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Sound Field Control in a Car Compartment

—A Study of an Asymmetrical Sound Stage Expander and New Developed Surround System—

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With the availability of CDs, DATs and other new media, sources are attaining dynamic ranges of almost 100 dB. Currently, agressive efforts are being directed at sound field control to create concert-hall presence in stereophonic reproduction or in listening rooms. However, efforts to obtain satisfactory stereophonic effects in an automobile are confronted with the problems of interference with acoustic transmission and degraded sound quality depending on the volume, geometry, and building material of the car compartment, listening position in it, and so on. This paper discusses our latest approaches to acoustic transmission characteristics control technology, namely, sound image control and a surround sound system, with reference to such topics as improvement of asymmetrical stereophonic reception and creation of auditory perspective effects through reflection sound control.

1. Introduction

While CD players, tape decks and other new media have undergone continual performance improvements so far, the stereophonic reproduction of music in the compartment of a car (herein called car compartment) may still give the listener a cramped or biased feeling in the sound stage [*1] due to the asymmetrical location of the listening position with respect to the left- and right-side loudspeakers. Certain characteristics of the car compartment, such as small volume and special geometry, make disordered transmission frequency characteristics inevitable. This paper discusses sound image control and a surround sound system developed by the authors as solutions to problems of spatial impression inherent in the car compartment, such as localization and auditory perspective, with regard to their advantages, how to achieve them, and so on.

*1 Sound stage
The sound field in the area enclosed by two loudspeakers in front or in a small area outside that area.

2. Our approaches to sound field control

2.1 Characteristics of car compartment reception

Generally, the car compartment differs from a listening room in the following acoustic conditions:

1) Distribution of the resonant frequencies resulting from the volume and geometry of the car compartment range from several tens to several humdreds of Hertz, causing a disturbance to the transmission resonant frequencies.
2) The listener is often located at an asymmetrical position with respect to the left- and right-side speakers, so that the proper stereophonic sense of the source is often not appreciated.
3) Rigid and soft interior materials are intermixed, limiting the direction of arrival of the reflected sound.
4) The reverberation time is short, with little echoing.
5) Constraints on the location, aperture, and physical dimensions of loudspeakers limit the sound pressure in the low-frequency range, with the sound pressure peaking in the vicinity of 1−2 kHz.

These factors combine with one another in the car compartment to produce conditions unfavorable to a listening room.

2.2 Sound field correction

To date, extensive technical discussions have taken place on the following problems encountered by listeners in the car compartment, and solutions have been developed:

1) Correction of peaks or dips in transmission frequency characteristics caused by reflected waves or natural vibrations.
2) Correction of shortage of low-pitched sounds resulting from declines in listening sensitivity in the low-frequency zone with a low sound volume.

Fujitsu TEN has approached these problems by comparing findings of hearing tests with physical data.

Fujitsu TEN's method of correcting transmission frequency characteristics is by determining ideal transmission frequency characteristics in the car compartment while measuring them with a microphone and FFT analyzer installed at the listening position, then equalizing the characteristics through a circuit called a fixed equalizer.

It also achieves the correction of shortage of low-pitched tones with a low sound volume by studying the amount of electrical correction needed on the basis of the findings of hearing tests in actual car runs with reference to the Robinson-Dadson's equisignal curve [1].

These treatments have offered significant improvements in frequency characteristics, the sense of sound volumes, and so on. On the other hand, few studies have focused on the problems of echoing caused by the volume, building materials, and other characteristics of the car compartment, and spatial impressions of the reproduced sound stage, such as deviations in the listening position, lack of stereophonic feeling due to limited space, and the direction of localization.

The authors would like to report an asymmetrical sound stage expander developed by Fujitsu TEN to solve the problems of stereophonic reception from loudspeakers at asymmetrical positions and a new surround sound system designed to create auditory perspective and depth in stereophonic reproduction in a small space.

3. Sound image control

With car-mounted stereophonic reproducers, the left- and right-side speakers are located asymmetrically with respect to the listener, so that the proper stereophonic reproduction effects are not attainable. To correct such reductions in stereophonic effects, an asymmetrical sound stage expander has been developed at Fujitsu TEN on a conceptual extension of the theory of $\triangle P - \triangle \phi$[2] with regard to sound-pressure difference $\triangle P$ and phase difference $\triangle \phi$ produced in the two ears of the listener.

3.1 Control method summary

General two-channel stereophonic signals are designed to produce the maximum stereophonic effects for a listener at a standard position in the listening room.

In stereophonic reproduction in the car compartment, the asymmetrical location of the left- and right-side loudspeakers in front of the listener biases the sound distribution toward whichever speaker is closer to the listener. To correct such loss of the stereophonic effect, an asymmetrical sound stage expander has been developed on conceptual extension of the theory of $\triangle P - \triangle \phi$, which is a technique for analyzing stereophonic reproduction sound stages. The device is asymmetrical with the listening position being restricted to the driver's seat as shown in Figure 1. In summary, the process works in the following sequence:

