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SERA Super-live Sound System

"funky" Shock

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The SERA development project started with the selection of team members. When people with similar views and ideas work together, the team can be close-knit but the results ordinary. By contrast, if people with different views and ideas work together, the team may not be so close but it is possible to create something extraordinary. The SERA project team was of the latter mold and many non Toyota employees participated with the Toyota planning and engineering staff. I was especially impressed with the people in charge of development at Fujitsu Ten. When I proposed the "Live and Performance" concept, they endorsed my idea. Thanks to them, a Super-Live Sound had been realized in a small car. This has caused a "FUNKY" shock, brought about by DIGITAL SIGNAL PROCESSING.

When the vehicle development had proceeded to the final stages, a demonstration onboard for well known audio journalists, was scheduled. One day before the demonstration, however, the intended sound could not be reproduced. Fujitsu Ten and Toyota's engineers worked hard trying to find the cause, but could not. I was feeling uneasy but could only stand there doing nothing. Late into the night, the cause was found to be improper installation of the right and left speaker. The next day's demonstration was very successful, the system was highly praised by the journalists as a milestone in car audio field. For the people at Fujitsu Ten and Toyota, it must have been a very satisfying day.

When territory and colleagues are chosen too rigidly, it is difficult to work to a new set of values. A new alloy can be created by combining different ideas and characteristics.

We expect to have more "FUNKY" shocks.

● **Shigeki Kato** ● **Masaaki Hamai** ● **Takao Shimizu**

● **Keijiro Katsumaru**

In March 1990, In cooperation with Toyota Motor Corporation, we developed the "Super-Live Sound System", installed in the new Toyota SERA. Our system uses a Digital Signal Processor (DSP) to control the sound field so that it gives listners a "presence" as compared with conventional car audio equipment. We measured sound characteristics of various types of halls as well as vehicle interiors to design the speaker arrangement and digital sound control. We also achieved the natural reproduction of original or unprocessed sound. The main unit, power amplifier, and speaker system were designed to give high quality of reproduction. As a result, we produced the new audio system that fits in with SERA's concept "Live and High Performance".

1. Introduction

Recently, the popularity of car audio equipment has grown remarkably, New systems are constantly released to try to meet the consumers desire for diverse and high-quality equipment.

Various new technologies have been introduced to improve sound quality, aiming for maximum sound pressure level, flat frequency characteristics, and for expanding frequency range.

A car interior is small and has unusual sound characteristics, therefore car audio equipment has difficulty in giving the impression of "presence." Until now, few system which achieved improved "presence" have been developed.

In March 1990, we, in cooperation with Toyota Motor Corporation, completed development of the "Super-Live Sound System."

This audio system has become a part of the new Toyota SERA. Unlike any conventional car audio equipment, the system uses a digital sound processor (DSP) to control the sound field. The music from this system is so realistic that our system gives a feeling of "presence" and "depth."

Here, we mainly expand on the following aims intended to provide the most effective audio reproduction with the SERA "Super-Live Sound System."

① Integrated design tailored to the car interior sound characteristics.

② To improve the quality of direct sound reproduction, we also give some quantitative analysis results.

2. Development background

2.1 Live & performance

In 1983, Toyota Motor Corporation organized a study group called the Young Project. They aimed to develop new products for the young or next generation by analyzing the recent trends towards individuality and diversity. Various studies have concluded that the characteristics of typical young people are as shown in Table 1.

Table1. Characteristics of young people

• Eat and dress well	• Unpredictable	Live
• Interested in Audio and video	• Want to be entertained	
	• Like the unusual	
	&	
• Catalog consumers	• Like to be conspicuous	Performance
	• Want to be someone else	

The development concept of SERA, "Live & Performance," evolved from these traits. The audio should give the dramatic impression of being at a live performance. The theme progressed to a complete car with a dome-shaped passenger compartment, expansive windows, and gull-wing doors. Fujitsu Ten was given the important task of cooperating with Toyota to develop the audio equipment.

2.2 Casual & funky

Music is essential part of life for young people. In the "Live and Performance" concept, the SERA audio equipment was considered one of the most important features.

By 1988, Fujitsu Ten had developed a practical sound field control system that gave the impression of "presence." This was done by using a digital signal processor (DSP) to artificially add reflected sound.

This system which can produce a sound field with "presence" is ideal for the concept of the SERA. It was decided to install such an audio system, using a DSP, in the SERA as the first car on the market.

