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REALTIME ACOUSTIC IMPULSE RESPONSE GENERATION USING STATIC RAY TRACING

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This paper presents a real-time method for simulating sound propagation using ray tracing techniques to generate impulse responses and modify sound playback through convolution processing. By modeling the behavior of acoustic waves in virtual environments, the approach makes it possible to reproduce realistic sound through accurate handling of reflections, diffractions, and reverberations. A novel static-ray technique and a listener directed diffuse rain technique are introduced, designed to improve stability by capturing the acoustic properties of the scene without the noise typically introduced when generating new rays each frame. The paper also introduces a novel running index method for updating the impulse response continuously, which prevents artifacts during auralization and ensures smooth playback. Together with real-time convolution processing, these contributions create an immersive audio experience that reacts dynamically to changes in the environment. The results show that the approach can effectively reproduce complex acoustic interactions, offering both high fidelity and responsiveness in real-time audio rendering, and providing a solid foundation for future spatial audio applications in virtual reality and game design.

Key words: sound propagation, ray tracing, convolution, impulse response, real-time audio, spatial audio, acoustic simulation.

1. Introduction

Achieving realistic sound propagation is important for building immersive digital experiences in such areas as virtual reality, computer games, and architectural acoustics. The behavior of sound in real environments depends on reflections, diffractions, and reverberations, which strongly affect how listeners understand spatial and environmental cues. Traditional audio design methods usually do not have enough adaptability to react to fast changes in virtual scenes, which limits their use in interactive applications.

This research proposes a new real-time method for sound simulation that combines ray tracing for modeling acoustic interactions with Fast Fourier Transform (FFT)-based convolution for applying impulse responses to audio playback [1]. The novelty of the method is in the use of static rays in the ray tracing process, which makes the impulse responses more stable. In many existing approaches rays are regenerated every frame, which causes fluctuations in the results. The introduced technique avoids such instability and keeps the responses consistent between frames, reducing audio artifacts that can break sound quality.

In addition, a novel running index method is presented for real-time updating of the impulse response. The long response is divided into smaller segments, and the system can switch between them smoothly when the environment changes. This design allows continuous playback without interruptions and preserves realistic perception for the listener.

By combining static-ray stability with the adaptability of the running index, the proposed framework provides a novel solution for real-time spatial audio simulation. The

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following sections describe the methodology, implementation, and results, demonstrating that the approach can achieve both stability and realism in interactive virtual environments.

2. Static Ray Tracing Approach

Impulse response generation is a critical component in simulating realistic sound propagation, capturing the environmental characteristics that shape how sound interacts with surfaces and obstacles [2]. In this implementation, impulse responses are generated through ray tracing, where sound rays are cast into the environment, interact with surfaces, and are traced back to a listener position. Each ray represents a potential path sound might take, accounting for factors like reflection, diffraction, and energy decay.

To achieve stable and consistent impulse responses, this approach employs static rays, ensuring that each frame captures a predictable pattern of reflections. By using static rays, the impulse response remains stable across frames, reducing audio artifacts and fluctuations that might occur due to rapid changes in ray distribution. This approach allows the system to process impulse responses consistently, preserving sound quality in real-time rendering.

The impulse response data for each ray is then stored and updated using a running index. This method segments the impulse response into manageable chunks, which can be processed individually without requiring a complete refresh of the entire response data each time. The running index cycles through these chunks, allowing the application to handle long impulse responses while seamlessly adapting to changes in the environment. This segmentation reduces computational load and enables efficient real-time updates, maintaining high-quality audio fidelity.

The following subsections detail the ray tracing setup, the calculation of reflection and attenuation factors, and the running index mechanism that manages real-time impulse response updates.

