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Automobile Passenger Compartment Sound Field Control and Analysis

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One of the final purposes of audio playback is a reproduction of "presence" which a listener cannot experience except in a place of performing music. Our company had been trying to get "presence" in a car compartment by the addition of reflective sounds to ordinary stereophonic.

Fujitsu TEN has developed a sound field control algorithm which design concepts are to keep characteristics of music source, to be front localization basically, and to let incident directions of reflective sounds coincide real sound fields.

This paper describes an algorithm for generating sound field which has good presence in a car compartment, and introducing a method of measurement and evaluation for quantitatively grasping its effect.

1. Introduction

The passenger compartment of a car is a special sound field. Any listening point is asymmetrical with respect to the speakers. Reflective objects and sound absorbing materials are concentrated in a small space.

The resulting sound lacks presence and stereophonic effect.

Fujitsu TEN has developed a sound field control algorithm that can modify the sound field of any passenger compartment to give the presence of a live performance. This paper introduces our concept of passenger compartment sound field control and a method of sound field evaluation.

Figure 1. Concert hall

2. Passenger compartment and concert hall sound fields

The term sound field refers to any space in which sound waves are present. When we listen to a sound, we hear reverberation from the sound field as well as the original sound. These reverberations add a sense of depth or presence to the music.

The sound fields reverberation characteristics are determined by its geometry and volume. Figures 1 to 6 show a concert hall, a listening room, and an automobile passenger compartment and their impulse responses. Table 1 gives their reverberation characteristics.

