the influential work [8] for creating interactive virtual acoustic environments. One advantage of time-domain models using variable delay lines (VDLs) is their potential of low-latency processing as well as mimicking effects of Doppler shift relevant in outdoor scenarios with fast moving sound sources.

One critical limitation for straight-forward application of GA in interactive applications is that “visibility” of the sound or image source can suddenly change when the line-of-sight or specular reflection path disappears. This is particularly critical in outdoor scenarios, where sound reflections are typically sparse and late reverberation, potentially masking perceptual effects of such discontinuities, has a relatively low level. To avoid such discontinuities and resulting inaudible sound sources behind obstacles such as corners and buildings, edge diffraction [23]–[26] can be integrated in GA. While efficient approximations have been limited to infinite edges [27], [28], recently a low-order recursive filter approximation of finite and infinite edges has been suggested [29], [30]. With focus on plausible diffraction effects in interactive sound propagation, [31] suggested precomputed diffraction kernels for more complex shaped dynamic objects in the scene. Besides simulating effects of diffraction, the determination of high-order diffraction propagation paths in scenes with complex geometry poses a significant bottleneck in real-time simulation of diffraction. Optimizations with edge visibility preprocessing [32] or mesh preprocessing to extract a reduced subset of “silhouette” diffraction edges [33] have been suggested for fast diffraction pathfinding.

Another limitation of straight-forward application of GA within the ISM is the restriction to specular reflections. However, in real environments, an increasing part of acoustic energy is diffusively reflected for higher reflection orders. Approaches to include non-specular, scattered reflections in the ISM have been suggested, e.g., [34], with [35] providing a filter-based implementation suited for real-time processing.

Taken together, a number of approaches exist for interactive or real-time suited acoustic models, focusing on specific aspects such as multi-slope decays, diffracted paths, or scattered reflections. No real-time model with consistently physics-based as well as perceptually evaluated processing steps exists.

Here, we suggest the Low-latency Interactive Virtual Environment and Room Acoustics Simulator (liveRAZR). The acoustic model of liveRAZR is based on technically and perceptually evaluated processing steps [20], [21], [36], [37] which were originally implemented in the offline version RAZR. liveRAZR extends the initial real-time implementation of [38] and the integration of (measured) source directivity filters as described in [39] using the widespread SOFA format [40].

2. GENERAL STRUCTURE AND OPERATION

liveRAZR is a standalone software, written in C++ using the Intel IPP library and JUCE framework. It is optimized for multi-threaded, real-time computation on typical PC hardware running the Windows operating system. liveRAZR is developed and maintained in tandem with the original RAZR implementation [20], [21] for Mathwork’s MATLAB and Octave. liveRAZR can run in two modes of operation:

i) In interactive operation liveRAZR runs as an agent application in parallel to a controller software (e.g., Unreal engine) which sends real-time updates of, e.g., source and receiver positions and rotations (6 degrees of freedom) via Open Sound Control (OSC) messages. Upon startup, liveRAZR is initialized via a configuration (.razr) file which describes the acoustic environment as well as the parameters of the room acoustic simulation. The

configuration file is a human-readable text format according to the libconfig library which was chosen to be self-explanatory and free of clutter. The configuration specifies the geometry model as Wavefront .obj file which is a *de facto* open, widely used, and itself human readable format.
ii) Without OSC control, liveRAZR can run in standalone mode with the sound source and receiver positions or trajectories defined in the .razr file. The output is then written to a waveform audio file (.wav).
The configuration file is typically generated by the accompanying MATLAB/Octave version of RAZR in a so-called augmentation step. During this, all parameters required to run liveRAZR are written to the configuration file. Source sounds are accepted in various file formats including wav and mp3, or alternatively as live audio stream.
At runtime, two main program parts are executed as depicted in Fig. 1, the multi-threaded audio processing and the geometry model, which are described in the following.

2.1 Audio processing

The Audio pipeline runs at the "heartbeat rate" of the system defined by the audio buffer size. The audio player generates continuous audio input for each source from pre-loaded audio files and a live stream (e.g., the microphone input of the sound card). The acoustic model then calculates the spatialized outputs for each source based on the source's input and the paths previously calculated by the geometry model. For a more detailed description of the acoustic model, see Section 3. Finally, the mixer sums up the spatialized outputs allowing to apply additional level adjustments and further equalization for direct sound, early reflections and late reverberation, separately. The audio player and the mixer can be controlled via OSC messages.