2.1. Impulse Response

An impulse response (IR) characterizes how an acoustic environment reacts to a brief, idealized sound—often represented as a Dirac delta function, $\delta(t)$. The impulse response captures all reflections, reverberations, and decays over time, encapsulating the environment's unique acoustic properties. For any input sound signal $x(t)$, the output $y(t)$ after passing through the environment is given by the convolution of $x(t)$ with the impulse response $h(t)$ [3]:

$$y(t) = x(t) * h(t) = \int_{-\infty}^{\infty} x(\tau)h(t-\tau)\,d\tau \tag{1}$$

In discrete terms, assuming a sampled input signal $x[n]$ and an impulse response $h[n]$, the convolution operation is represented as:

$$y[n] = \sum_{k=0}^{N-1} x[k] \cdot h[n-k], \tag{2}$$

where N is the length of the impulse response. This convolution describes how each element of the input signal is modified by the environment's acoustic properties as defined by $h(t)$.

The impulse response itself is generated by tracing sound rays as they interact with the environment. Each ray contributes a component to $h(t)$ based on its path length, angle, and energy decay. By summing the contributions from multiple rays, the total impulse response accurately models the environment's acoustic signature, allowing realistic sound simulation.

The ray-based approach enables efficient generation of impulse responses for real-time applications, balancing accuracy and performance. In the following subsection, the setup of ray tracing and the parameters used to calculate each ray's impact on the impulse response are discussed.

2.2. Ray Tracing Using Static Rays

Ray tracing is a technique used to simulate sound propagation by following the paths of individual rays when they interact with surfaces in the environment [1]. In this study, a novel static-ray tracing is introduced, where rays are cast in fixed directions in every frame. In traditional approaches, rays are regenerated each frame, which produces unstable responses [2, 5]. The novel static-ray approach introduced in this study avoids this problem by keeping the same set of rays across frames. As a result, the impulse response stays stable in time and random fluctuations are reduced, which also helps to minimize artifacts in the audio output.

Algorithm 1 Ray Tracing Using Static Rays

```
Initialize number of rays, maximum reflections, energy decay factor, and maximum distance
for all frames do
    Initialize rays in fixed directions
    for all rays do
        Set reflection count = 0
        while reflection count < max reflections and ray energy > threshold and distance < max distance do
            if ray hits a surface then
                Record ray energy
                Reflect ray direction based on surface orientation
                Apply decay factor to ray energy
                Update distance traveled with hit distance
                Set new ray origin to hit position
                Increment reflection count
            else
                Exit loop for this ray
            end if
        end while
    end for
end for
```

Each ray starts from the sound source and moves outward, reflecting from surfaces according to the laws of reflection. When a ray hits a surface, its direction is updated based on the surface normal, and its energy is decreased to simulate decay with distance and after impact. This process repeats until the ray's energy becomes too small or the maximum number of reflections is reached.

For each frame, the algorithm calculates the paths and interactions of all static rays with the environment, recording the energy and delay (time of travel) of each ray that reaches the listener. This information is then used to build an impulse response that reflects the environment's acoustic characteristics.

This approach of using static rays and capturing their reflections ensures that the impulse response generated in each frame remains consistent, even as the environment or listener position changes. By recording each ray's path and energy, this ray tracing algorithm captures the core characteristics of the environment's acoustic properties, contributing to a realistic and stable audio experience.

In earlier work on non-static ray tracing [7], new rays were regenerated every frame. While this approach was flexible, it also introduced a noticeable drawback: the impulse response kept changing slightly from frame to frame, even when the scene, source, and listener were completely still. These random fluctuations acted like noise in the system, masking the finer details of acoustic changes and sometimes creating audible artifacts. The static-ray method avoids this issue by reusing the same set of rays in every frame. As shown in Figure 1, this keeps the noise level flat and predictable, while the non-static method produces large, irregular jumps. The benefit is that any variation in the impulse response now comes only from real changes in the environment—such as a moving source, listener, or obstacle—rather than from randomness in the algorithm. This makes the auralization smoother, more reliable, and easier to trust in real-time applications like VR or game audio.