The sound field control system can create various effects. For the SERA, we chose two and named them, after their image, as "Casual mode" and "Funky mode." The driver can choose between these modes with the sound warp button.

3. System

3.1 System configuration

Figure 1 shows the system block diagram of the SERA Super Live Sound System. The system consists of a main unit, power amplifier unit, front loudspeakers, a center loudspeaker, rear loudspeakers, and a woofer speaker.

The main unit has a CD player with FM/AM tuner and a pre-amplifier integrated into a 1-DIN size unit.

The power amplifier has a sound field controller using a DSP and an equalizer to correct the acoustic characteristics of the car interior. The power amplifiers has five channels including one channel to drive the center loud speaker.

The two-way front loudspeaker consist of a 12-cm full-range and a 2.5 cm tweeter.

The center loudspeaker is a 8-cm full-range (inverted-cone).

The box type two-way rear loudspeaker consist of a 10-cm full-range speaker and a 2.5-cm tweeter.

The sound warp button changes the DSP function and the axes of rear loudspeakers. The speakers incorporate a mechanism with which the speaker axes are changed by a motor. The Acoustic Resonance Woofer (ARW) has a 12-cm drive unit and its own power amplifier.

Figure 2 shows the system component layout in the SERA.

3.2 Main unit

3.2.1 Specification

The 1-DIN size main unit consists of a CD player

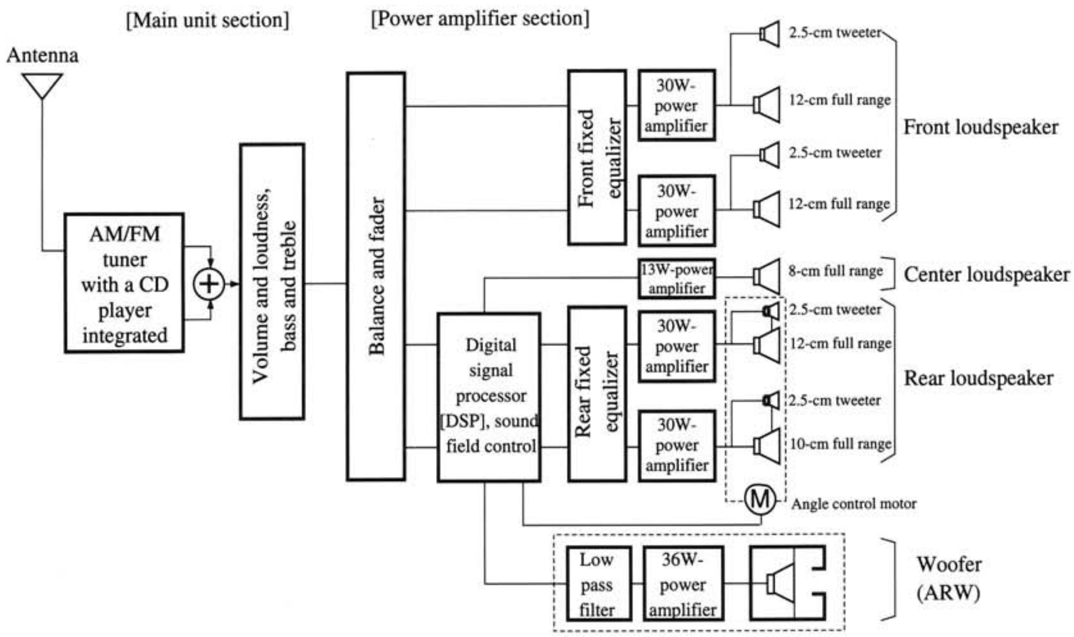


Figure 1. System block diagram