2.2 Geometry thread

The trajectory module generates a continuous trajectory for the receiver and each source and provides transforms for further processing. The trajectory module is continuously updated by receiving positions and orientations via OSC messages. The path finder then calculates all sound propagation paths according to the maximum specified reflection and diffraction order for the current source and receiver transforms. To reduce the number of paths, only diffractions into the shadow zone and only a single path per image source (the shortest path) are taken into account. The geometry model can be run synchronously with the audio processing to allow low-latency geometry updates for simple geometries like a shoebox, or asynchronously at a defined rate of multiple integers of the audio processing heartbeat to allow more complex geometries without blocking the audio processing. If run asynchronously, the paths are being interpolated for every audio block processed by the acoustic model.

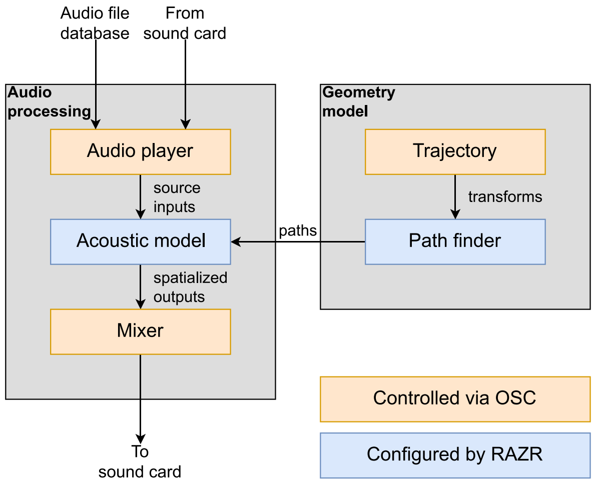

Figure 1. General structure of liveRAZR with the two main modules, the audio processing (left), and the geometry model (right). Processing blocks within each module are controlled interactively via OSC messages (orange). The acoustic model and the path finder are additionally configured at startup (blue).

3. ACOUSTIC MODEL

Figure 2 shows the block diagram of the acoustic simulation model implemented in liveRAZR. For each sound source, the associated audio signal from the Audio player is transformed by the acoustic model to an output signal either rendered binaurally to headphones or to a multi-channel output for loudspeaker playback in the Spatialization stage. The acoustic model consists of two parts, i) the geometric acoustic path (blue) which is calculated for the direct sound and each image source, including diffraction as calculated in the Path finder of the Geometry model; ii) the late reverberation model (red) which consists of a spatially rendered and room-informed FDN. For spatial rendering, the Spatialization stage either uses a parametric extended spherical head model, SHM [41], or measured head related impulse responses (in SOFA format, [40]) for binaural headphone output. Vector base amplitude panning, VBAP [42], is used for multi-channel loudspeaker rendering.

3.1 Direct sound and early reflections

For each sound path, the propagation delay is implemented using a variable delay line with truncated-sinc-interpolated read out for fractional delays. Source directivity is either based on the parametric extended SHM filter or on measured data, specified in the SOFA format. An air absorption filter and distance attenuation account for free field sound propagation. Object scattering implements temporal diffusion as consequence of small scale geometric objects along and near the propagation path which are not explicitly modelled in the geometric representation of the environment, see [35]. Mimicking the effect of multiple short-throw reflections and scattering, temporal diffusion with a Gamma-function-shaped envelope is realized using a cascade of four allpass filters. A cascade of reflection and diffraction filters is applied depending on the specific acoustic path. Reflection filters are designed as series of second-order section filters based on the frequency-dependent absorption coefficient, typically specified at octave frequencies. The diffraction filters are a series of two shelving filters to mimic a fractional half-order lowpass according to [30]. For surface scattering, depending on the random-incidence scattering coefficient [43], a specular-diffuse decomposition filter is applied [35]. The specular reflection part is fed to the spatialization stage and is auralized as early specular reflection. The diffusely reflected energetic part is fed into the late reverberation model. For the highest simulated reflection order, both the specular and diffuse part are passed to the late reverberation model.