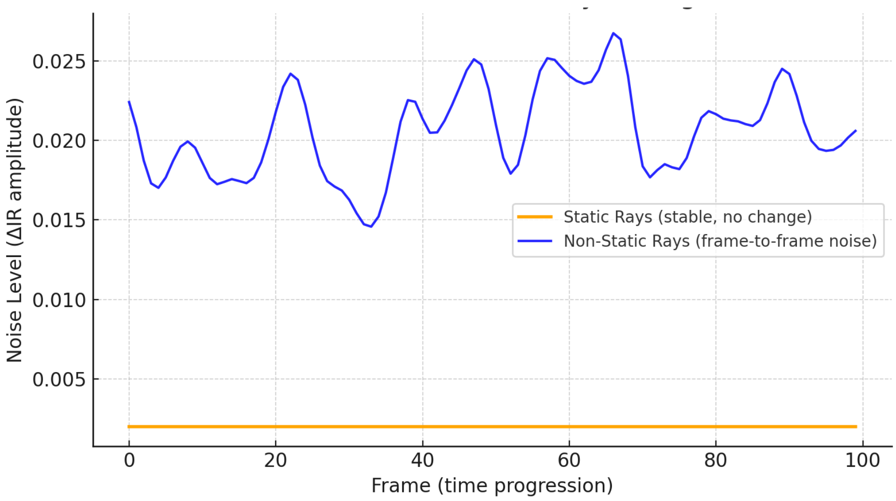

Fig. 1. Frame-to-frame noise level in impulse responses for static versus non-static ray tracing. Static rays maintain consistently low noise, while non-static rays introduce fluctuations due to random ray regeneration

2.3. Listener Directed Diffuse Rain

The *Diffuse Rain* technique is a sound propagation model used to simulate the natural diffusion of sound energy as it interacts with complex environments [4]. In this method, sound rays are scattered in random directions from each reflection point, emulating the way sound diffuses upon hitting irregular surfaces. This approach effectively captures the random scattering and softening of sound that occurs in real-world settings, enhancing the realism of spatial audio simulations. However, by directing rays uniformly in all directions, the original Diffuse Rain technique can be computationally intensive, as it requires many rays to ensure that sound energy reaches all parts of the virtual environment. This creates a trade-off between realism and efficiency, particularly in real-time applications.

The novelty of technique introduced in this study is that the rays are directed toward the listener from each reflection point, optimizing computational efficiency while maintaining realistic sound diffusion. Rather than scattering rays randomly, this approach selectively directs rays from each hit point toward the listener, incorporating energy decay based on the angle between the ray's direction and the listener's location. This strategy minimizes unnecessary calculations, preserving impulse response accuracy and improving performance.

In this method, when a ray reaches a reflection point, a new ray is directed toward the listener's position. To account for natural energy decay with indirect paths, an energy loss factor based on the angle between the incoming ray direction and the listener vector is calculated. The energy loss factor, E_{loss}, is defined as:

$$E_{\text{loss}} = \max\left(\left(\mathbf{d} \cdot \frac{\mathbf{l} - \mathbf{p}}{\|\mathbf{l} - \mathbf{p}\|}\right)^2, 0\right), \tag{3}$$

where:

- $\mathbf{d}$ is the direction of the incoming ray,
- $\mathbf{l}$ is the listener's position,
- $\mathbf{p}$ is the ray's current origin (hit point),
- $\mathbf{d} \cdot \frac{\mathbf{l}-\mathbf{p}}{\|\mathbf{l}-\mathbf{p}\|}$ represents the cosine of the angle between $\mathbf{d}$ and the normalized vector from $\mathbf{p}$ to $\mathbf{l}$.

This calculation determines the amount of energy reaching the listener based on the alignment of the ray's path with the listener. Higher alignment (i.e., smaller angles) results in lower energy loss, enhancing the intensity of sound reaching the listener.

Once E_{loss} is computed, the energy of the ray is adjusted:

$$\mathrm{E}_{\text{adjusted}} = \mathrm{E}_{\text{original}} \times E_{\text{loss}} \tag{4}$$

The direction of the adjusted ray, $\mathbf{d}_{\text{new}}$, is then normalized towards the listener:

$$\mathbf{d}_{\text{new}} = \frac{\mathbf{l} - \mathbf{p}}{\|\mathbf{l} - \mathbf{p}\|} \tag{5}$$

If a direct line of sight exists between the current point and the listener, the adjusted ray is added to the list of resulting rays with the computed energy and distance.