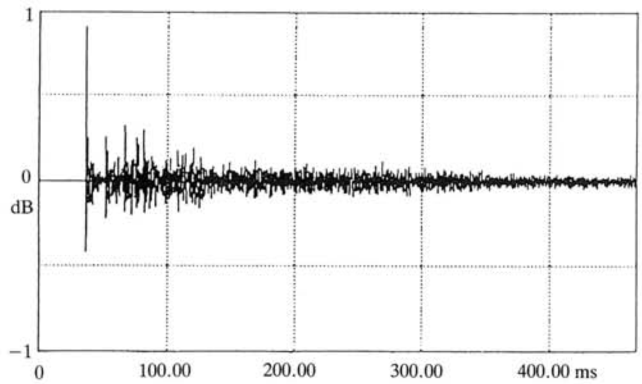

Figure 2. Concert hall impulse response

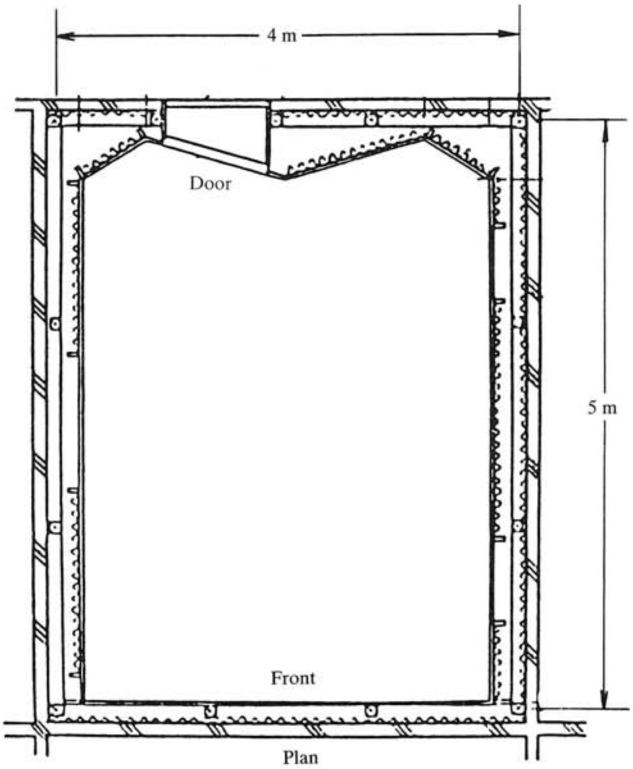

Figure 3. A listening room

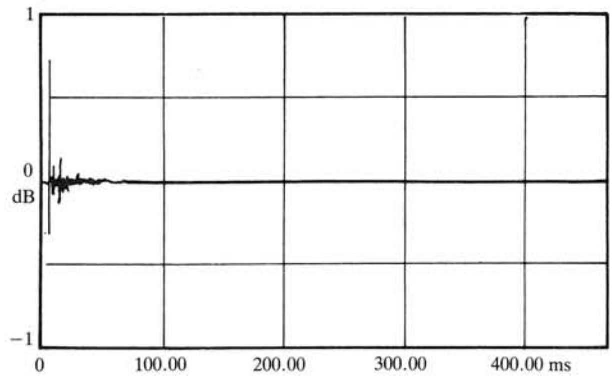

Figure 4. Listening room impulse response

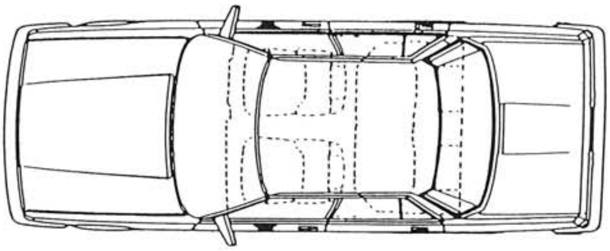

Figure 5. Automobile passenger compartment

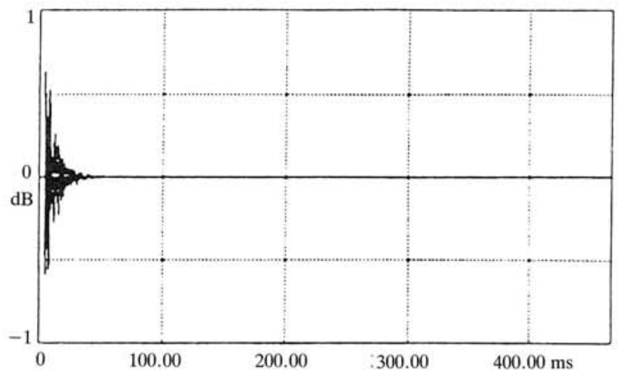

Figure 6. Passenger compartment impulse response

Table 1. Sound field volumes and reverberation times

	Passenger compartment	Listening room	Concert hall
Volume	3 m^3	100 m^3	15,000 m^3
Seating capacity	5 persons	10 persons	2,000 persons
Reverberation time (500 Hz)	0.04 second	0.3 second	1.6 seconds

It is apparent from Figures 2, 4, and 6 that the reverberation of each sound field is related to its impulse response. The three cubic meter passenger compartment, has an extremely short mean free path and has little reverberation due to its abundant sound absorbing material. It should also be noted that the direct sound and the initial reflection are in close proximity. This makes the reflected sound more like a modification of the direct sound than a reverberation.

Speakers are often mounted in the front door panels or on both sides of the underdash panel due to physical constraints imposed by the structure of the automobile. Consequently, the sound stage extends below and in front of the listener. A common remedy to this problem is to raise the sound stage by installing the speakers in the rear tray. This speaker layout, however, is uncomfortable for the listener in the rear seat. The sound this passenger hears is direct and is coming from behind.

3. Sound field control algorithm

Our objectives for achieving passenger compartment sound field control were:

① To create presence and depth not attainable with the previous methods.
② To preserve the original stereophonic sound quality without impairing the integrity of the sound source.
③ Front localization.
④ To allow for the directions of arrival of the initial reflected sound and reverberation as in an actual concert hall.
⑤ To produce multiple user-variable sound field patterns to adaptable to any musical source.

To develop an algorithm of sound field control, we visited several concert halls and assessed their sound field characteristics (Figure 7).

Figure 8 shows a speaker layout designed to control the sound field by the addition of reflected sounds. The speakers are mounted in the front doors, rear tray, and in the center of the underdash panel. The center and rear speakers face upwards. Reflected sounds are generated by a digital signal processor and are distributed to the speakers as shown in Table 2.

One way of producing presence or concert-hall-like reverberation would be to have a speaker installed in each location from which reflected sound would arrive. This being impractical, only one auxiliary speaker is used – the center speaker. The center speaker is the minimum prerequisite for achieving front localization, improving stereophonic balance, ensuring presence using the initial reflection, and creating a sense of front depth due to reverberation.