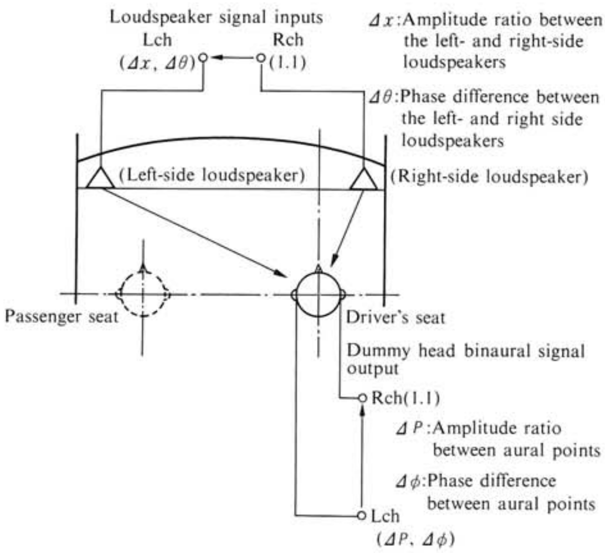


Figure 1. Asymmetrical listening condition in a car compartment

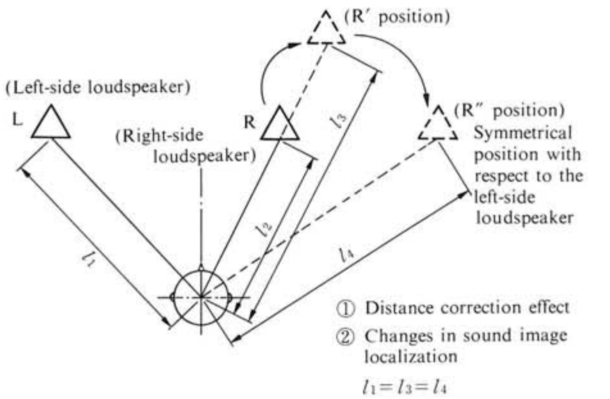


Figure 2. Moving of the imaginary right-hand sound source

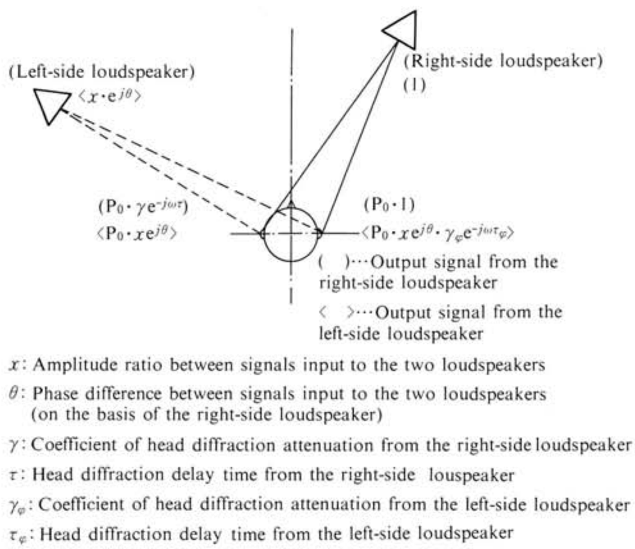


Figure 3. Calculation of sound pressure levels on aural points

1) Calculate amplitude ratio $\triangle x$ and phase difference $\triangle\theta$ between the output signals from the left- and right-side loudspeakers by using the theory of $\triangle P - \triangle\phi$.
2) Assign this relationship between the output signals from the left- and right-side loudspeakers through electrical signal processing for output from the two speakers.
3) Set the $\triangle P$ and $\triangle\phi$ values for the listener from these output signals and produce an apparent sound image at the position symmetrical with the left- and right-side loudspeaker about the front of the listener.

If the sound image thus created could be reproduced stereophonically from the right-side speaker, it would produce a sound image with the proper localization of the sound image and a uniform sense of lateral auditory perspective. In an asymmetrical condition like that shown in Figure 2, the difference between the left- and right-side loudspeakers in their distance to the listening position creates the following problems:

1) The sound pressure output from the loudspeaker that is closer to the listener is higher than that output from the other speaker.
2) Signals output from the loudspeaker closer to the listener reach him sooner.

These considerations make it necessary to factor the timing difference, in addition to $\triangle x$ and $\triangle\phi$, between the signals output from the left- and right-side loudspeakers for the listener to hear. Hence, attenuators and delay units are connected to the right channel side to adjust the sound pressure level and the arrival time difference, thereby correcting the difference between the left- and right-side loudspeakers because of their distance to the listening position.

Given this correction, all that is necessary to establish the $\triangle P$ and $\triangle\phi$ values is to assign the relationship between $\triangle x$ and $\triangle\theta$ between the output signals from the left- and right-side loudspeakers — a simple implementation of the theory of $\triangle P - \triangle\phi$. The theoretical expression of $\triangle P - \triangle\phi$ can be easily derived by allowing for the head diffraction in the listener with respect to the output signals from the left- and right-side loudspeakers based on Figure 3.

$$\triangle P = 20\,\mathrm{Log}\,|\,Pr\,/\,Pl\,| \qquad (1)$$

$$\triangle\phi = \mathrm{Arg}\,|\,Pr\,| - \mathrm{Arg}\,|\,Pl\,| \qquad (2)$$

$$Pr = 1 + x^2 \cdot \gamma_\varphi{}^2 + 2x \cdot \gamma_\varphi \cos(\theta - \omega\tau_\varphi)$$

$$Pl = x^2 \cdot \gamma^2 + 2x \cdot \gamma\cos(\theta - \omega\tau)$$

Pr: Sound pressure applied to the right ear

Pl: Sound pressure applied to the left ear

3.2 Implementation method

The following paragraphs describe how the asymmetrical sound stage expander achieves the goals of distance correction and sound stage expansion:

3.2.1 Distance correction

The expander corrects the distance between the positions of the left- and right-side loudspeakers by adjusting the sound pressure level and the arrival time difference with attenuators and delay units connected to the right channel, so that the $\triangle P$ and $\triangle \phi$ values can be established from the $\triangle x$ and $\triangle \theta$ value alone.

At this time, an apparent move occurs in the sound image from the R position to the R′ position shown in Figure 2.

3.2.2 Sound stage expansion

After sound distance correction, expansion of the symmetrical sound stage using the theory of $\triangle P - \triangle \phi$ was attempted in the following procedures:

① Signal input is allowed only to the right channel.

② The $\triangle x$ and $\triangle \theta$ values set in every frequency band from this signal through attenuators and phase shifters is input to the left-side loudspeaker.

③ The output signals from the left- and right-side loudspeakers are imparted to the listener to effect an apparent move in the sound image from the R′ position to the R″ position in Figure 2.

④ Stereophonic reproduction is provided by the apparent sound image moving in the R″ direction and the left-side speaker.

3.2.3 Circuit blocks

Actually, due to the technical difficulty to set the $\triangle x$ and $\triangle \theta$ values electrically in all frequency bands, the frequency was divided into frequency bands with a band-pass filter (BPF) and the $\triangle x$ and $\triangle \theta$ values were set within each band. Figure 4 is a block diagram of the asymmetrical sound stage expander designed to provide stereophonic reproduction.

3.2.4 Setting of $\triangle x$ and $\triangle \theta$ values

Procedures for setting $\triangle x$ and $\triangle \theta$ values are described below.

① In an anechonic room in the measurement configuration in the block diagram shown in Figure 5, one real sound source is installed at the left- and right-hand loudspeaker positions and at position R″, which constitutes an imaginary sound source.

② Measurement is made of the frequencies versus $\triangle P$ and $\triangle \theta$ characteristics from each of the sound sources. The $\triangle P - \triangle \phi$ characteristics measured with a real sound source installed at position R″ are noted as R″ target characteristics.

③ Attenuation coefficient r and delay time t are derived by head diffraction at each frequency from these three classes of characteristic data.

④ The $\triangle x$ and $\triangle \theta$ values at each frequency are calculated by using the theoretical expression of $\triangle P - \triangle \phi$ as an asymmetrical condition.