This Listener Directed Diffuse Rain approach efficiently directs sound rays toward the listener while incorporating natural energy decay based on the alignment of each ray. This technique significantly reduces computational demands by eliminating unnecessary scattering, thus optimizing the simulation of realistic sound propagation in spatial audio applications.

This approach is detailed in Algorithm 2.

Algorithm 2 Listener Directed Diffuse Rain

Require: Reflection point, listener position, incoming ray direction
Ensure: Ray directed toward listener with adjusted energy and distance

```
Compute vector v_l from reflection point to listener
Compute energy loss with angle between incoming ray and v_l using Eq. (3)
if direct path from reflection point to listener exists then
    Update ray direction toward listener with Eq. (5)
    Adjust ray energy with computed loss using Eq. (4)
    Update traveled distance by adding distance reflection point → listener
    Add adjusted ray to resulting ray set
end if
```

2.4. Impulse Response Generation

The *Impulse Response Generation* process captures the acoustic characteristics of an environment by aggregating the contributions of individual sound rays. Each ray represents a distinct path of sound, interacting with the environment through reflection, decay, and diffusion. By calculating the effect of each ray, an impulse response is generated that encodes the environment's overall acoustic signature.

The impulse response generation process involves several steps:

1. **Arrival Time Calculation**: For each ray, the arrival time is determined based on the distance traveled from the sound source to its reflection point. Given the speed of sound, c, the arrival time t is calculated as:

$$t = \frac{d}{c}, \tag{6}$$

where d is the total distance traveled by the ray. This arrival time maps each ray to a specific point in the impulse response array.

2. **Distance-Based Attenuation**: As sound travels, it loses energy. This attenuation follows an inverse square law, where energy decreases as the distance increases. The attenuation factor E_d based on distance d can be described by:

$$E_d = \frac{1}{d^2 + \epsilon}, \tag{7}$$

where ϵ is a small constant to avoid division by zero.

3. **Energy Adjustment and Amplitude Conversion**: The final energy E_r of each ray, after accounting for all collisions and reflections, is converted directly into an amplitude that contributes to the impulse response array at the calculated sample index. This amplitude represents the acoustic influence of that ray at a particular moment within the environment.

The amplitude A contributed by each ray to the impulse response can be expressed as:

$$A = E_r \times E_d \times s, \tag{8}$$

where:

- E_r: The ray's energy after all collisions and reflections. This final energy value incorporates both the initial energy and any reductions from reflections.

- E_d: Distance-based attenuation factor, reducing energy based on the distance traveled, as described in Equation (7).
- s: A phase-shifting factor, alternating between $+1$ and -1 to introduce phase shifts.

This amplitude A is then added to the impulse response array at the corresponding sample index, representing the ray's arrival time. By summing the contributions from all rays, the impulse response array accumulates the acoustic energy patterns over time, encoding the environment's reflective and absorptive properties into an audio signal.

4. **Normalization and Final Phase Shifts**: After all rays have been processed, the impulse response is normalized by dividing each entry by the maximum energy recorded. This ensures that values are scaled within a standard range, making the impulse response suitable for further processing. A final phase-shifting factor is applied to introduce minor phase variations across samples, enhancing realism in the resulting audio.

The impulse response generated through this method provides a time-encoded representation of the environment's acoustic response. This response can be convolved with audio signals to simulate realistic sound propagation, making it adaptive to virtual environments. By accounting for timing, energy decay, and phase shifts, this approach captures essential aspects of acoustic behavior in a computationally efficient way.