With this speaker layout, the listener is first exposed to direct sound at eye level, followed by the initial reflection coupled with reverberations. This creates presence.

Table 2. Sound composition

Speaker	Output
Front door speakers	Stereo signals
Center speaker	Monaural signals + reverberation + initial reflection
Rear speakers	3D bass + reverberation

Figure 7. Measurement

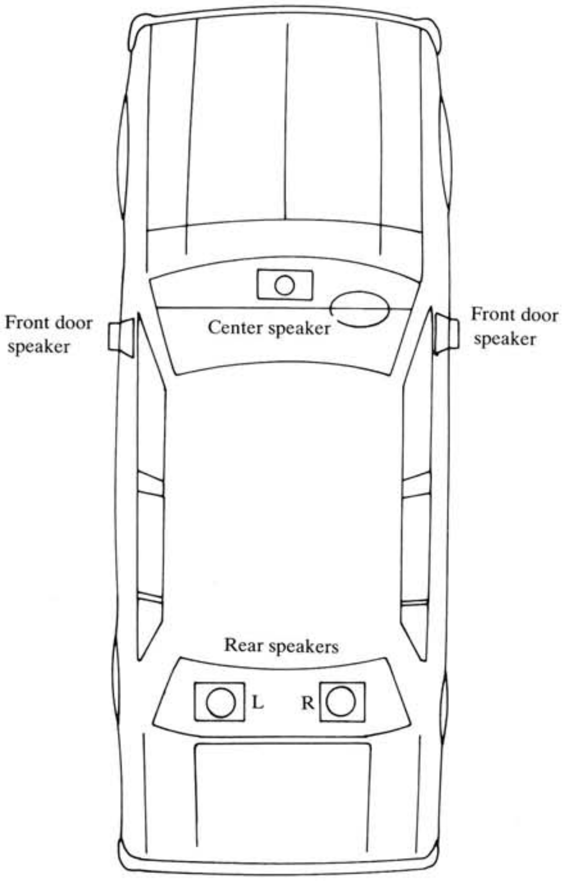

Figure 8. Speaker layout

The diagrams in Figures 9 and 10 show the directional distribution of reflected sounds in the passenger compartment as measured by the closely-located 4-point microphone method. Figure 9 is the pattern of conventional four-speaker system; Figure 10 is the pattern of the five-speaker system with sound field control. The car used is a 2,000 cc four-door sedan. Measurements were made at the driver's seat. With the conventional system, directivity is asymmetrical and is distributed towards the right front speaker close to the listener (driver's seat) and the rear speakers. With sound field control, the reflected sound seems to come from many directions. A significant improvement of front balance associated with the center speaker was also noted.

Similar assessments were made with larger or smaller cars. Similar effects were attained by varying the sound field signals. Different car profiles had little or no effect on the sound quality. Quality was heavily dependent on the power or directivity of the speakers used for direct sound reproduction.

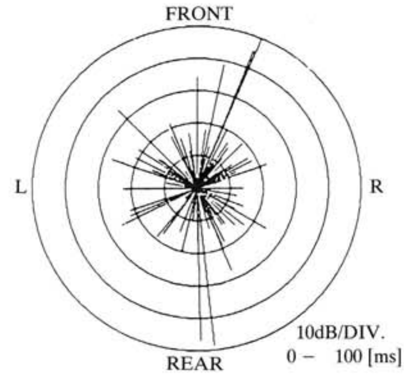

Figure 9. Direction distribution pattern (4 speakers: Stereo reproduction)

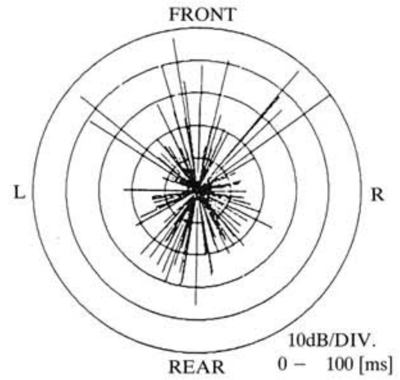

Figure 10. Direction distribution pattern (5 Speakers: Sound field control)