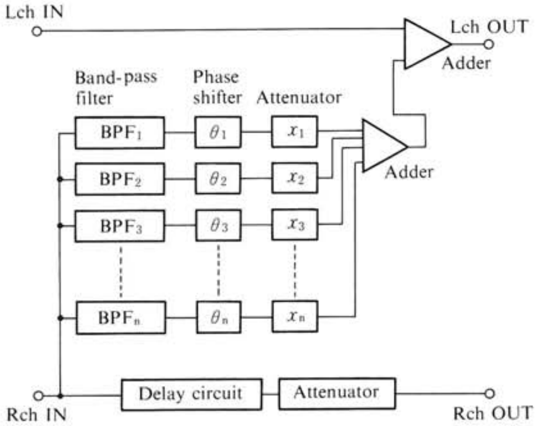


Figure 4. Asymmetrical Sound stage expander block diagram

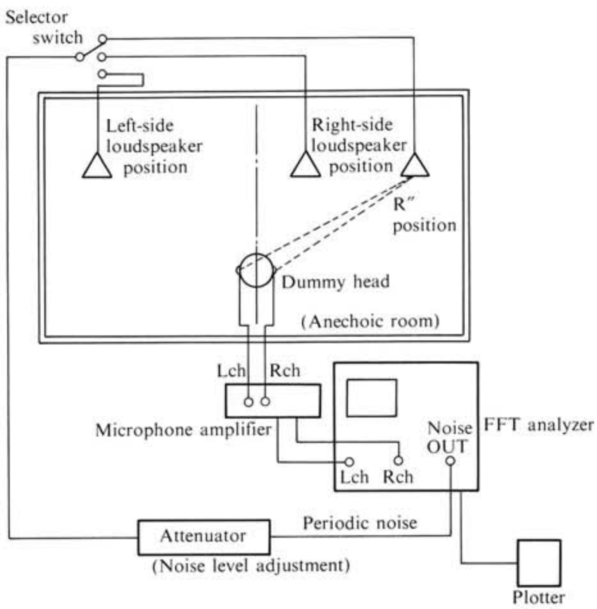


Figure 5. ΔP-$\Delta\phi$ measurement block diagram

The car compartment, smaller than a listening room and with specific building material properties, has more reflected waves with timing delays on the

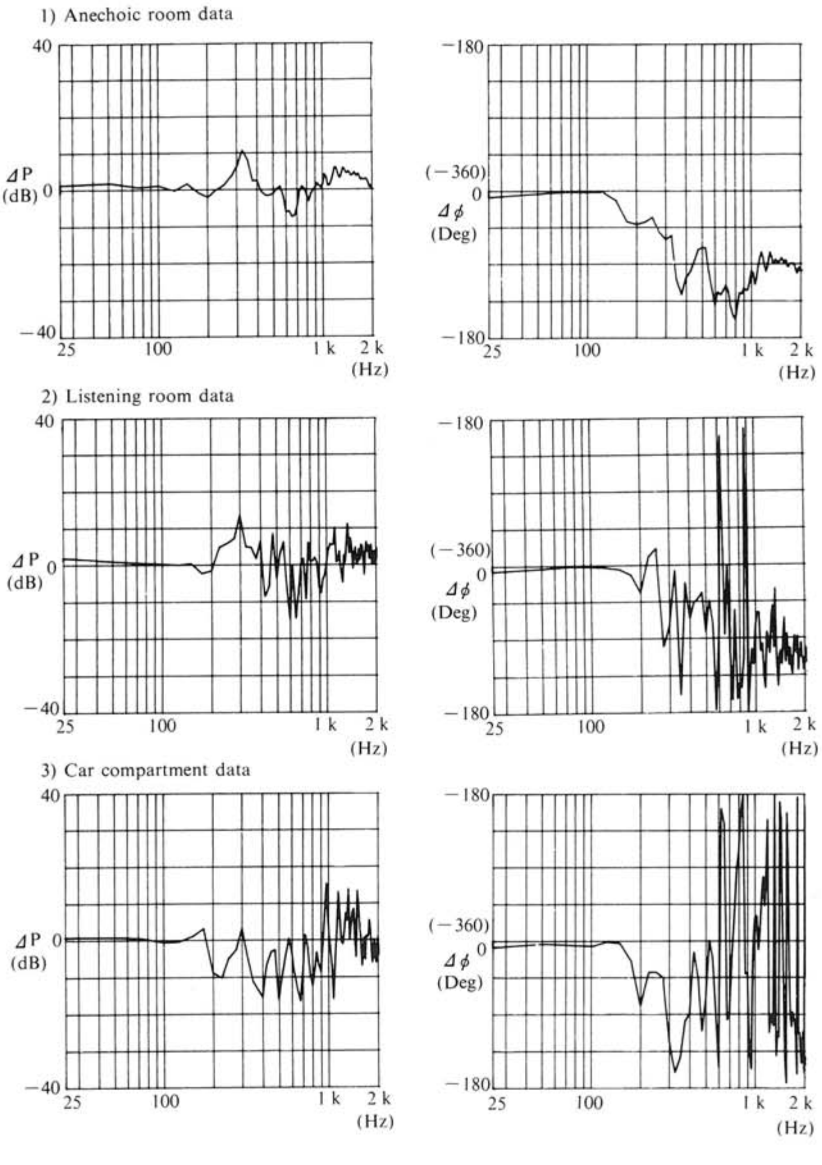


Figure 6. ΔP-$\Delta\phi$ data obtained with reflected waves

order of several milliseconds, with interfere with direct waves transmitted from the loudspeakers and thereby exert a major influence on the $\triangle P - \triangle \phi$ characteristics. Figure 6 shows significant changes in the $\triangle P - \triangle \phi$ values in the car compartment under the influence of reflected waves. The data was obtained by setting the $\triangle x$ and $\triangle \theta$ values derived from the theoretical expression in the asymmetrical sound stage expander and leading signals only to the right channel, then measuring the resultant $\triangle P - \triangle \phi$ values in an anechoic room, a listening room, and a car compartment.

It is extremely difficult to calculate the optimal values of $\triangle x$ and $\triangle \theta$ by factoring the influence of the reflected waves in the car compartment into the theoretical expression. The following methods, however, enabled the authors to derive the $\triangle x$ and $\triangle \theta$ values with relative ease to achieve sound stage expansion effects in the car compartment:

① To present the reference values of $\triangle x$ and $\triangle \theta$ in the attenuators and phase shifters in the asymmetrical sound stage expander.

② To install a dummy head at the listening position in the car compartment as shown in Figure 7 to observe $\triangle P - \triangle \phi$ characteristics obtained from the dummy head with an FFT analyzer.