3. Auralization

Auralization is the process of rendering audible sound from simulated or measured acoustic data, allowing listeners to experience an environment's acoustic characteristics. Impulse responses are typically used to model how sound interacts with a given space. The impulse response, which represents the acoustic signature of an environment, is convolved with an input audio signal to recreate how that sound would be perceived in the target environment. This convolution process combines the original audio with the time and frequency characteristics of the impulse response, effectively simulating reflections, reverberations, and absorptions unique to the environment. Mathematically, this is expressed by Equation (1), where

$$y(t) = x(t) * h(t) \tag{9}$$

$y(t)$ represents the resulting auralized sound, $x(t)$ is the input signal, $h(t)$ is the impulse response of the environment, and $*$ denotes convolution.

3.1. Running Index Method

In real-time sound simulation, the impulse response must be updated every frame to reflect dynamic changes in sound sources, listener positions, and environmental interactions. This is essential for immersive experiences in virtual reality and interactive simulations. However, recalculating the impulse response every frame requires multiple Fast Fourier Transforms (FFTs), which can be computationally intensive.

The **running index method** enables real-time impulse response updates by segmenting FFT calculations over successive frames. Rather than recalculating the entire impulse response each frame, the method divides it into smaller segments and updates only one or a few segments per frame.

A running index tracks the current segment, cycling through each chunk sequentially. This staggered approach minimizes the immediate processing load, distributing FFT calculations across frames and integrating results gradually. This method ensures stable audio quality and prevents abrupt changes in the impulse response, effectively balancing performance with audio fidelity in real-time simulations.

4. Results

This section presents a comparative analysis of impulse response characteristics in a simulated room with dimensions of 50 meters by 50 meters by 20 meters. In this environment, the sound source is positioned at the center of the room at a height of 1 meter, while the listener is placed 1 meter to the right of the source. The simulated room is perfectly reverberant, with no absorption on any surface, creating an idealized setting to observe the effects of ray count and reflection depth on the impulse response.

The impulse response was captured over a duration of 3 seconds at a sample rate of 41 kHz, resulting in a total of 123,000 samples. This high sample rate allows for precise temporal resolution, enabling an accurate representation of the room's reflective properties across time. The following analyses compare impulse responses based on variations in the number of rays and the number of reflections, as well as comparisons with theoretical reverberation estimates.

4.1. Number of Rays

The impulse response generated using varying numbers of rays-10, 100, and 1000-each with a maximum of 8 bounces is compared. This comparison helps illustrate how the number of rays affects the accuracy and density of reflections in the impulse response, particularly in capturing mid and late reverberations that contribute to the richness of the acoustic environment.

As shown in Figure 2, the impulse response displays a greater density of reflections over time as the number of rays increases, capturing more of the mid and late reverberation details.

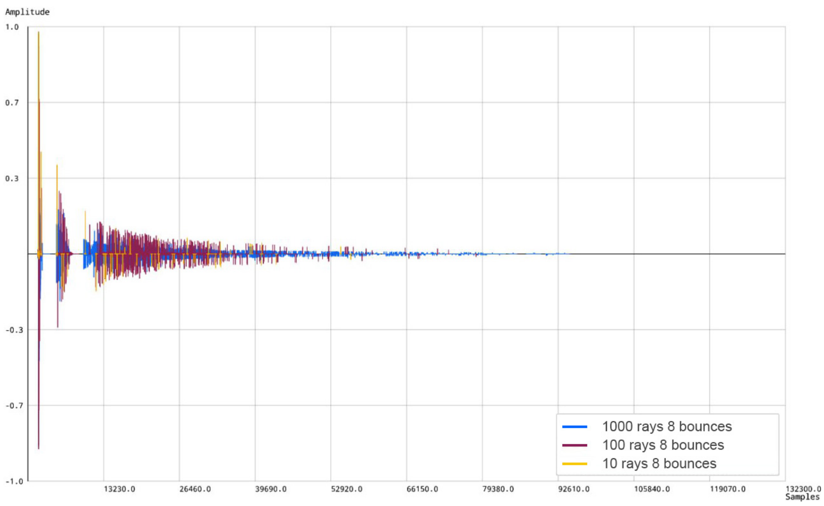

Fig. 2. Impulse responses generated with 10, 100, and 1000 rays, each with 8 bounces

With only 10 raysthe impulse response lacks density, capturing only a limited number of early reflections. This sparse response fails to represent the full reverberant behavior of the room, as many potential paths and late reverberation components are not accounted for.