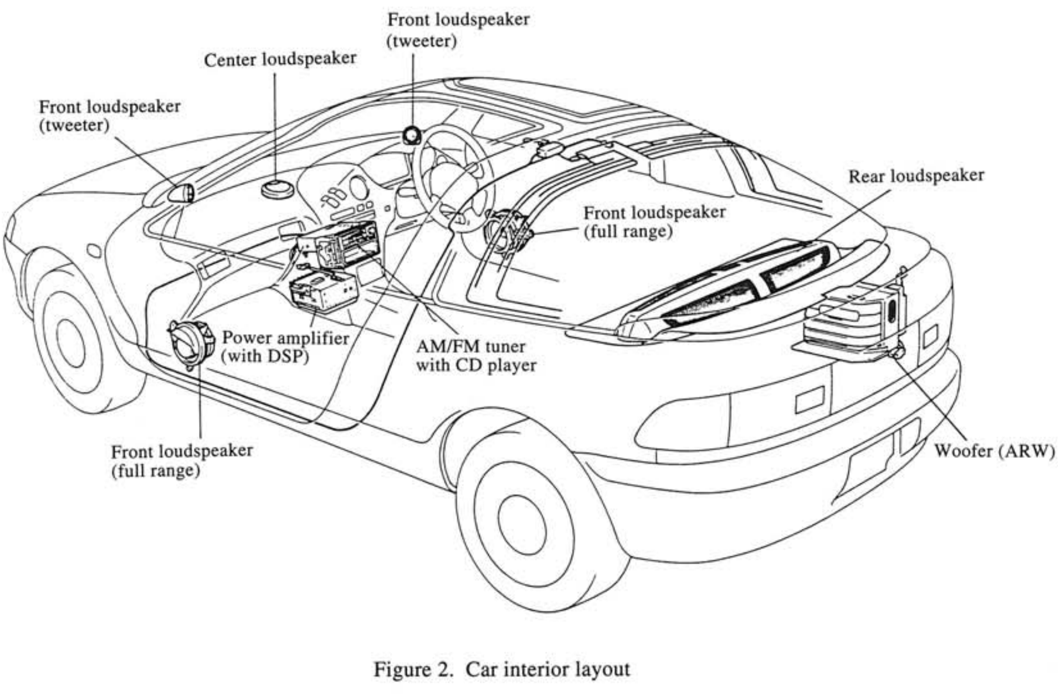


Figure 2. Car interior layout

(which can play 8-cm and 12-cm discs) and an FM/AM tuner. Figure 3 shows its front panel. Its functions are:

(1) DSP sound warp function

The major innovation in this system is the sound field control using the DSP. Pressing the sound warp button switches the sound field to either "Casual mode" or "Funky mode." The user can vary the sound effect by pressing the level button.

(2) Direct change of source

When ACC pow is ON, the sound sources (CD player, tuner, or optional cassette deck) can be directly selected at any time by pushing one of the source select buttons.

(3) Illumination of buttons by function

Only the buttons to be used in the current mode (CD or tuner) are illuminated, making the button selection at night easier.

3.2.2 Key technology

We have introduced the latest technology make highly functional and high-performance product with good appearance. The key technologies used are as follows.

(1) To make the product compact

Some conventional CD player systems consist of a 1-DIN size CD player unit and a radio and cassette deck in another unit. Some combine all three in a 2-DIN size unit. In our systems, the CD player unit had to include a tuner, tone control function, DSP control function, antenna jack, and a DIN connector for the cassette deck.

Therefore, making the product compact was very important.

The key points making the product compact were as follows:

① In the tuner, circuitry from the RF to the audio output is integrated as a front-end.