③ To repetitively align the $\triangle x$ and $\triangle \theta$ values with the attenuators and phase shifters while observing the $\triangle P - \triangle \phi$ characteristics, to bring the $\triangle P - \triangle \phi$ characteristics as close as possible to the R″ target characteristics as shown in Figure 8.

Figure 9 shows the $\triangle P - \triangle \phi$ characteristics as controlled.

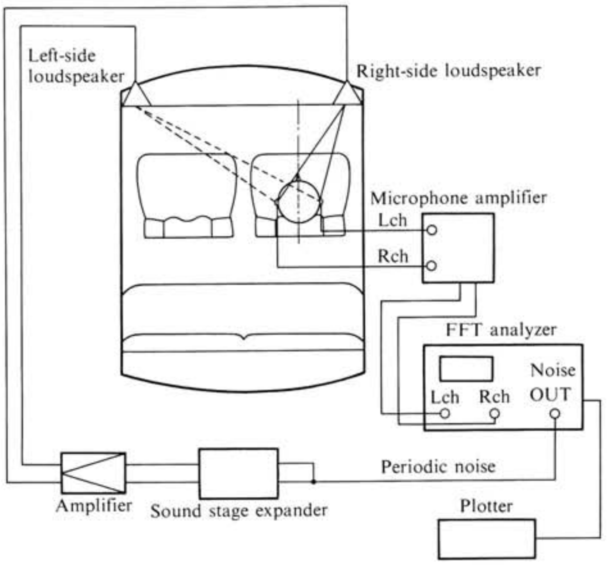


Figure 7. Car compartment ΔP-$\Delta\phi$ alignment block diagram

3.3 Effects of an asymmetrical sound stage expander

The authors verified the asymmetrical sound field correction effects of the sound stage expander through hearing tests. A description of the hearing tests, including findings, follows.

The tests were performed with regard to localization of the sound image at the right end by inputting signals only to the right channel with the listening position limited to the driver's seat, and the quality of the sound image (natural-sounding quality) and to displacements in the sound images to be localized at the front during stereophonic reproduction, and its quality. Table 1 details the test methods used. Tests were conducted by analysis of variance in a two-way layout. The effects of the asymmetrical sound stage expander were then verified by using test sources as parameters.

Figures 10 and 11 present analysis findings.

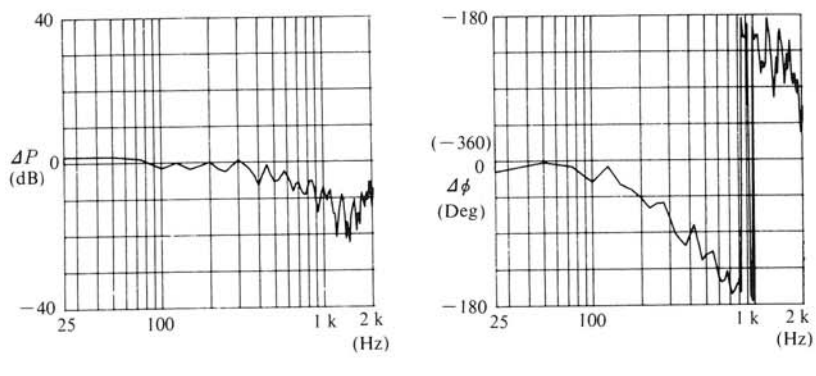


Figure 8. R″ target characteristics

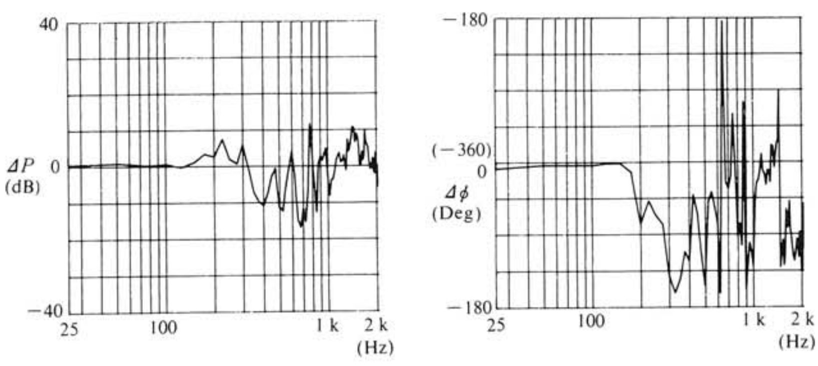
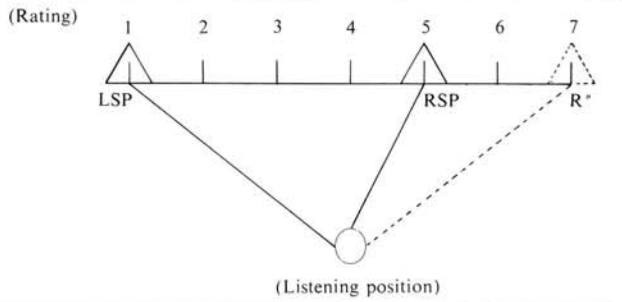


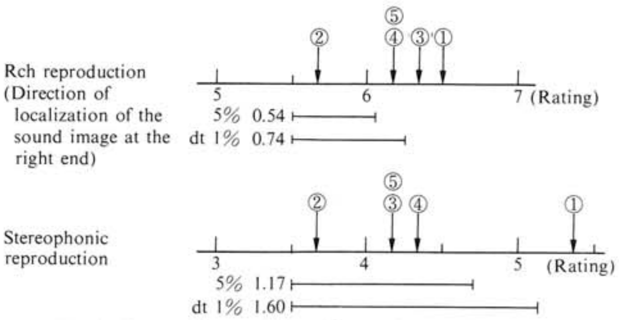
Figure 9. ΔP-$\Delta\phi$ data obtained with a sound stage expander

When signals were input only to the right channel, the asymmetrical sound stage expander was found to localize the sound image in the vicinity of the target R″ position. Thus, the findings demonstrated the localization of the stereophonic sound image in the right direction was confirmed, attesting to a well-balanced sound stage around the listener. The quality of the sound image was free from unnatural feeling.

The authors wish to carry on further studies to achieve well-balanced stereophonic reproduction in all the seats on the basis of these findings.