4. Outline of digital signal processing

We have investigated the mechanisms of initial reflection and reverberation by analyzing the acoustics of concert halls. These analyses and reviews of sound reproduction techniques have led us to develop an effective form of sound field control for an automobile passenger compartment. Figure 11 is a block diagram of the sound field control signal processing system. The main processor is a Fujitsu TEN FT8800 DSP which features high-speed floating point arithmetic for real-time generation of high-quality, high-precision digital delays, initial reflection sound convolution (low-order), reverberation, filtering, and mixing.

Reflected sound in a sound field is interleaved with repetitions of complicated and attenuated reflections of radiated sound from the floor, walls, and ceilings. Our scheme of sound field control keeps precise track of this process. As shown in Figure 12, the sound propagation delay is divided into several conceptual units. Natural sound fields such as concert halls, large clubs, and stadiums are simulated by using the acoustic characteristics of actual sound fields as parameters. In this way, presence and depth of a live performance are reproduced inside the car.

In passenger compartment sound field control, as with previous methods of stereo reproductions, the quality of original sound is directly related to the performance of the system. This should remind us of the importance of preserving the quality of the musical source.

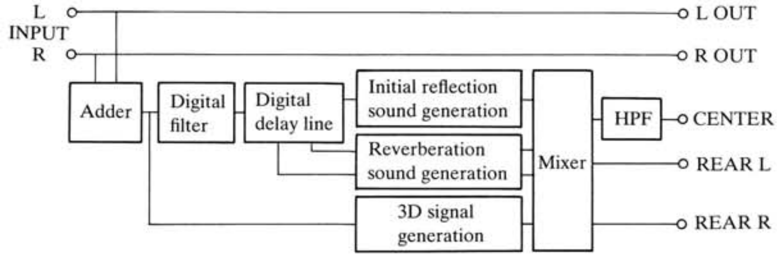

Figure 11. Digital signal processing

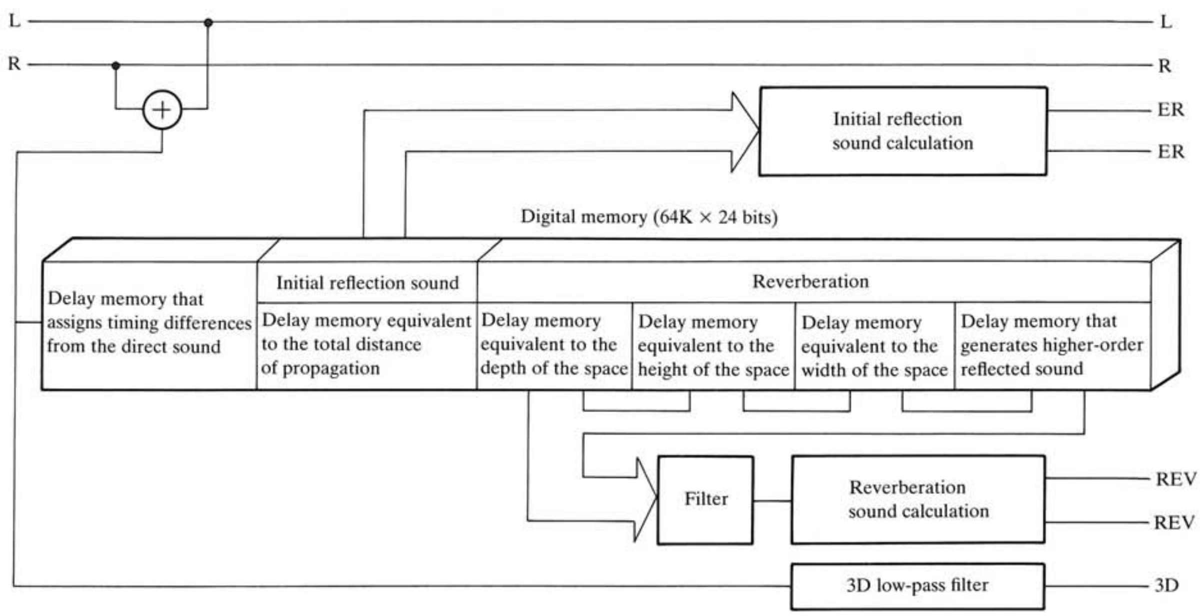

Figure 12. Memory allocation of delay time

5. Assessment method (Closely-located 4-point microphone method)

The closely-located 4-point microphone method was developed at the Waseda University Institute of Science and Technology to analyze reflected sound. Space coordinates and power of image sound sources are determined by analyzing the time differences of sound impulses arriving at four closely-spaced microphones assuming that reflected sound impulses are output from the image sound sources. We used this method in our measurement of passenger compartments under the supervision of Waseda University professor Yoshio Yamazaki. This chapter describes how the closely-located 4-point microphone method works. Sample measurements are included.

