

Sound field evaluation on acoustical experiment with several loudspeaker locations

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Abstract

When a laboratory at a technical college is used as a site for psychoacoustic experiments, there are objects in the experimental environment such as walls, floors, ceilings, and chairs that obstruct the propagation of sound waves, which may change the accuracy of sound direction localization by altering the sound waves reaching the listener's ears. Therefore, the purpose of this study was to clarify the effects of these environments on sound localization characteristics using simulations. In this paper, we evaluated the effect of the reflection of sound waves on a chair to keep posture of the subject on the sound image localization for several loudspeaker positions. Results showed that it isn't strongly possible that the early reflections due to the chair will clearly shift the direction of the sound image.

Keywords: large-scale simulation, acoustical sound field, non-steady state analysis, interaural time difference

1. Introduction

When a laboratory at a technical college is used as a site for psychoacoustic experiments, there are objects in the experimental environment such as walls, floors, ceilings, and chairs that obstruct the propagation of sound waves, which may change the accuracy of sound direction localization by altering the sound waves reaching the listener's ears. Therefore, the purpose of this study is to clarify the effects of these environments on sound localization characteristics using simulations. We determined the parameter values of the non-stationary analysis method of the sound field needed for the above evaluation at last year's ICAROB [1]. In this paper, we evaluated the effect of the reflection of sound waves by a chair set up to immobilize the subject on the subject's sound image localization for loudspeaker positions placed every 45 degrees in the horizontal plane.

2. Adventure Sound

2.1. Brief introduction

ADVENTURE_Sound is an open-source software for sound simulation. This allows us to simulate parallel finite element analysis software for the sound fields in huge space. Analyses are performed by solving the large-scale linear system with parallel computing based on the Mandel's domain decomposition method (DDM) [2].

2.2. Simulation procedures with ADVENTURE_Sound

The procedure to use ADVENTURE_Sound is as follows; 1) generate the mesh data, 2) add boundary conditions (BCs), 3) subdivide the mesh data into subdomains, 4) simulate the sound field distributions, 5) convert to the results to vtk data for visualization. On the fourth procedure, it needs to calculate Helmholtz equation in the time domain. For this, Newmark β method is adopted to the equation. As the boundary conditions, input wave function and first and second derivatives of that function.

3. Simulation on a typical psychoacoustic experimental environment

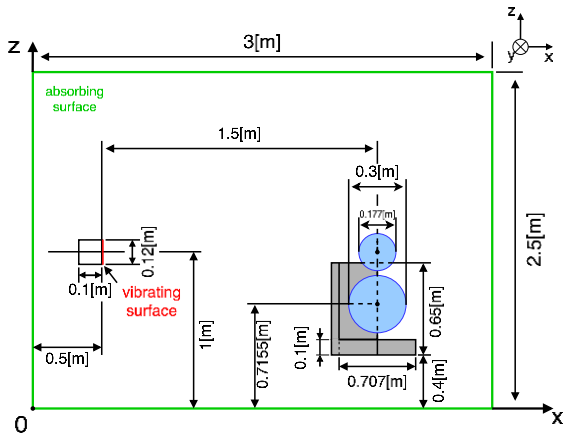
3.1. simulation model

Fig. 1 shows that a simulation model that includes a loudspeaker-listener pair with a chair that holds listener's body. This is designed for typical psychoacoustical experiment. The shape of the listener is a snowman consisted of two spheres, their diameters approximating the average head and body [3]. The loudspeaker is located at 135° which is behind it as seen by the listener.