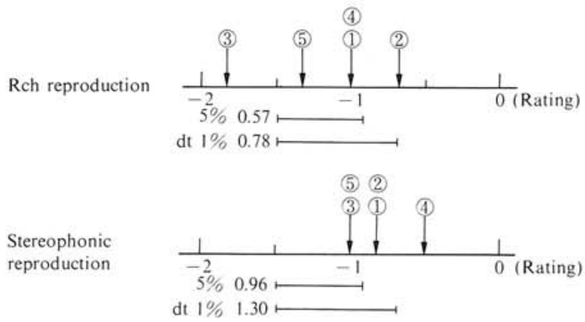
Table 1. Test details

	Description	
Listening condition	Driver's seat in the car compartment (right-side seat) (with doors and windows closed)	
Test items	Signal input only to the right channel (Rch)	1) Direction of localization of the sound image – Assessment of the direction of localization of the sound image at the right end 2) Quality of the sound image (natural-sounding quality) – Absolute assessment
	Stereophonic reproduction	3) Sound field balance – Assessment of displacements in the sound image to be localized in front center 4) Sound image quality (Natural-sounding quality) – Absolute assessment
Rating	1) Direction of localization of the sound image on Rch signal input 3) Sound field balance during stereophonic reproduction	(Rating) 
	2) Quality of the sound image on Rch signal input 4) Quality of the sound image during stereophonic reproduction	(Rating) −4 Unable to localize −3 Very unnatural −2 Unnatural −1 Somewhat unnatural 0 Natural
Number of judges	6	
Test sources	5 kinds 1) 500 Hz tone burst 2) 800 Hz tone burst 3) Drum solo performance 4) Chorus 5) Female vocal	



No significant difference if the difference in the rating is within dt.

Figure 10. Results of assessment of localization of sound images



No significant difference if the difference in the rating is within dt.

Figure 11. Results of assessment of sound images quality (natural-sounding quality)

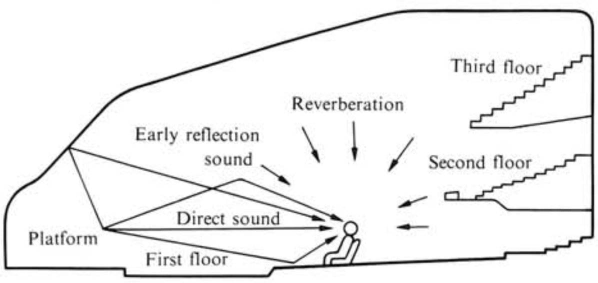


Figure 12. Cross-section of a concert hall

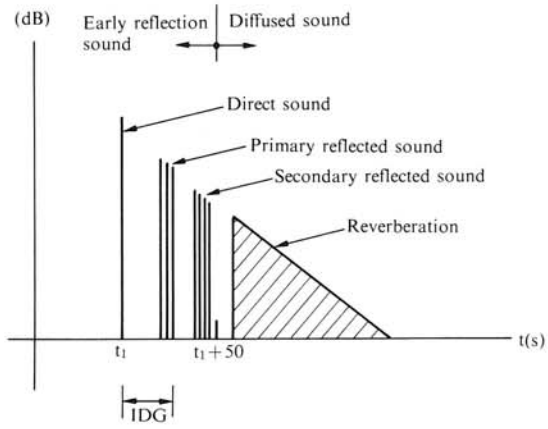


Figure 13. Structure of reflected sounds on a time base

4. Surround sound system

The surround sound system is known to give the listener a music reproduction more powerful than the usual reproduction and a feeling of being surrounded by music. Different manufacturers have different schemes for surround sound systems. The authors have examined the effects that an ideal surround sound system should provide and developed a system in line with this study. This system adds reflected sounds and reverberation available in concert halls or elsewhere to stereophonically reproduced sound fields, usually received through two speakers, to create auditory perspective and presence beyond the reach of conventional stereophonic reproduction. This chapter provides a summary of the new surround sound system, including its principles of operation.

4.1 Acoustic characteristics of concert halls

To give a spatial impression and presence to reproduced sound fileds, it is essential to note the architectural and acoustic characteristics of the

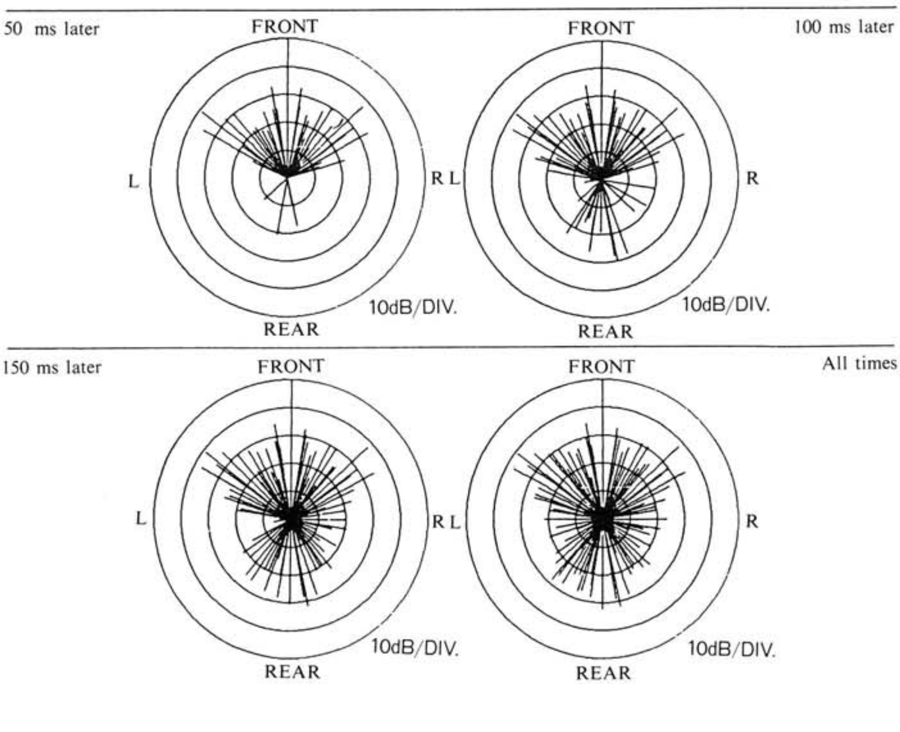


Figure 14. Pattern of directivity of reflected sound in Herkuressaal

Table 2. Reverberation times in representative concert halls

Hall name	Volume (m^3)	REV. time (s)
Boston Synphony Hall	19,000	2.4
Metropolitan Opera, New York	25,000	1.6
Carnegie Hall, New York	24,000	1.8
Philharmonic Hall, New York	24,000	2.2
Theater National de l'Opera, Paris	9,800	1.2
Teatre alla Scala, Milano	11,000	1.4
Wiener Staatsoper	12,200	1.5
Deutsche Oper Berlin	15,000	1.5
Grosser Musikvereinsaal, Wien	15,000	2.2
Salle Pleyel, Paris	27,000	1.6

"live" scene or concert hall in which music is played.