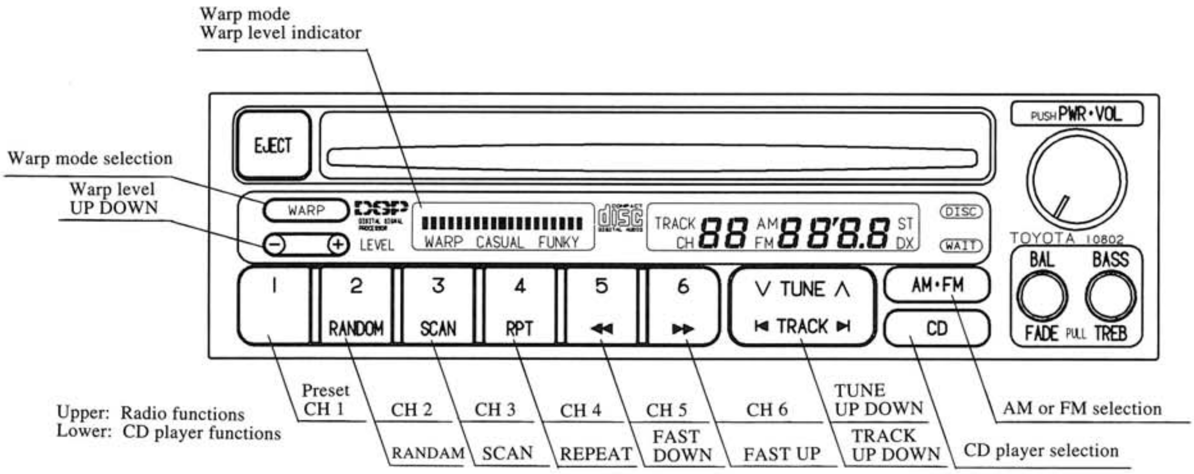


Figure 3. Front panel of main unit

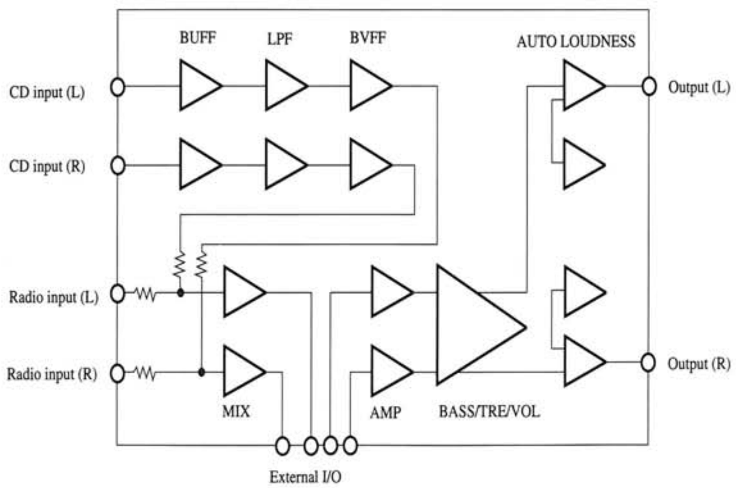


Figure 4. New preamplifier HIC

② The antenna jack and cassette deck DIN connectors are vertically installed to minimize the space for installation.

③ The compact disc player LPF and the audio electronic variable resistor circuit use HICs. (see Figure 4)

④ The front panel has a duplicated PC-board-structure combining the display and display driver sections. We adopted this structure for accommodating many function buttons and displays that are needed for increased functions. (See Figure 5.)

⑤ The depth of the deck section was shortened by 7 mm by using small LSI chips, surface-mount devices (SMD), and 250-μm PC board patterns.

(2) For achieving high sound quality

For effective sound field control using a DSP, high-quality play-back of the original, unprocessed sound is important. We used the following techniques for obtaining high sound quality:

① Reproduction of high tones was improved by using a four-times over-sampling digital filter in the CD player.

② We reduced noise by using low-noise-type electronic variable resistor ICs (4 dB lower than conventional variable resistors) and high-performance operational amplifiers and by minimizing the lengths of the PC board patterns for low-level-signal circuits.

③ We used a low-tone resonance type auto-loudness circuit with a semiconductor inductor to correct the loudness control curve at low sound volumes.

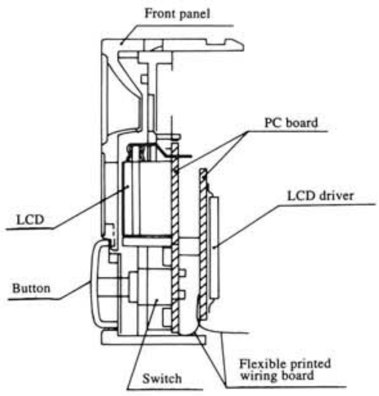


Figure 5. Front panel structure

3.3 Power amplifier

3.3.1 Specification

Table 2 shows the major specifications of the power amplifier.

Table 2. Power amplifier specifications

Parameter	Front	Rear	Front
Maximum output	30 W	30 W	13 W
TED (1W)	0.04 %	0.04 %	0.04 %
Frequency range	10 to 50 kHz	10 to 50 kHz	10 to 50 kHz
S/N ratio (flat)	70 dB	70 dB	70 dB
Channel separation	60 dB or more	60 dB or more	60 dB or more
Load impedance	4 Ω	4 Ω	4 Ω

3.3.2 Circuit configuration

The power amplifier consists of a fixed equalizer, power amplifier, DSP, control section, and power supply. The following describes the circuits, mainly the DSP which is the key device of this system.

(1) We used a 24-bit floating point operation DSP LSI chip, made by Fujitsu Ten, in the digital signal processor. This processor can generate a fine echo sound and reverberations.

(2) We used the 1-bit MASH method in the A to D converter, this provides the digital signal to the DSP input. This method improves the S/N ratio and lowers distortion.