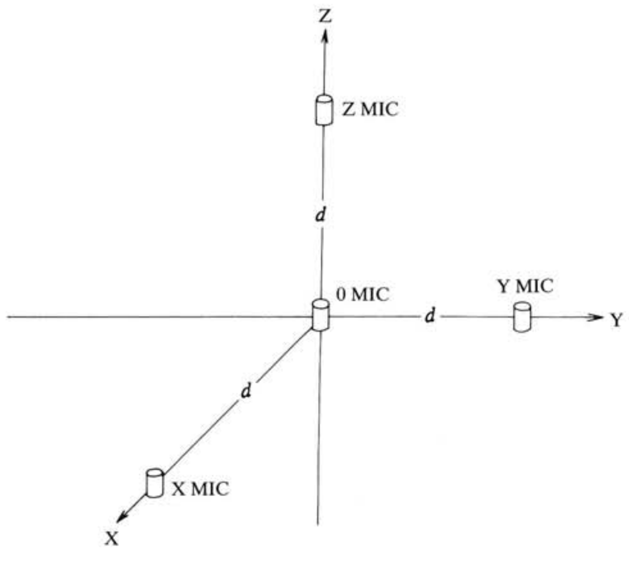

Figure 13. Microphone orientation

5.1 Principle of operation

Two points in a space can be located if their distances from three points can be established. Further, one of these two points is determined if its distance from a point off the plane passing through these three points is established. To facilitate postprocessing, the three other points (microphone locations) are set an equal distance (d) from the origin in a 3-dimensional cartesian coordinate system as shown in Figure 13.

The principle of closely-located 4-point measurement is explained by referring to a simple experiment in which one reflecting plate is installed in a free sound field (Figure 14). The impulse responses recorded at each points have differences, though slight, in their timing as shown in Figure 15. If the coordinates (Xn, Yn, Zn) of the virtual sound sources associated with the reflected sounds are determined from these timing differences, they can be represented by the following equations.

$$Xn = (d^2 + r_{on}^2 - r_{xn}^2)/2d$$
$$Yn = (d^2 + r_{on}^2 - r_{yn}^2)/2d$$
$$Zn = (d^2 + r_{on}^2 - r_{zn}^2)/2d \qquad (1)$$

where

$r_{on} = ct_{on}$
$r_{xn} = ct_{xn}$
$r_{yn} = ct_{yn}$
$r_{zn} = ct_{zn}$ c: Sound velocity

Generally, the impulse response waveforms are not as simple as shown in Figure 15. The speaker or microphone characteristics are convoluted in the direct sound, and the characteristics of the reflective surface are convoluted in the reflected sound. Analyses of actual impulse responses, involve the determination of the same reflected sound by the use of a short-period crosscorrelation coefficient.

The procedure for establishing the coordinates of an image sound source is outlined below.

1) Isolate the reflected sound of interest to a window from the microphone impulse response at the origin. This is the reference waveform.
2) Calculate the crosscorrelation coefficient between the reference waveform and response of microphone X over the range constrained by the microphone interval. Determine the arrival time of the reflected sound from the timing difference that is maximum.
3) Perform a similar operation with another reflected sound and with microphones Y and Z.
4) Establish the coordinates of the image sound source of interest from Equations (1), and the power of the image sound source from the isolated waveform.