3.2 Late reverberation

The channel mapper assigns the diffuse reflection to those FDN channels associated with its spatial direction. The input is energetically split using the dot product of the reflection direction and the direction to which the FDN output is spatially assigned as virtual reverb source (VRS; see [21], [35]). The following scattering filter accounts for the temporal diffusion of the scattered reflection implemented as cascade of four allpass filters with an exponential decaying envelope (see [35]). The FDN mimics scattered sound reflections in the surrounding volume or environment. Typically, 12 or 24 FDN channels are used. The delay lines of the FDN are associated with the dimensions of the room or environment and the absorption filters in the FDN are adjusted to obtain a prescribed frequency-dependent reverberation time (see [14]). Each output of the FDN is mapped to a VRS with an associated direction with respect to the receiver. The output is passed through the reflection filter associated with the boundary in

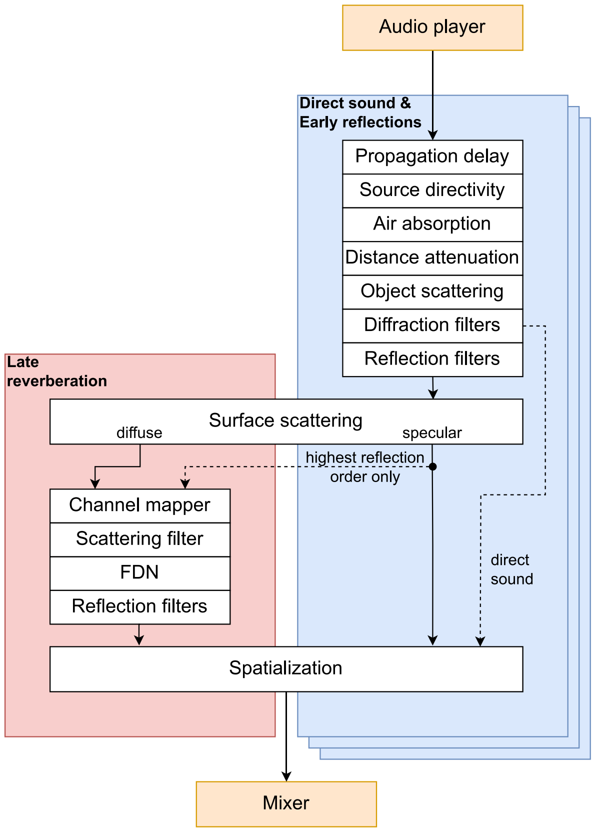

Figure 2. Block diagram of the hybrid acoustic model implemented in liveRAZR. The direct sound and early reflections (blue) are modelled for each reflected and diffracted path found by the path finder. For each reflection, scattered sound energy is passed to the late reverberation model (red) from the output of the specular-diffuse decomposition filter.

the respective VRS direction (see [21]). Finally, the VRSs are spatialized in the same way as the specular reflections.

4. PERFORMANCE EVALUATION

To analyze the performance of liveRAZR, three rooms have been chosen for evaluation: a simple shoebox room (SB) with 6 surfaces and no edges, a U-shaped room with 10 surfaces and 2 edges and a more complex "maze" environment with 30 surfaces and 18 edges, referred to as room 42. The rooms and the respective paths calculated by the path finder are depicted in a top-down view in Fig. 3.

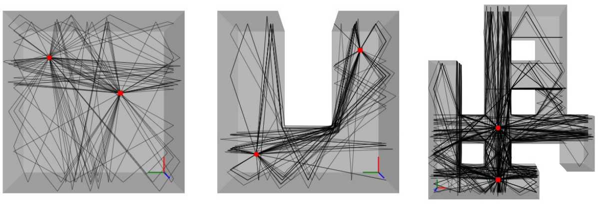

Figure 3. Top-down view of the three environments used for the performance evaluation: Shoebox, U-room and room 42, from left to right. Source and receiver positions are indicated in red, the resulting reflected and diffracted paths are shown in black.

For the evaluation a sample rate of 44.1 kHz was used. The maximum reflection order was set to 3 and the maximum diffraction order was set to 2. The path finder was run single-threaded and the acoustic model was run multi-threaded with 8 concurrent threads used for processing. For all simulations fractional delay was used for the variable delay lines. Simulations were performed on a standard desktop PC (Windows 10) with Intel i9-9900k processor and 32 GB of RAM.