(3) We used a four-times over-sampling digital filter in the D to A converter, which converts the DSP output into an analog signal.

(4) We used low noise operational amplifiers, polypropylene film capacitors, and special electrolytic capacitors for audio circuits in the LPF and analog MIX circuits around the DSP. We achieved a wide dynamic range by using a dual rail power supply, obtained from a DC to DC converter, for the linear circuit.

(5) We used an 8-bit microprocessor to control the DSP. This processor serially transmits button operation, signals and display information to and from the main microprocessor in the main unit to control the audio system.

3.3.3 Innovative design features

The following explains the innovations in designing the audio system using a DSP.

(1) Use of a volume limiter for noise reduction

The maximum input level of the power amplifier is

0 dB which is the output level of the CD player at maximum volume.

However, the average signal level of CD music is rather low. Matching the maximum CD output level (0 dB) to the maximum input level (all bits 1) of the A to D converter, does not give the best S/N ratio and lowest distortion at usual operation levels. We raised the output signal level of the CD to give the maximum (full scale) input level for the A to D converter at a CD volume position −10 dB lower than the maximum position. This volume margin improved the S/N ratio and lowered distortion.

Input signals from CD exceeding −10 dB are suppressed by a volume limiter circuit whose input/output characteristics are shown in Figure 6. This avoids overdriving the A to D converter. In the DSP block, signals are attenuated after the D to A convertion so that the gain of the DSP block is kept at 1 when a CD signal level is −10 dB or lower. This attenuates noise generated after A to D conversion and improves the S/N ratio.

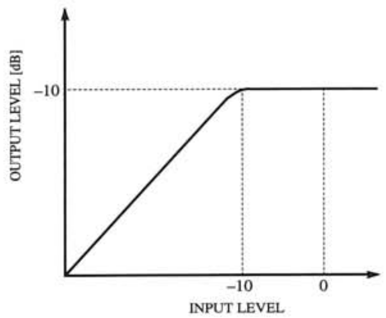


Figure 6. Volume limiter amplitude response

(2) The Hybrid IC DSP

Clock pulses include high frequency components, which disturb or affect the tuner by propagation or radiation.

For noise reduction, we integrated the DSP, crystal oscillator, and memory into HICs. The following are key considerations when using HICs.

① Using HICs reduces the parts mounting area. As a result, energy radiated from parts and wiring patterns will be reduced. The interference between circuits or parts is also reduced.

② Using four-layered PC boards allows using larger ground patterns which reduces the grounding impedance in high frequency range.

③ Noise propagated from HIC pins can be reduced by using beadfilters on each HIC pin.

These techniques reduced noise by 10 to 20 dB.

3.3.4 Inspection of sound field control

Whether the DSP is controlling sound field normally can be checked using the transfer function of the digital signal processing section. In our audio system, echo sounds are generated by using a combination of many delay circuits and each output has a unique phase versus frequency characteristic.

Even for the same output line, the phase characteristics change when sound field control patterns are switched between "Casual mode" and "Funky mode."

For each output line, a signal frequency at which the signal level difference and phase difference are obtained is measured.

A sinusoidal-wave signal having a measured frequency is input to the power amplifier to check the output level of the amplifier.

Figure 7 shows the test set up. Standard test equipment can be used.

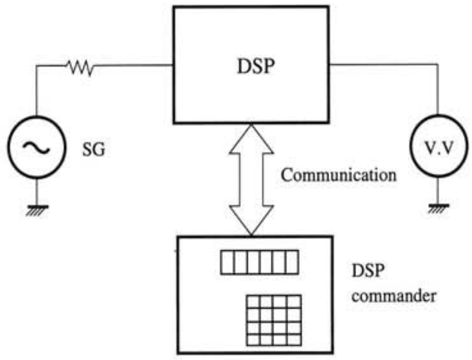


Figure 7. Testing the DSP circuit

3.4 Loudspeaker system

The SERA "Super-Live Sound System" includes full range loudspeakers having pulp-woven cones and tweeters with a polyimide diaphragm. These components were used in the CELICA "Super-Live" Sound System" to give high-quality sound reproduction. This section explains new technologies used in the SERA speaker system.

3.4.1 Woofer loudspeaker

For realistic music play back, bass sound reproduction is very important. However large speakers required for this purpose are difficult to install in a compact car like the SERA. Most audio systems for this size of car do not reproduce bass sound well.