4.1.1 Kinds of reverberation in halls

Figure 12 shows a cross section of a concert hall. The sounds that are emitted from a sound source on the stage and that reach the audience can be thought of in three broad parts as follows:

1) Direct sound which reaches the audience first after being emitted from the sound source.
2) Early reflected sound, which is reflected from a reflector or the like at the rear of the stage, reaches the audience immediately following the direct sound.
3) Reverberation which is sound that reaches the audience from every direction as a result of complex reflections from the ceiling, floor, walls, and so on.

Figure 13 is a time-base structure of these three kinds of reflected sounds in a concert hall.

First, the direct sound comes at the position corresponding to the distance of propagation between the sound source and listener, followed by an early reflected sound 20-50 ms later, then reverberation. An analysis of the paths of arrival of reflected sounds in Figure 14 shows the 20-50 ms period in which early reflected sounds arrive at the audience is overwhelmingly dominated by forward components. Then, with the lapse of time, reflected sounds reach the audience from every direction.

4.1.2 Hall reverberation times

Figure 15 shows an actual measurement of impulse responses observed in a certain concert hall. The attenuation pattern is uniquely determined by the building material and geometry of the hall, and the location of the measurement equipment.

Reverberation time is defined as the time over which the energy of the sound emitted from a sound source in the stationary state is attenuated to 10^{-6} after the sound source has stopped[3]. According to Sabine, reverberation time is directly proportional to the volume of the room and inversely proportional to the room surface area and the average coefficient of sound absorption. Table 2 lists the reverberation times observed in representative concert halls all over the world. The reverberation time varies slightly with frequency. As shown in Figure 16, it is shorter in the high-frequency range than in the low-frequency range.

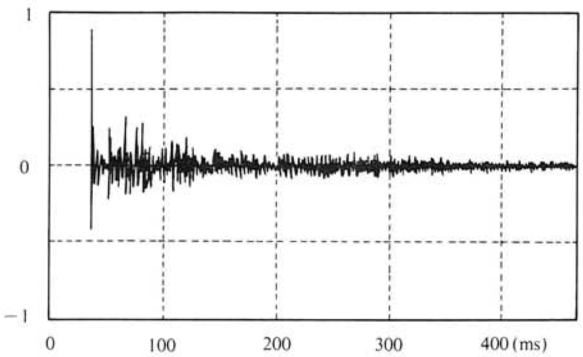


Figure 15. Example of impulse responses in the concert hall

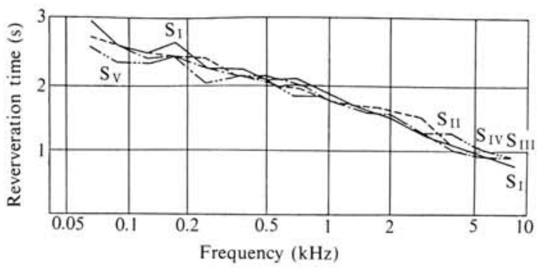


Figure 16. Reverberation response time characteristics

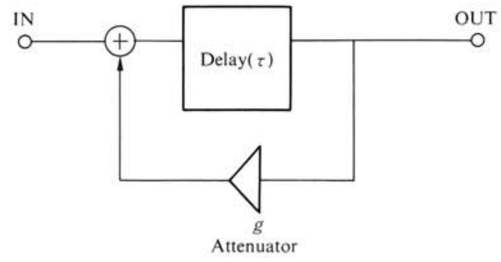


Figure 17(a). Reverberation model

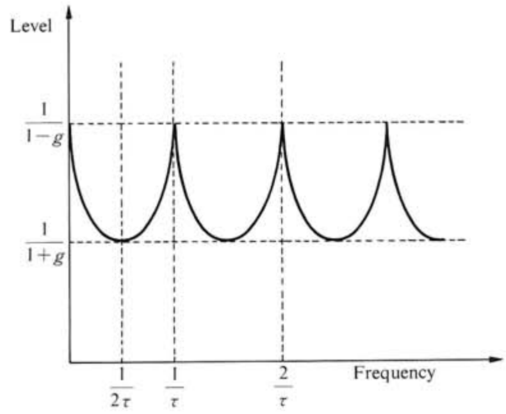


Figure 17(b). Amplitude-frequency characteristics

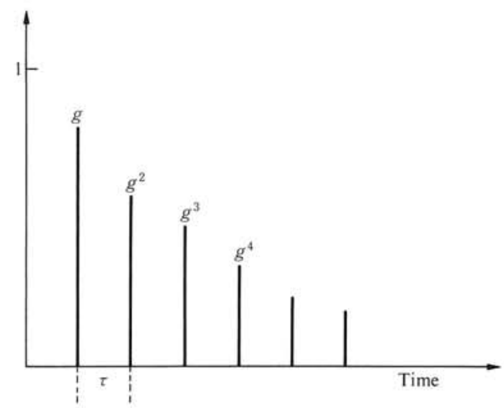


Figure 17(c). Impulse responses

4.1.3 Effects of reflected sounds

Called initial delay gap, the IDG in Figure 13 is a measure of the human awarenss of spatial impression. The IDG in a typical hall is 20-40 ms. In the car compartment, however, the IDG falls to a few milliseconds, interfering with the transmission of direct sounds and influencing the sound quality, localization, and so on. The concentration of early reflection sounds in the forward direction also plays a dominant role in providing auditory perspective.

These and other characteristics have a close bearing on the spatial impression of concert halls.

4.2 Generation of surround sound signals

Surround sound signals are generated with these characteristics in mind. This section describes how to generate reverberation and early reflected sounds, and the whole algorithm.

4.2.1 Generation of reverberation

Reverberation to be generated should satisfy, among others, the following requirements:

1) Flat frequency characteristics
2) Absence of periodic attenuation patterns
3) High-reverberation density
4) Sufficient reverberation times

Sounds radiated within a hall are attenuated in various directions and with various delays to produce reverberation as they are reflected from boundary surfaces (such as walls, ceilings, and floors). Then, a simple reverberation model can be constructed by allowing sounds to circulate through a delay unit and an attenuator as shown in Figure 17(a). On the basis of the definition of the reverberation time given earlier, if the delay time is τ (ms) and the damping coefficient is g, reverberation time T (s) is given by

$$T = \frac{-3}{\log_{10} g} \times \tau \times 10^{-3} \tag{3}$$

τ corresponds to the wall-to-wall propagation delay time of sounds, or the volume of the room, and g corresponds to the reflectivity of the walls.