4.1 Acoustic model

To test the performance of the acoustic model, the three rooms have been simulated with a varying number of sources. For headphone rendering, the parametric extended SHM and an HRTF with a filter length of 512 and a spatial resolution of 1 degree with respect to azimuth and elevation were used. Two alternative audio buffer sizes were tested: 256 samples corresponding to 5.8 ms and 64 samples corresponding to 1.45 ms at a sample rate of 44.1 kHz. All simulations were run with the scattering filters (object and surface scattering) enabled and disabled.

The results are shown in the upper two panels of Fig. 4. As can be seen the execution time linearly depends on the number of paths to be processed. The workload is calculated as the average execution time per block divided by the block duration. In Table 1 the maximum number of sound sources that can be calculated with a workload of 50 % are listed. In a shoebox room, the number of paths per source is constant at 63 for up to 3rd-order image sources (no diffraction). For the rooms with diffraction, the number of paths depends on the positions of the sources and receiver. As a typical example, a number of 79 paths for the U-shaped room and 292 paths in room 42 are used for the calculation. These numbers correspond to the source and receiver positions as well as the paths depicted in Fig. 3.

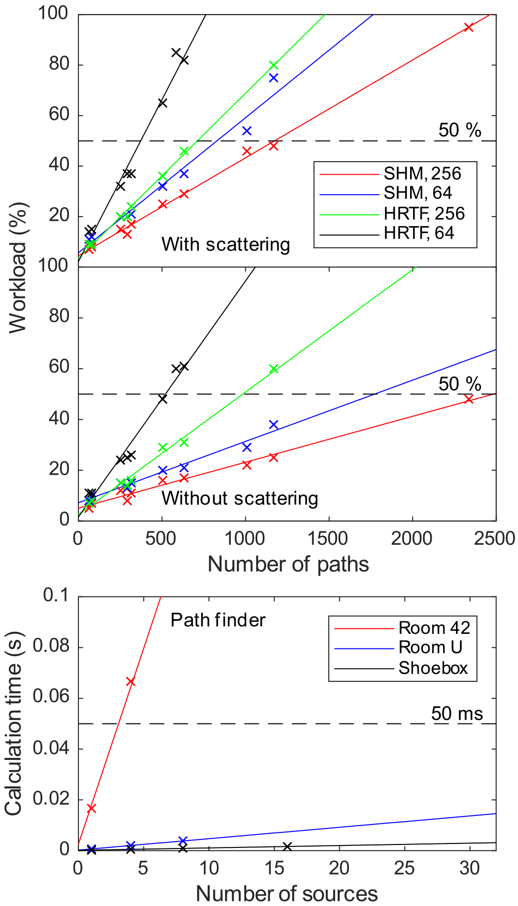

Figure 4. Performance evaluation. For the acoustic model (upper two panels) the workload is shown as a function of the number of acoustic paths (including reflections and diffraction). Red and blue are results for the extended SHM with 256 and 64 sample frames, respectively. Green and black are for the HRTF. The lower panel shows the calculation time of the geometric model (path finder) as a function of the number of sound sources for the three rooms.

4.2 Geometry model

To analyze the performance, the average execution time of the geometry model has been measured. The results are shown in the lower panel of Fig. 4. Also here a linear dependence of execution time on the number of sources can be observed. In Table 2, the calculation time of the path finder per sound source and maximum number of sources are listed that can be calculated within 50 ms,

Table 1. Maximum number of paths and according number of sound sources in the three rooms for a workload of 50 %.

Spatial -izer	Block size	Scatter -ing	Paths @ 50 %	Max. sources		
				SB	U	42
SHM	256	no	2500	39	31	8
		yes	1180	18	14	4
	64	no	1790	28	22	6
		yes	830	13	10	2
HRTF	256	no	980	15	12	3
		yes	710	11	8	2
	64	no	520	8	6	1
		yes	370	5	4	1

corresponding to an update rate of 20 Hz. It is obvious that the computational effort in the path finder significantly increases with the number of edges in the environment.

5. DISCUSSION AND USE CASES

For interactive use, liveRAZR runs as an agent application receiving real-time updates from a controller application via OSC messages. Thus users can work with a controller application of their choice. So far, we have used computer game engines for this purpose such as Unreal Engine and Godot, handling head and position tracking as well as creating the visual environment and control elements. Readily available integration of virtual reality head-mounted displays in these game engines makes real-time head tracking easily available, e.g., using SteamVR.