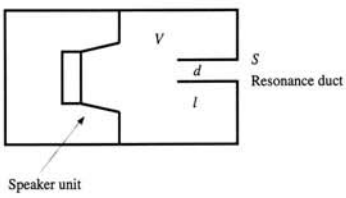
For good bass sound reproduction, the SERA has adopted a Kelton type Acoustic Resonance Woofer (ARW) which has the basic structure shown in Figure 8. It generates sounds using air in a duct as a virtual diaphragm, known as Helmholtz's resonance phenomena. The ARW has the following features:

(1) Compact and highly efficient: Over the specified frequency range, the 12-cm loudspeaker with a 7-liter box has the same bass reproduction as a conventional 16-cm loudspeaker with a 20-liter box.

(2) Low distortion: Distortion is low because the air in the duct serves as the speaker diaphragm.

Figure 9 compares sound pressure frequency characteristics of a 12-cm loudspeaker and that of a ARW in an anechoic chamber. The average improvement in the efficiency is 6 dB for the specified frequency range.

In the SERA, the desired overall audio characteristics and improved bass response have been achieved by matching the ARW acoustic pass band to the



$$\text{Resonant frequency } fr = \frac{C}{2\pi} \times \sqrt{\frac{S}{l\,e\,V}}$$

l: Duct length
S: Duct cross sectional area
d: Duct diameter
V: Speaker-box capacity
C: Acoustic velocity
le: Effective duct length (l+0.8d)

Figure 8. ARW structure

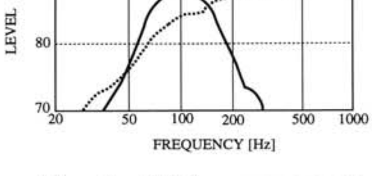


Figure 9. ARW frequency response

acoustic transfer characteristics of the car interior.

3.4.2 Center loudspeaker unit

For effective sound reproduction, an additional speaker unit is installed near the center of the instrument panel. An inverted-cone-type speaker is used because of the space limitation.

Figure 10 shows a cross section of the center loud speaker. An inverted cone speaker is about 60% of the thickness of a conventional speaker of the same diameter. This speaker, having wide directivity angle, gives natural sound reproduction.

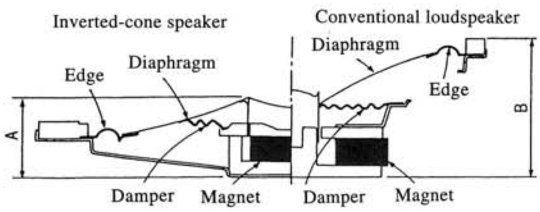


Figure 10. Center loudspeaker structure

3.4.3 Rear loudspeakers

The rear loudspeaker characteristics are as follows:

The speaker orientation changes from the upper to the lower position when switched from the Casual to the Funky mode (Figure 11).

The speaker orientation is switched to give the effect of "presence" by spreading the sound reflected by the rear window or for reproducing direct sound from the speaker. This function helps the DSP have effective sound field control. By considering the angle of the rear window glass, the angle of the orientation change was determined to be 45°. The transition time between the two position was chosen to be two seconds for a smooth switching between the modes. The speaker position is switched by a motor using a belt and speed reduction gears. The following were the requirements for the driving mechanism.

(1) The load on the motor may change because the center of gravity of the loudspeaker shifts when the speaker moves. This must not affect the rotation speed of the motor and the transition time must always be two seconds.

(2) The drive mechanism must withstand the environmental conditions in the car, particularly changes in temperature and vibration.