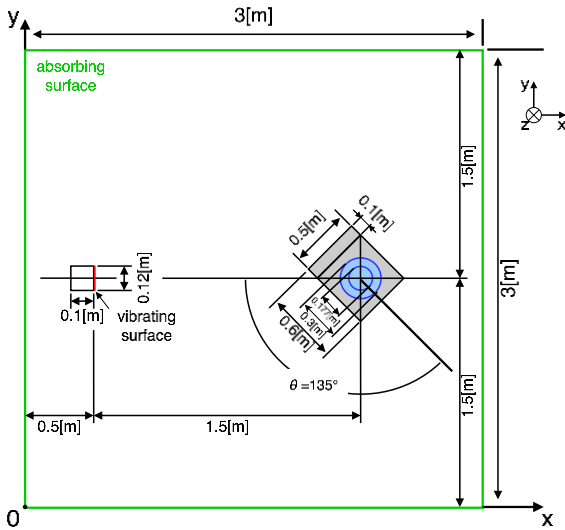
The loudspeaker is assumed to be a small-diameter, full-range product often used in sound image localization experiments. This model is used to evaluate the time waveform caused by sound waves emitted from a loudspeaker and reflected or diffracted by a chair.

In this model, the acoustic impedance of the room walls, ceiling, and floor is set to a value of 445.9kg/m²s to ensure that sound waves are completely absorbed. On the

other hand, the acoustic impedance of the surface of the loudspeaker and the chair is set to $4.56 \times 10^9 \text{ kg/m}^2\text{s}$ to ensure that sound waves are completely reflected.



(b) side view



(a) Bird's view

Figure 1. simulation model of psychoacoustical

Table 1 shows that the dimensions of the simulation model. The distance between the loudspeaker and the subject is 1.5 m, which is known as the transitional distance from spherical waves to plane waves in the audible bandwidth.

Table 1. dimension of the simulation model

Room, width[m] × depth[m] × height[m]	3 × 3 × 2.5
Room volume [m ³]	22.5
Loudspeaker, width[m] × depth[m] × height[m]	0.1 × 0.12 × 0.12
Diameter of the listener head, torso [m]	0.177, 0.3

Distance from loudspeaker to listener [m]	1.5
Backrest, width[m] × depth[m] × height[m]	0.5 × 0.1 × 0.55
Seat, width[m] × depth[m] × height[m]	0.5 × 0.5 × 0.1

3.2. Specs of the parallel computer

Our parallel computer is called as TOMAHAWK.

Table 2 shows that the detailed specs of the parallel computer.

Table 2. Specs of parallel computer

CPU	Intel Xeon E5-2650L, 1.8GHz with 16 cores L2 20480[KB]
Main memory/node [GB]	32
Number of nodes	7
Total cores	112
OS	CentOS 6.2
Compiler	gcc-4.4.7, mipch-3.2

3.3. Simulation conditions

Table 3 shows that analysis conditions. These parameter values come from previous our study [4].

Because of the mesh size is 0.025m and the time resolution is 1 μs and the Courant number equals to 0.0238, therefore the Courant condition is satisfied.

Table 3. Analysis conditions for the test model

Mesh size[m]	0.025
Number of elements	16,264,673
Acoustic impedance of the Room wall, floor, ceiling [kg/m ² s]	445.9
Acoustic impedance of the Loudspeaker, chair, listener	4.56×10^9
Medium(air) ρ_0 [kg/m ³]	1.3
Speed of sound c_0 [m/s]	343
Time resolution Δt [μs]	1.0
Duration time to simulate [μs]	8000
Courant number $\sqrt{3} \times c \times \Delta t / \Delta x$	$0.0238 \ll 1$
Velocity potential $\{\Phi\}_t$	Gaussian pulse
Full width at half maximum [ms]	0.264
Cut-off frequency f_c [Hz]	646

For boundary conditions as input wave, the single wave is useful to analysis the transient sound field, because sound reflections are observed clearly. In this study, a gaussian pulse as velocity potential is adopted as follows,

$$\{\Phi\}_t = e^{-\alpha(t-t_0)^2} \dots (1)$$

In the Eq(1), decay parameter α is set to 3141, delay parameter t_0 is set to 0.001[s].