As the ray count increases to 100, more mid and late reflections are present in the impulse response. This increase provides a more detailed representation of the room's acoustics, as additional rays capture more complex interaction paths and reflections.

With 1000 rays, the impulse response becomes even richer, with a continuous decay pattern that closely resembles a dense reverberant field. This high ray count allows the simulation to capture extensive mid and late reverberations, which contribute to the perception of spaciousness and realism. The impulse response becomes smoother and more continuous, accurately reflecting the room's reverberant nature with minimal gaps in the decay profile.

Thus, increasing the number of rays significantly enhances the amount of collected amplitude data of the impulse response, allowing for a more comprehensive capture of the acoustic energy over time. This density of reflections is essential for realistic sound simulations in highly reverberant spaces, particularly when capturing the full extent of reverberation is crucial for immersive audio experiences.

4.2. Number of Bounces

The effect of varying the number of bounces is compared using impulse responses generated with 2, 5, and 8 reflections per ray, with each setup using 1000 rays. This analysis reveals how additional bounces contribute to the persistence and richness of reverberation within the room, allowing for a more detailed representation of the room's acoustic behavior over time.

As shown in Figure 3, the impulse response reflects a greater accumulation of reflections as the number of bounces increases, contributing to the reverberant tail and enhancing the perception of space within the room.

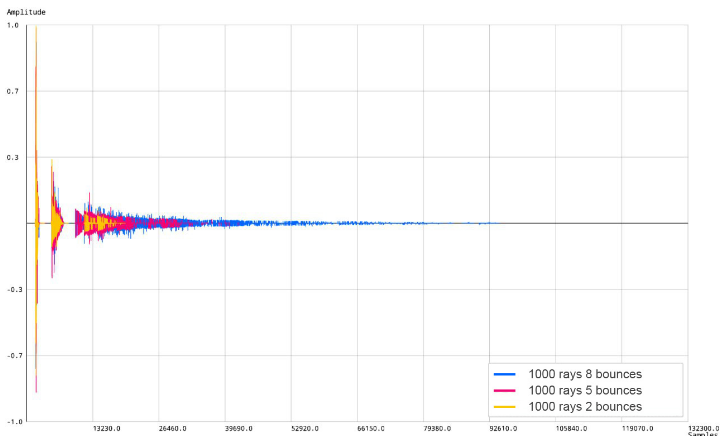

Fig. 3. Impulse responses generated with 1000 rays and varying bounces: 2, 5, and 8

With only 2 bounces the impulse response captures primarily early reflections, with a limited reverberant tail. This configuration reflects a sharp decay, as fewer reflections contribute to the late reverberation, resulting in a relatively sparse response that may not fully represent the room's reverberant nature.

As the number of bounces increases to 5, more mid and late reflections appear in the impulse response, extending the reverberation tail and enhancing the response's density. This increase in reflections provides a more accurate representation of sound persistence within the room, capturing more complex sound paths that contribute to a realistic acoustic experience.

With 8 bounces, the impulse response shows an even richer and more continuous decay profile, representing a high level of reverberant energy over time. The extended reverberation tail captures late reflections that contribute to the perception of spaciousness and echo, as the additional bounces allow sound to interact with more surfaces within the room. This setup provides a closer approximation to a continuous reverberant field, which is critical in accurately simulating large or highly reflective spaces.

Thus, increasing the number of bounces enhances the impulse response's ability to capture late reverberations and create a full reverberant tail. This added complexity is essential for applications requiring realistic sound reflections, as it contributes significantly to the perceived acoustic depth and richness within the simulated environment.