5.2 Measurement system

Figure 16 is a block diagram of the measurement system. Impulse sound is regenerated from pulse samples converted from digital to analog and fed to a speaker (for measurement as in a concert hall, opposing in-phase speakers are used as sound sources). The impulse responses of the four microphones are recorded after obtaining average of the responses according to the S/N ratio during measurement.

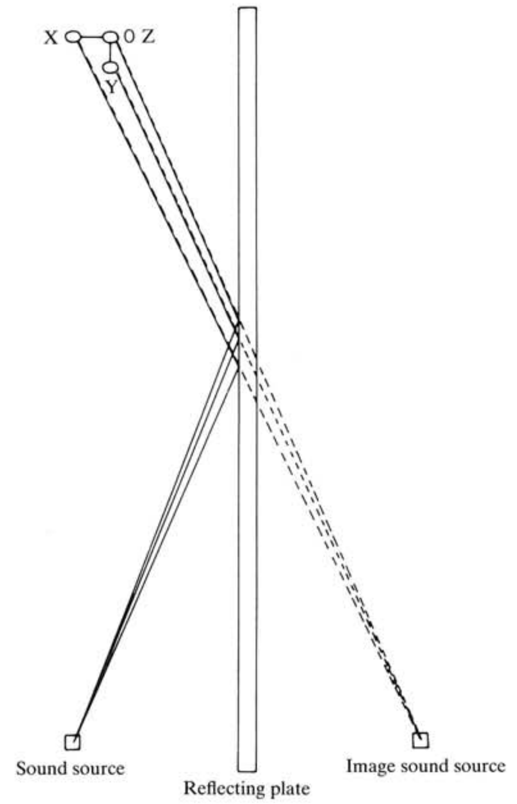

Figure 14. Relation between sound source and image sound source

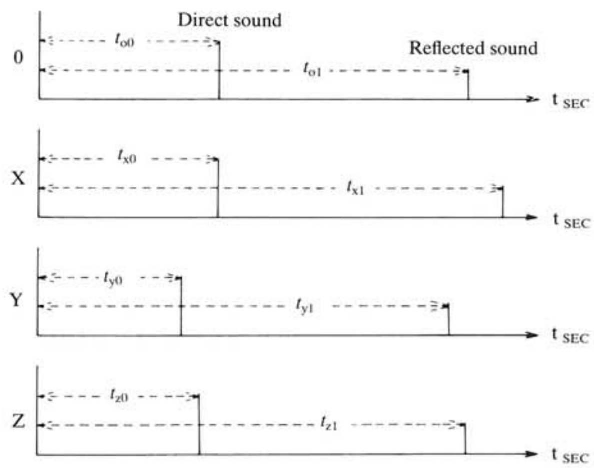

Figure 15. Inpulse responses of four points

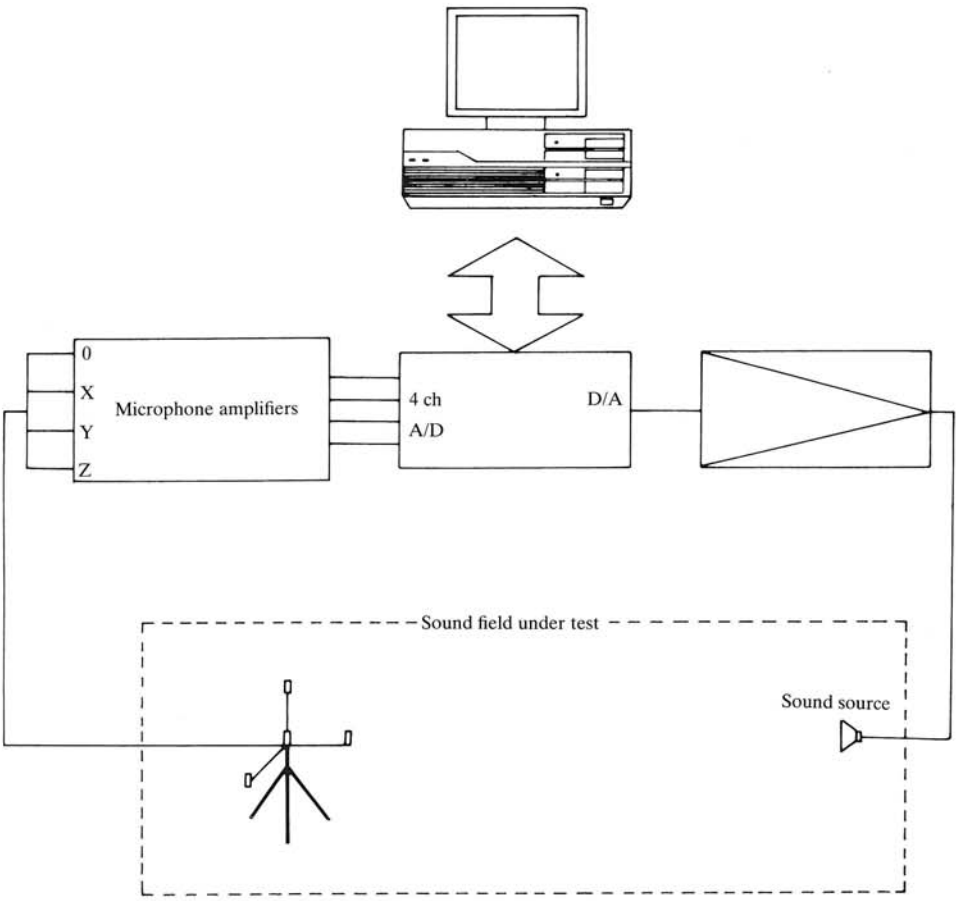

Figure 16. Measurement setup

5.3 Measurement examples

Figures 17 and 18 show the impulse response, image sound source distribution, and directional distribution measurements of actual concert halls. In the microphone layout, the center of the front stage is on the Y-axis. The center of each circle in the image sound source distribution diagrams denotes the coordinates of the image sound source, and the radius represents its power. The square symbol □ in each diagram indicates the point of sound reception. The line segment in the lower right corner is the distance scale and represents 10 m.

In the directional distribution, the power of the image sound source incident at opening angles of ± 45 degrees or less in the vertical direction of the display surface is projected on the display surface. Power is represented as line segment length.

5.4 Impulse sound source for narrow space measurement

When the passenger compartment is measured by the closely-located 4-point microphone method, the location of the sound sources (speakers) is important because transient response is impaired if the speaker baffle lacks the necessary rigidity. In a narrow spaces, such as the interior of a car, the time difference between the arrival of direct and the reflected sounds is so small that the use of sound sources with poor transient responses can make it difficult to distinguish the reflected sound. As a solution to this problem, the use of an impulse sound source with short-gap spark is now under study. This sound source generates an impulse sound by discharging the electric energy stored in a capacitor in an electrode space and allowing the air in the space