By making these two parameters variable, var-

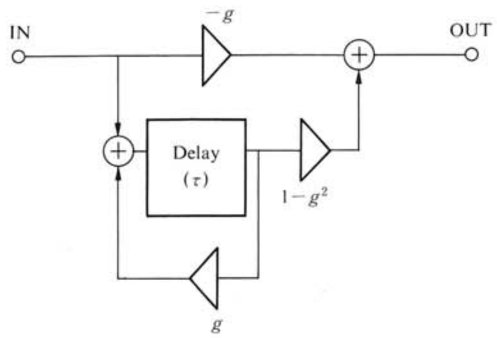


Figure 18(a). All-pass filter

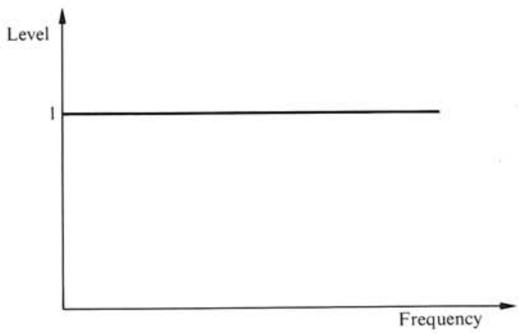


Figure 18(b). Amplitude-frequency characteristics

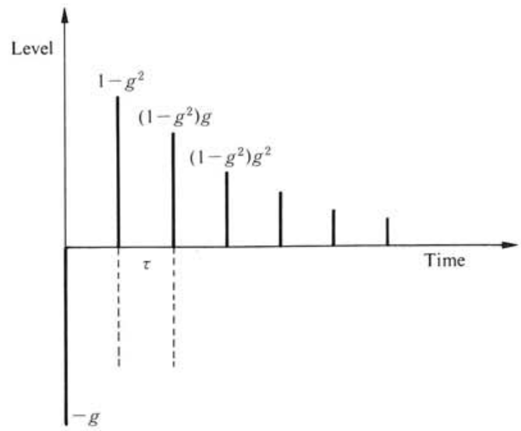


Figure 18(c). Impulse responses

ious kinds of reverberation can be generated. Obviously from Equation (3), there are a vast number of combinations of τ g that yield a given reverberation time T. The range of sounds that can be heard as natural reverberation can be empirically limited by the building materials and physical dimensions of the walls in actual halls.

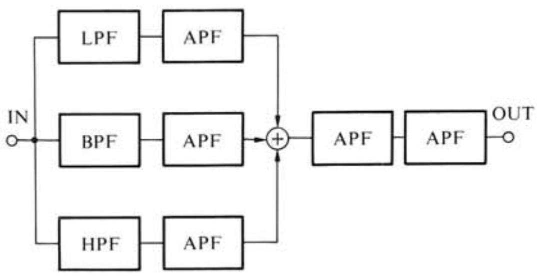


Figure 19. Reverberation generator

Table 3. Reverberation generator settings

Reverberation time	1.5 (s) [500 Hz]
Average reflected sound density	1,500 lines/s
Reverberation times by frequency band (100 Hz) (4 kHz)	 1.75 (s) 1.20 (s)
Crossover frequency	250 Hz, 2 kHz
Frequency characteristics	20~10 kHz

The feedback loop depicted in Figure 17(a) is called a comb filter. In the loop, frequency characteristics are not flat as shown in Figure 17(b). Therefore, an all-pass filter (APF)[4)] with identical attenuated pattern but flat frequency characteristics (Figures 17(c) and 18(c)) is used (Figure 18(a)). Figure 18 (b) shows characteristics of the APF.

The reverberation generator (Figure 19) breaks down input signals into three frequency bands according to reverberation time frequency characteristics as shown in Figure 16 and assigns reverberation characteristics suited to the specific bands to the signals through APFs. This is important to obtain natural reverberation. Processed signals are added, but are still incomplete as reverberation because of many periodic attenuation patterns involved in them. APFs in a later stage process the signals to have a detailed reflected sound density for output. Table 3 lists the settings of the reverberation generator.

4.2.2 Generation of early reflection sound

The structure of an early reflection generator is shown in Figure 20. The generator does not have a

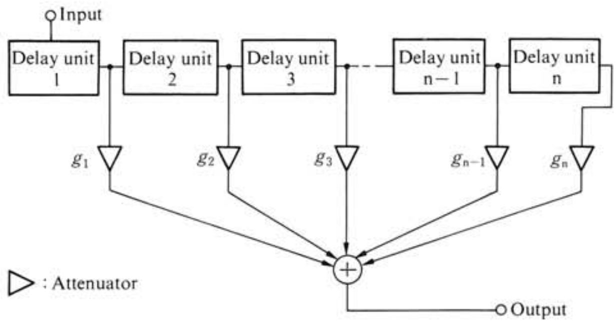


Figure 20. Early reflection generator

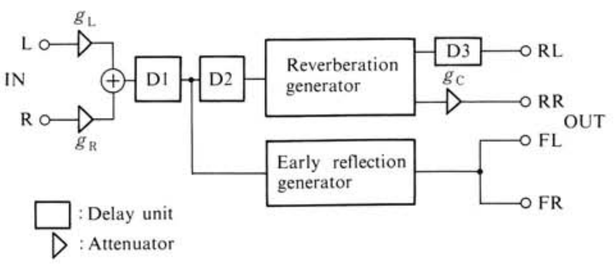


Figure 21. Overall algorithm

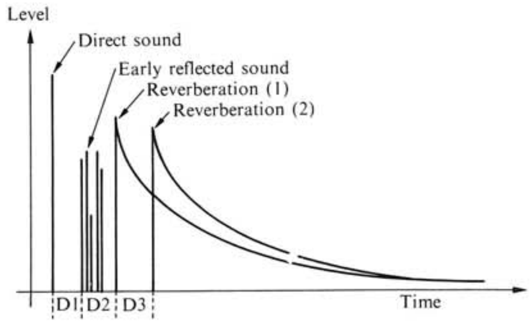


Figure 22. Output sound composition

feedback loop that is used for reverberation generation. Two or more delay units are cascaded and all outputs from connections between delay units have an attenuator. All the attenuator outputs are added to make the early reflection sound. Actually measured reflection patterns in halls shown in Figure 15 are used as reference. Total delay time is 10 to 30 ms. The number of delay units is 22.