3.4. Simulation results

Fig. 2 shows the simulation results for every 1000 microseconds on xz plane at y=1.5m. In this figure, the red color indicates a high intensity of velocity potential proportional to sound pressure. It can be seen that sound waves are emitted from the loudspeaker, and as time passes, the sound waves travel toward the listener, and at approximately 5049 μ s, the direct waves reach the listener's head. After 6049 μ s, the sound waves are seen to be distributed in a complex manner due to reflections from the chair.

Fig. 3 shows the simulation results on xy plane at z=1m. In this figure, the direct waves emitted from the loudspeaker arrive at the backrest of chair between 3049 μ s and 4049 μ s. At t=6049 μ s, the first reflected waves occur around the listener.

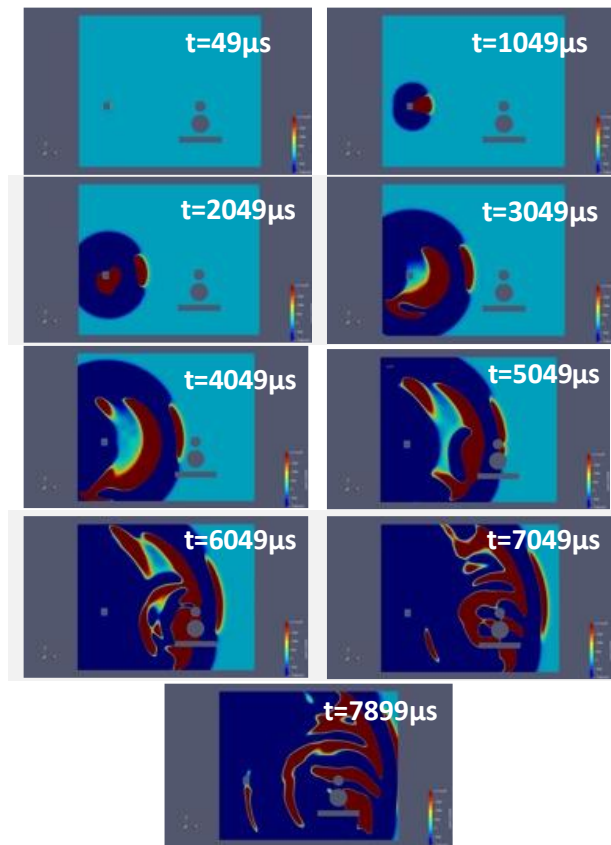


Figure 2. Visualization of the sound field at several times on xz plane at y=1.5m

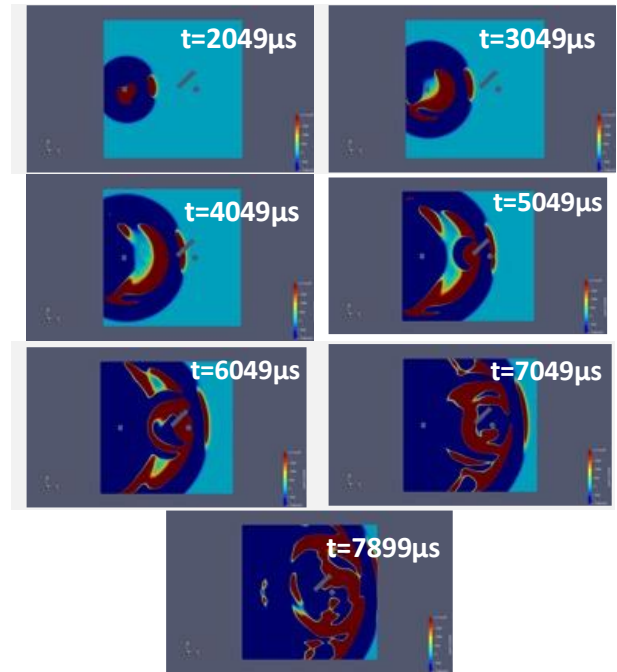


Figure 3. Visualization of the sound field at several times on the xy plane at z=1m

3.5. Discussions

Fig. 4 shows the simulation results on the xz plane at y=1.5m when the sound wave arrived at the listener.