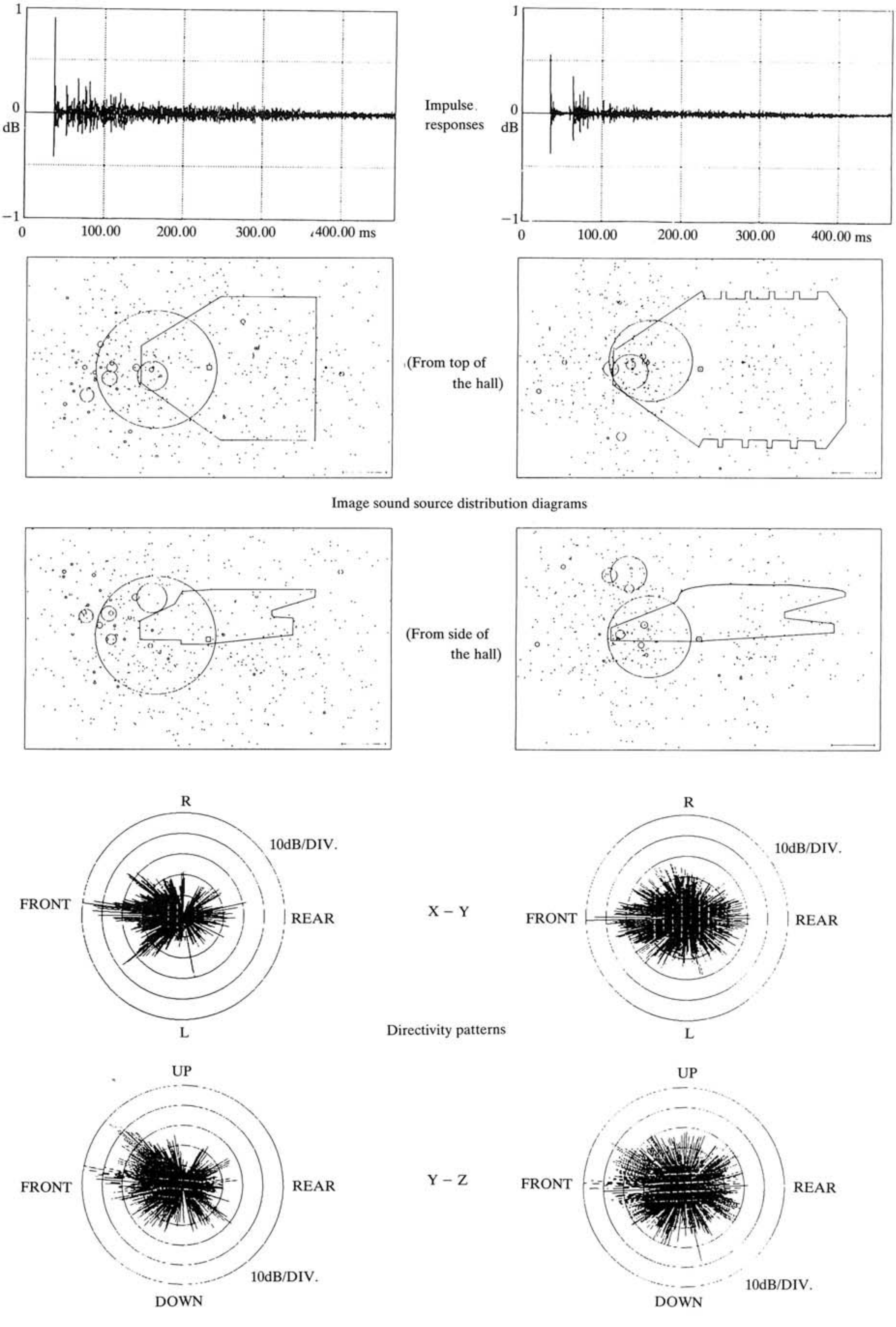

Figure 17. Measured characteristics of concert hall A

Figure 18. Measured characteristics of concert hall B

to expand thermally (Figure 19). This sound source is free from the mechanical vibration associated with conventional speakers and produces an impulse sound with a short pulse width and excellent transient response. Figure 20 shows a coaxial spark gap of speaker unit type.

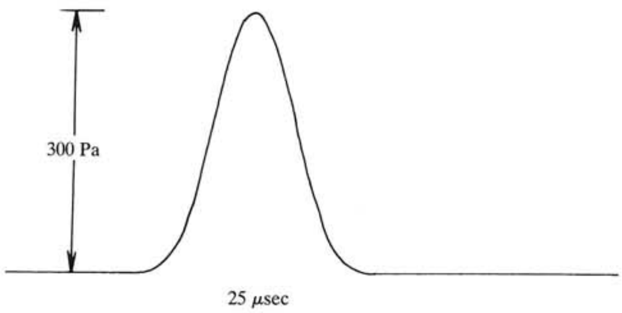

Figure 19. Impulse sound waveform

6. Conclusion

We have presented a sound field control system which adds reflected sounds to an auxiliary speaker and rear speakers to create presence. The auxiliary speaker is mounted facing upwards in a panel under the dashboard. We think this method of sound field control will find popular acceptance in future automobile sound systems.

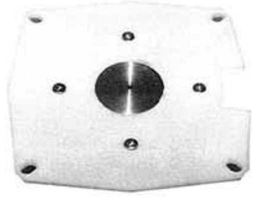

Figure 20. Coaxial spark-gap sound source

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