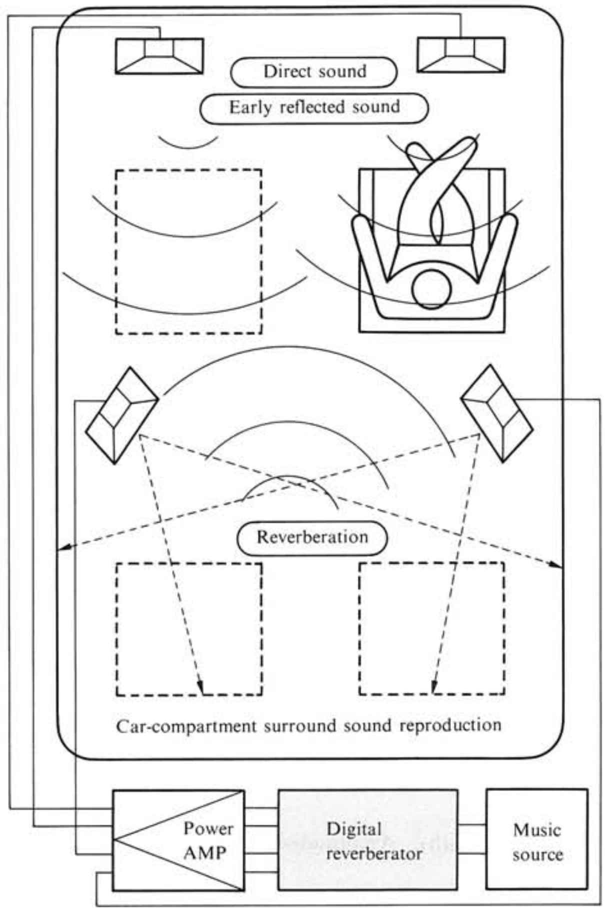


Figure 23. Surround sound reproduced in a car compartment

4.3 Overall algorithm of the surround sound signal generator

Figure 21 shows an overall algorithm of the surround sound signal generator. Each surround sound signal is produced in the following timing relation (Figure 22):

First, an early reflected sound is produced from the two forward channels with a delay of *D*1 after the direct sound. It is followed *D*2 later by reverberation from the rear left channel and *D*3 later by the same signal as the reverberation from the rear right channel. Parameter *D*3 plays a vital role in assigning a timing difference to output sounds and augmenting concert-hall presence.

4.4 Examples of surround sound effects

The output sounds generated are reproduced in the car compartment through four front and rear speakers as shown in Figure 23. Original signals and

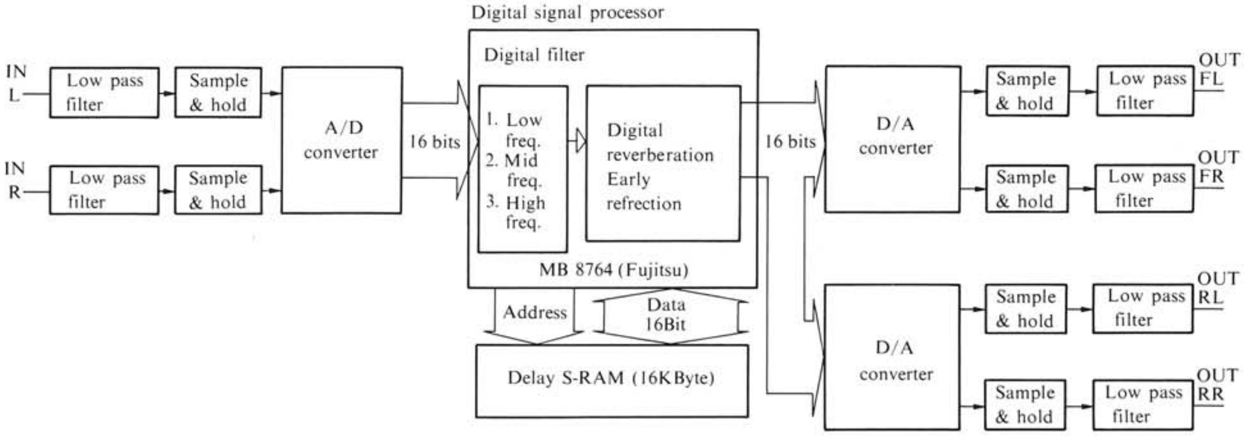


Figure 24. Hardware block diagram

early reflected sounds are output from the two front speakers, while reverberation is output from the two rear speakers.

The reason the early reflected sounds are output from the front speakers is to give natural auditory perspective in the direction of their arrival as shown in Figure 14[5].

The rear speakers are installed where their openings are not directly visible to the driver. They face the opposite rear windows so reverberation sounds will reach the listener from every direction. The extremely short reverberation time in the car compartment could make for control application as designed.

4.5 Implementation by digital signal processing

As stated previously, this surround sound system has delay processing as its primary function. In addition, it requires numerous multipliers and adders. If all their operations were to be executed by analog processing, a large circuit would have to be used. Besides, the bucket brigade device (BBD) used in delay processing has a weakness in its S/N ratio. For now, the authors resorted completely to digital signal processing (DSP) technology. DSP, when executed, must be implemented on a real-time basis, as man perceives sounds as consecutive sequences of information. The processing speed of a large computer is required. A general-purpose processing LSI, the MB8764 (Fujitsu) is used as the main processor[6]. A delay storage capacity of 16K bytes is used.

4.6 Hardware

Figure 24 shows the hardware configuration.

The two-channel analog signal is limited to the required band by a low-pass filter (LPF), then converted into a 16-bit digital signal by an A/D converter. The signal processed in DSP enters a D/A converter and generated through a LPF as an analog signal. Table 4 summarizes the principal specifications.

Table 4. Principal specifications of the surround sound system

Number of input channels	2
Number of output channels	4
Amplitude frequency characteristics	20 Hz − 10 kHz
Sampling frequency	24 kHz
Quantizing frequency	16 bits
Arithmetic cycle time	100 ns
Multiplier	$16\times16 \rightarrow 26$ bits
Delay memory	16k bytes
Processor	MB8764 (Fujitsu)
Reverberation time (500 Hz)	1.5 (s)
Processed sound equivalent volume	Approx. 5,000 m^3 (small hall)

5 Conclusion

Unlike the previously available techniques for enhancing electrical characteristics of single acoustic equipment, sound image control and surround systems introduced above aim at achieving auditory perspective, localization, presence, and other acoustic effects in stereophonic reproduction in a fully integrated audio system concept of acoustic equipment and its reprocducion space. Acoustic measurement technologies, which explore the behavior of sound in car compartments, and studies of the human audiology underlie these evolving control systems.

At present, the authors are working on the flow of sound particles based on studies of acoustic intensity (AI) relating to the direction of arrival of reflected waves and their amplitude in a car compartment using the closed located four-point microphone method, analyzing car-compartment sound field from various angles. It is necessary to continue studies to verify various physical data obtained from measurement against hearing parameters. These acoustic measurements should be handled as digital values in the light of the amounts of data processed and ease of recording. Also, the generalization of controls to extend to various kinds of car compartments would also greatly benefit from digital signal processing.

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