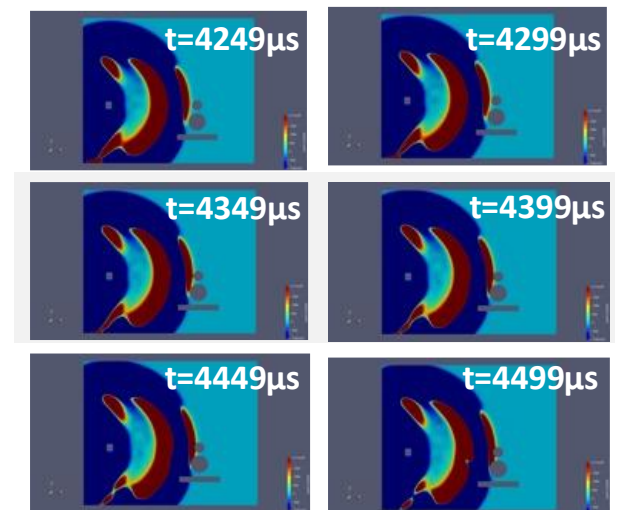


Figure 4. Visualization of the sound field when the sound waves arrived at listener on the xy plane at y=1.5m

The timing obtained from this figure, is approximately 4399 μ s. Because the sound velocity was 343m/s and the elapsed time was 4399 μ s, the distance between the loudspeaker and the listener is estimated at 1.50m. This is consistent with the distance set in the simulation model, indicating the validity of this simulation.

Fig. 5 shows that the simulation results on the xy plane at z=1m when the sound waves arrived at the listener. The

time at which the sound waves reach the listener's right ears and left ears is $4499\mu\text{s}$ and $4799\mu\text{s}$, respectively.

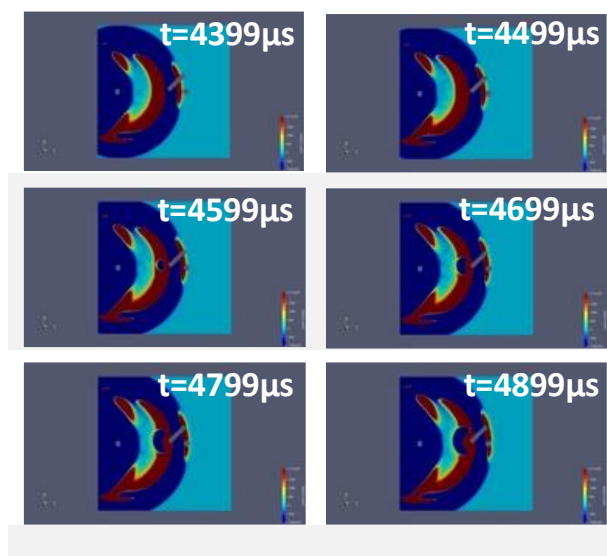


Figure 5. Sound field when the first reflection waves arrived at both ears on the xy plane at $z=1\text{m}$

Fig. 6 shows that the first reflections (called as early reflections) arrived at both ears of listener. The time at which the reflected wave reaches the listener's two ears is $5499\mu\text{s}$ and $5999\mu\text{s}$, respectively.