The inclusion of edge diffraction in liveRAZR enables realistic interactive acoustics in conditions without line-of-sight to the sound source. This is particular important for acoustics-based indoor navigation using corridors and mazes as investigated in [44], [45] to test prototype acoustically augmented reality (AAR) methods for orientation and navigation of blind people. Moreover, it also enables realistic outdoor acoustics, e.g., at a city street corner with surrounding buildings.

The current performance evaluation shows that the bottleneck is the path finder in the geometry model (see also [32], [33]). The number of potential diffracted paths increases with the number of edges in the environment and the maximum allowed reflection and diffraction order (here 3 and 2, respectively). In future updates, the performance of the current geometry model can be easily improved using multi-threading. Moreover, several other optimizations for

Table 2. Calculation time of the geometry model (path finder) per sound source and maximum number of sources for a geometry update rate of 20 Hz.

Room	Sur-faces	Edges	Paths per source	Time per source	Max. sources @ 20 Hz
SB	6	0	63	0.1 ms	500
U	10	2	79	0.5 ms	100
42	30	18	292	16 ms	3.125

geometry processing known from computer graphics have also not yet been utilized to their full potential.

The low-latency, time-domain implementation of liveRAZR, here tested with audio frame sizes down to 64 samples, enables simulation of the acoustic environment in response to self-produced sounds. In [45] liveRAZR was used to investigate human echolocation. The own vocalizations were picked up using a near-field microphone. Using open headphones, simulated sound reflections were continuously played back, in addition to the real direct sound propagating from the mouth to the ears. In such applications, the nearest wall distance which can be simulated and thus the temporal lag of the earliest reflection is determined by the latency of the system, which is 1.45 ms for 64 sample frames and a sampling rate of 44.1 kHz. Including additional delays introduced by the analog-to-digital and digital-to-analog conversions and buffering, with a resulting overall latency of 128 samples, reflections from a nearby wall as close as 50 cm can be simulated.

6. SUMMARY AND CONCLUSION

The suggested Low-latency Interactive Virtual Environment and Room Acoustics Simulator implements a technically and perceptually well evaluated acoustic model [20], [21] using VDLs, recursive filters and partitioned convolution for measured receiver and source directivity filters. The physics-based edge diffraction model and the high-performance diffracted path finder enable 3 reflection and 2 diffraction orders in the most complex here tested environment at interactive update rates. In future work, we plan to implement late reverberation for coupled volumes and for outdoor acoustics.

Currently, liveRAZR is freely available upon request from the authors with a public release planned for the near future. RAZR (offline) for MATLAB/Octave is freely available at www.razrengine.com.

7. ACKNOWLEDGMENTS

This work was supported by the Deutsche Forschungsgemeinschaft, DFG SPP Audictive – Project-ID 444827755 and Project No. 406277157 "Acoustical Awareness, Orientation, and Navigation in Rooms."

8. REFERENCES

[1] J. H. Rindel, "The Use of Computer Modeling in Room Acoustics," *J. Vibroeng.*, pp. 219–224, 2000.

[2] N. Raghuvanshi and J. Snyder, "Parametric directional coding for precomputed sound propagation," *ACM Trans. Graph.*, vol. 37, no. 4, pp. 1–14, 2018.

[3] V. Valimaki, J. D. Parker, L. Savioja, J. O. Smith, and J. S. Abel, "Fifty Years of Artificial Reverberation," *IEEE Trans. Audio Speech Lang. Process.*, vol. 20, no. 5, pp. 1421–1448, 2012.

[4] G. M. Naylor, "ODEON—Another hybrid room acoustical model," *Applied Acoustics*, vol. 38, no. 2–4, pp. 131–143, 1993.

[5] "ISO-9613-2 Acoustics — Attenuation of sound during propagation outdoors — Part 2: Engineering method for the prediction of sound pressure levels outdoors." 2024.

[6] B. U. Seeber and S. W. Clapp, "Interactive simulation and free-field auralization of acoustic space with the rtSOFE," *J. Acoust. Soc. Am.*, vol. 141, no. 5_Supplement, pp. 3974–3974, 2017.

[7] D. Schröder and M. Vorländer, "RAVEN: A real-time framework for the auralization of interactive virtual environments," presented at the Forum Acusticum, Aalborg Denmark, 2011, pp. 1541–1546.

[8] L. Savioja, J. Huopaniemi, T. Lokki, and R. Väänänen, "Creating interactive virtual acoustic environments," *J. Audio Eng. Soc.*, vol. 47, no. 9, pp. 675–705, 1999.