4.3. Comparison with Sabine's Formula

The simulated impulse responses generated with varying numbers of ray bounces (2 through 10) are compared against the theoretical reverberation time, T_{60}, predicted by Sabine's formula [6]:

$$T_{60} = \frac{0.161 \times V}{A \times \alpha}, \tag{10}$$

where V is the room volume, A is the total surface area, and α is the average absorption coefficient. For a room with dimensions of 50 m by 50 m by 20 m and $\alpha = 0.1$, Sabine's formula yields a reverberation time of approximately 8.05 seconds. This theoretical value provides a benchmark for evaluating the accuracy of our simulation.

As shown in Figure 4, increasing the number of bounces in the simulation progressively brings the impulse response closer to the theoretical T_{60}. With only 2 bounces, the response is dominated by early reflections, resulting in a shorter decay time than predicted. As the number of bounces increases, mid and late reflections are captured, extending the decay time and yielding a more realistic reverberant field.

By around 8 to 10 bounces, the simulated reverberation time closely aligns with Sabine's theoretical value, demonstrating that additional bounces enhance the accuracy of the model by better capturing the room's reverberant characteristics.

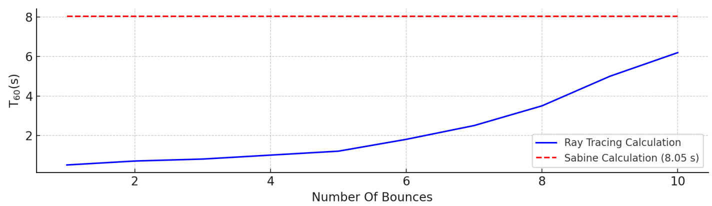

Fig. 4. Simulated reverberation times with varying ray bounces compared to Sabine's theoretical T_{60}

5. Conclusions

This study examined how the number of rays and bounces affects the accuracy of impulse response simulations in a perfectly reverberant room, using ray tracing combined with a running index method for real-time processing. The results show that more rays increase the level of detail in mid and late reflections, while additional bounces extend the reverberant tail and create a more realistic sense of space. A central part of this work is the introduction of novel methods of static ray tracing and listener directed diffuse rain, which keep the impulse response consistent from frame to frame. By avoiding the constant regeneration of new rays, this approach removes the noise and fluctuations that often reduce sound quality in real-time systems. We also introduced a running index method for updating impulse responses smoothly across frames. Instead of recalculating everything at once, the impulse response is broken into segments that are updated gradually, which prevents audible artifacts and ensures stable playback even in dynamic environments. Together, these techniques make it possible to combine stability, efficiency, and realism in a single system. The approach not only captures complex acoustic interactions with high fidelity but also remains practical for real-time use in applications such as virtual reality, architectural acoustics, and game audio.

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ГЕНЕРАЦІЯ АКУСТИЧНОЇ ІМПУЛЬСНОЇ ХАРАКТЕРИСТИКИ В РЕАЛЬНОМУ ЧАСІ З ВИКОРИСТАННЯМ СТАТИЧНОГО ТРАСУВАННЯ ПРОМЕНІВ

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У роботі описано метод, який дозволяє в реальному часі моделювати поширення звуку за допомогою трасування променів. Такий підхід дає змогу будувати імпульсні характеристики та змінювати відтворення аудіо через згорткову обробку. Запропонований метод статичного трасування та напрямлених променів враховує поведінку акустичних хвиль у віртуальному середовищі, та уникає шумів в реальному часі, що виникають при швидкій зміні траєкторій трасуючих променів, та забезпечує більш реалістичне звучання завдяки розрахунку відбиттів, дифракцій і реверберацій.

Також представлено метод «running index» для безперервного оновлення імпульсної характеристики, що запобігає артефактам під час ауралізації та забезпечує плавне відтворення. Такий підхід може стати основою для удосконалених систем просторового аудіо у віртуальній реальності та при розробці ігрового звукового дизайну. Експерименти показують, що метод здатний відтворювати складні акустичні взаємодії з достатньою точністю та водночас залишатися придатним для використання в реальному часі.

Ключові слова: поширення звуку, трасування променів, згортка, імпульсна характеристика, аудіо в реальному часі, просторове аудіо, акустична симуляція.