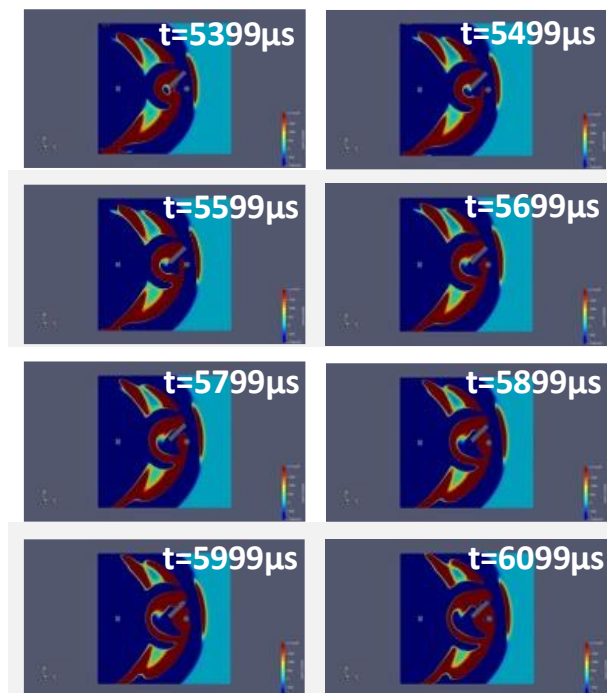
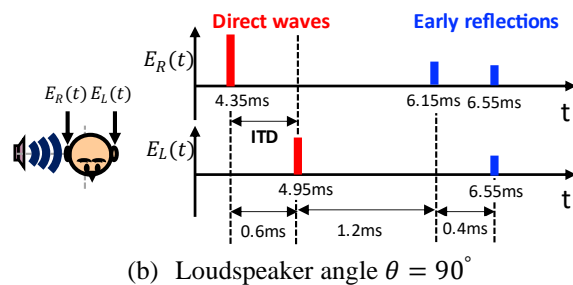
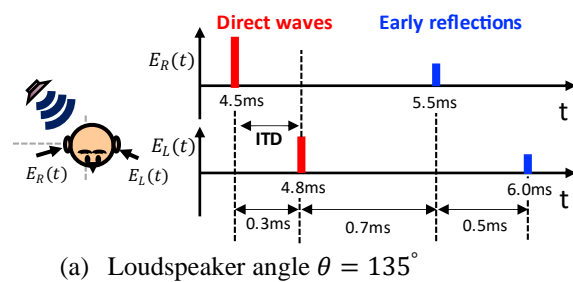


Figure 6. Sound field when the first reflection waves arrived at both ears on the xy plane at $z=1\text{m}$

The time profile under different loudspeaker locations is shown in fig. 7. In this figure, (b) is the results from previous study [4].



(b) Loudspeaker angle $\theta = 90^\circ$



(a) Loudspeaker angle $\theta = 135^\circ$

Figure 7. Time profile of sound waves at both ears at different loudspeaker locations

At $\theta = 135^\circ$, the time difference between the arrival of sound waves at the two ears is known as ITD. The ITD was estimated to be approximately 0.3ms. The time difference between right direct waves and left early reflections is 1.5ms. Another time difference between left direct waves and right early reflections is 0.7ms. Because these differences are greater than the maximum of theoretical ITDs whose value is 0.67ms obtained from $\Delta s/c = d(\theta + \sin \theta)/2c$ [5] where the Δs is the path difference when sound waves reach both ears, c is the sound velocity, d is the diameter of listener's head and θ is the azimuth angle with respect to the front of the listener.

From fig. 7(b), at $\theta = 90^\circ$, also, both time differences between the direct waves and the early reflections on the facing side exceed the maximum of ITD.

Therefore, it isn't strongly possible that the early reflections will clearly shift the direction of the sound image, since these time differences are slightly greater than the maximum ITD.

4. Conclusion

Purpose of this paper is to clarify the effect of these environments on the human auditory characteristics, especially sound image localization ability. The simulation target is loudspeaker and snowman model with chair, was adopted as a typical psychoacoustical experimental environment.

The simulation results showed that 1) The arrival times of sound waves at both ears when emitted from a loudspeaker placed at 135° behind the listener were 4.5ms and 4.8ms, respectively, 2) The arrival times of the early reflections at both ears were 5.5ms and 6.0ms, respectively, 3) The time differences from the direct wave

in the right ear to the early reflections in the left ear and from the direct wave in the left ear to the early reflections in the right ear, were 1.5ms and 0.5ms, respectively.

Our conclusion is that it isn't strongly possible that the early reflections will clearly shift the direction of the sound image as perceived by the listener, because these time differences between the direct waves and early reflections slightly exceeds the range in the time domain that affects sound image localization.

Acknowledgements

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