[9] J. B. Allen and D. A. Berkley, "Image method for efficiently simulating small-room acoustics," *J. Acoust. Soc. Am.*, vol. 65, no. 4, pp. 943–950, 1979.

[10] J. Borish, "Extension of the image model to arbitrary polyhedra," *J. Acoust. Soc. Am.*, vol. 75, no. 6, pp. 1827–1836, 1984.

[11] A. Krokstad, S. Strom, and S. Sørsdal, "Calculating the acoustical room response by the use of a ray tracing technique," *Journal of Sound and Vibration*, vol. 8, no. 1, pp. 118–125, 1968.

[12] H. Kuttruff, "Auralization of impulse responses modeled on the basis of ray-tracing results," *J. Audio Eng. Soc.*, vol. 41, no. 11, pp. 876–880, 1993.

[13] M. R. Schroeder, "Natural Sounding Artificial Reverberation," *J. Audio Eng. Soc.*, vol. 10, no. 3, pp. 219–223, 1962.

[14] J.-M. Jot and A. Chaigne, "Digital Delay Networks for Designing Artificial Reverberators," in *Audio Engineering Society Convention 90*, Feb. 1991. [Online]. Available: http://www.aes.org/e-lib/browse.cfm?elib=5663

[15] D. Rocchesso and J. O. Smith, "Circulant and elliptic feedback delay networks for artificial reverberation," *IEEE Trans. Speech Audio Process.*, vol. 5, no. 1, pp. 51–63, 1997.

[16] H. Bai, G. Richard, and L. Daudet, "Late Reverberation Synthesis: From Radiance Transfer to Feedback Delay Networks," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 23, no. 12, pp. 2260–2271, 2015.

[17] E. De Sena, H. Haciihabiboglu, Z. Cvetkovic, and J. O. Smith, "Efficient synthesis of room acoustics via scattering delay networks," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 23, no. 9, pp. 1478–1492, 2015.

[18] O. Das and J. S. Abel, "Grouped Feedback Delay Networks for Modeling of Coupled Spaces," *J. Audio Eng. Soc.*, vol. 69, no. 7/8, pp. 486–496, 2021.

[19] T. B. Atalay, Z. S. Gul, E. De Sena, Z. Cvetkovic, and H. Hachabiboglu, "Scattering Delay Network Simulator of Coupled Volume Acoustics," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 30, pp. 582–593, 2022.

[20] T. Wendt, S. van de Par, and S. D. Ewert, "A Computationally-Efficient and Perceptually-Plausible Algorithm for Binaural Room Impulse Response Simulation," *J. Audio Eng. Soc.*, vol. 62, no. 11, pp. 748–766, 2014.

[21] C. Kirsch, J. Poppitz, T. Wendt, S. van de Par, and S. D. Ewert, "Computationally Efficient Spatial Rendering of Late Reverberation in Virtual Acoustic Environments," in *2021 Immersive and 3D Audio: from Architecture to Automotive (I3DA)*, 2021, pp. 1–8.

[22] M. Scerbo, L. Savioja, and E. De Sena, "Room Acoustic Rendering Networks With Control of Scattering and Early Reflections," *IEEE Trans. Audio Speech Lang. Process.*, vol. 32, pp. 3745–3758, 2024.

[23] M. A. Biot and I. Tolstoy, "Formulation of Wave Propagation in Infinite Media by Normal Coordinates with an Application to Diffraction," *J. Acoust. Soc. Am.*, vol. 29, no. 3, pp. 381–391, 1957.

[24] A. D. Pierce, "Diffraction of sound around corners and over wide barriers," *J. Acoust. Soc. Am.*, vol. 55, no. 5, pp. 941–955, 1974.

[25] R. G. Kouyoumjian and P. H. Pathak, "A uniform geometrical theory of diffraction for an edge in a perfectly conducting surface," *Proc. IEEE*, vol. 62, no. 11, pp. 1448–1461, 1974.

[26] U. P. Svensson, R. I. Fred, and J. Vanderkooy, "An analytic secondary source model of edge diffraction impulse responses," *J. Acoust. Soc. Am.*, vol. 106, no. 5, pp. 2331–2344, 1999.

[27] T. Kawai, "Sound diffraction by a many-sided barrier or pillar," *Journal of Sound and Vibration*, vol. 79, no. 2, pp. 229–242, 1981.

[28] Z. Maekawa, "Noise reduction by screens," *Applied Acoustics*, vol. 1, no. 3, pp. 157–173, 1968.

[29] S. D. Ewert, "A filter representation of diffraction at infinite and finite wedges," *JASA Express Letters*, vol. 2, no. 9, p. 092401, 2022.

[30] C. Kirsch and S. D. Ewert, "A Universal Filter Approximation of Edge Diffraction for Geometrical Acoustics," *IEEE/ACM Trans. Audio Speech Lang. Process.*, vol. 31, pp. 1636–1651, 2023.

[31] A. Rungta, C. Schissler, N. Rewkowski, R. Mehra, and D. Manocha, "Diffraction Kernels for Interactive Sound Propagation in Dynamic Environments," *IEEE Trans. Visual. Comput. Graphics*, vol. 24, no. 4, pp. 1613–1622, 2018.

[32] A. Erraji, J. Stienen, and M. Vorländer, "The image edge model," *Acta Acust.*, vol. 5, p. 17, 2021.

[33] C. Schissler, G. Mückl, and P. Calamia, "Fast diffraction pathfinding for dynamic sound propagation," *ACM Trans. Graph.*, vol. 40, no. 4, pp. 1–13, 2021.

[34] S. Siltanen, T. Lokki, S. Tervo, and L. Savioja, "Modeling incoherent reflections from rough room surfaces with image sources," *J. Acoust. Soc. Am.*, vol. 131, no. 6, pp. 4606–4614, 2012.

[35] S. D. Ewert, N. Gößling, O. Buttler, S. Van De Par, and H. Hu, "Computationally-efficient rendering of diffuse reflections for geometrical acoustics based room simulation," *Acta Acust.*, vol. 9, p. 9, 2025.

[36] F. Brinkmann, L. Aspöck, D. Ackermann, S. Lepa, M. Vorländer, and S. Weinzierl, "A round robin on room acoustical simulation and auralization," *J. Acoust. Soc. Am.*, vol. 145, no. 4, pp. 2746–2760, 2019.

[37] M. Blau, A. Budnik, M. Fallahi, H. Steffens, S. D. Ewert, and S. van de Par, "Toward realistic binaural auralizations – perceptual comparison between measurement and simulation-based auralizations and the real room for a classroom scenario," *Acta Acust.*, vol. 5, pp. 1–13, 2021.

[38] M. Schutte, "Aspects of Room Acoustics, Vision and Motion in the Human Auditory Perception of Space," PhD, LMU, Munich, 2021.

[39] J. Sommerfeldt, J. M. Arend, F. Brinkmann, S. D. Ewert, and S. Weinzierl, "Source Directivity Implementation for Real-Time Room Acoustic Simulation with liveRAZR," in *DAGA*, Hannover, Germany, 2024, pp. 1616–1618.

[40] P. Majdak, F. Zotter, F. Brinkmann, J. De Muynke, M. Mihocic, and M. Noisternig, "Spatially Oriented Format for Acoustics 2.1: Introduction and Recent Advances," *J. Audio Eng. Soc.*, vol. 70, no. 7/8, pp. 565–584, 2022.

[41] S. D. Ewert, O. Buttler, and H. Hu, "Computationally Efficient Parametric Filter Approximations for Sound-Source Directivity and Head-Related Impulse Responses," in *2021 Immersive and 3D Audio: from Architecture to Automotive (I3DA)*, IEEE, 2021, pp. 1–6.

[42] V. Pulkki, "Virtual sound source positioning using vector base amplitude panning," *J. Audio Eng. Soc.*, vol. 45, no. 6, pp. 456–466, 1997.

[43] M. Vorländer and E. Mommertz, "Definition and measurement of random-incidence scattering coefficients," *Applied Acoustics*, vol. 60, no. 2, pp. 187–199, 2000.

[44] H. Steffens, M. Schutte, and S. D. Ewert, "Acoustically driven orientation and navigation in enclosed spaces," *J. Acoust. Soc. Am.*, vol. 152, no. 3, pp. 1767–1782, 2022.

[45] H. Steffens, M. Schutte, and S. D. Ewert, "Auditory orientation and distance estimation of sighted humans using virtual echolocation with artificial and self-generated sounds," *JASA Express Letters*, vol. 2, no. 12, p. 124403, 2022.