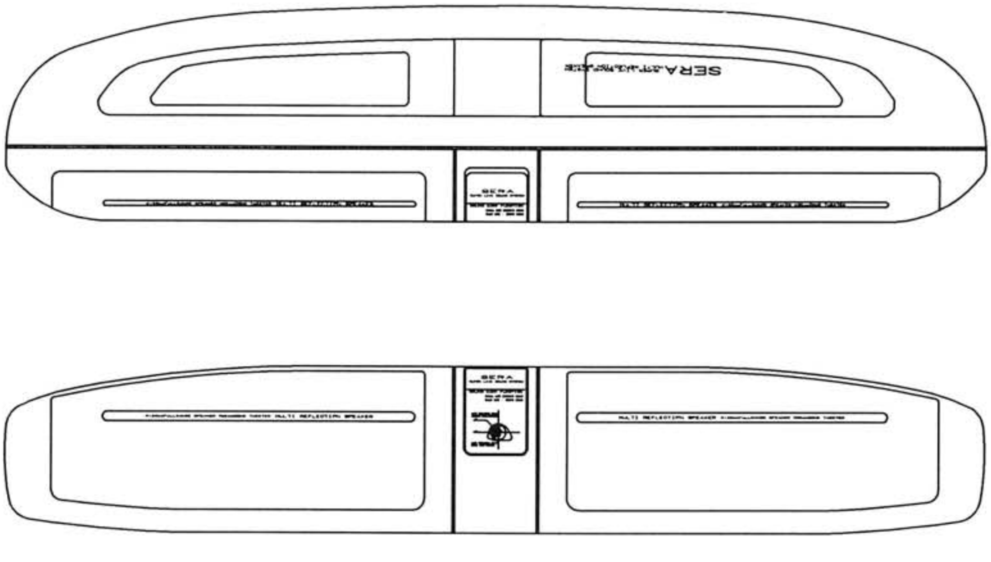


Figure 11. Rear loudspeakers

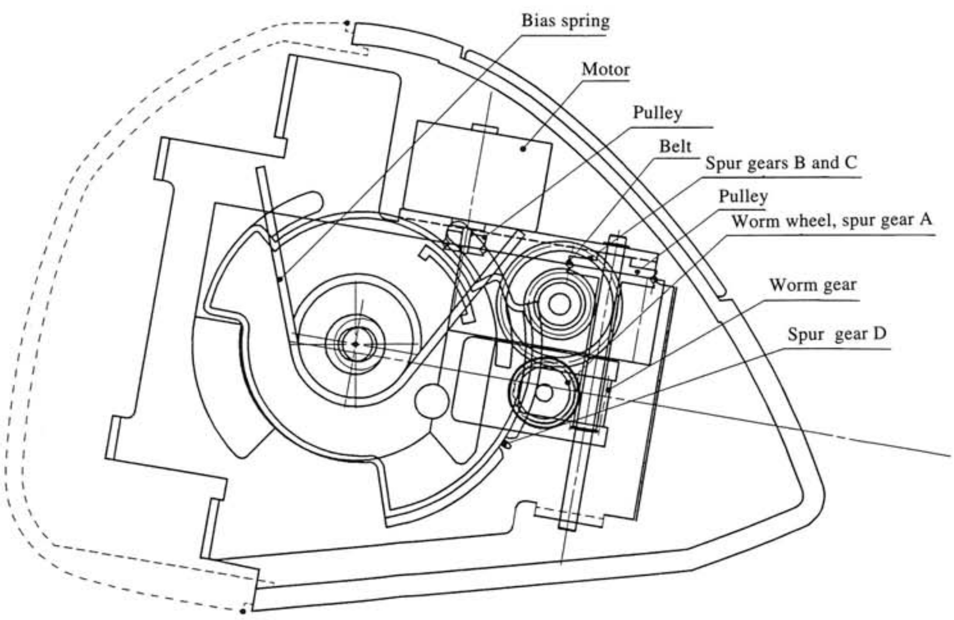


Figure 12. Rear loudspeaker drive mechanism

As the drive motor, we adopted an electronic governor motor which has shown good results in cassette decks, and which can cope with the environmental conditions in the car. This motor keeps a constant speed if the load is within a certain range.

To protect against vibration, mechanical bias is added at the each speaker position (Figure 12). This should prevent the speaker from moving the normal positions and gear backlash caused by vibration.

4. Car interior sound field control

In our previous reports, we explained the difference between the acoustic properties of a car and that of a concert hall. We suggested a best method for reproducing sound in a car. Here, we focus on the features of the SERA sound field control system.

4.1 Configuration of SERA sound field control system

The design concepts for the SERA sound field control were as follows:

(1) To create "presence" and "depth" that could not be obtained with conventional car audio equipment.
(2) Attaching great importance to the quality of original, unprocessed sound.
(3) Front sound-image localization
(4) To determine the control prarameters based on acoustic data obtained in concert halls
(5) Effective reproduction of many types of music and simplified operation by minimizing the number of audio modes to two and making only the effect sound level variable

With all these aims, the sound field control system shown in Figrure 13 was configured.

The FT 8800 DSP, which is manufactured by Fujitsu Ten, is the main processor. Table 3 gives the control specifications for the two sound field control modes.

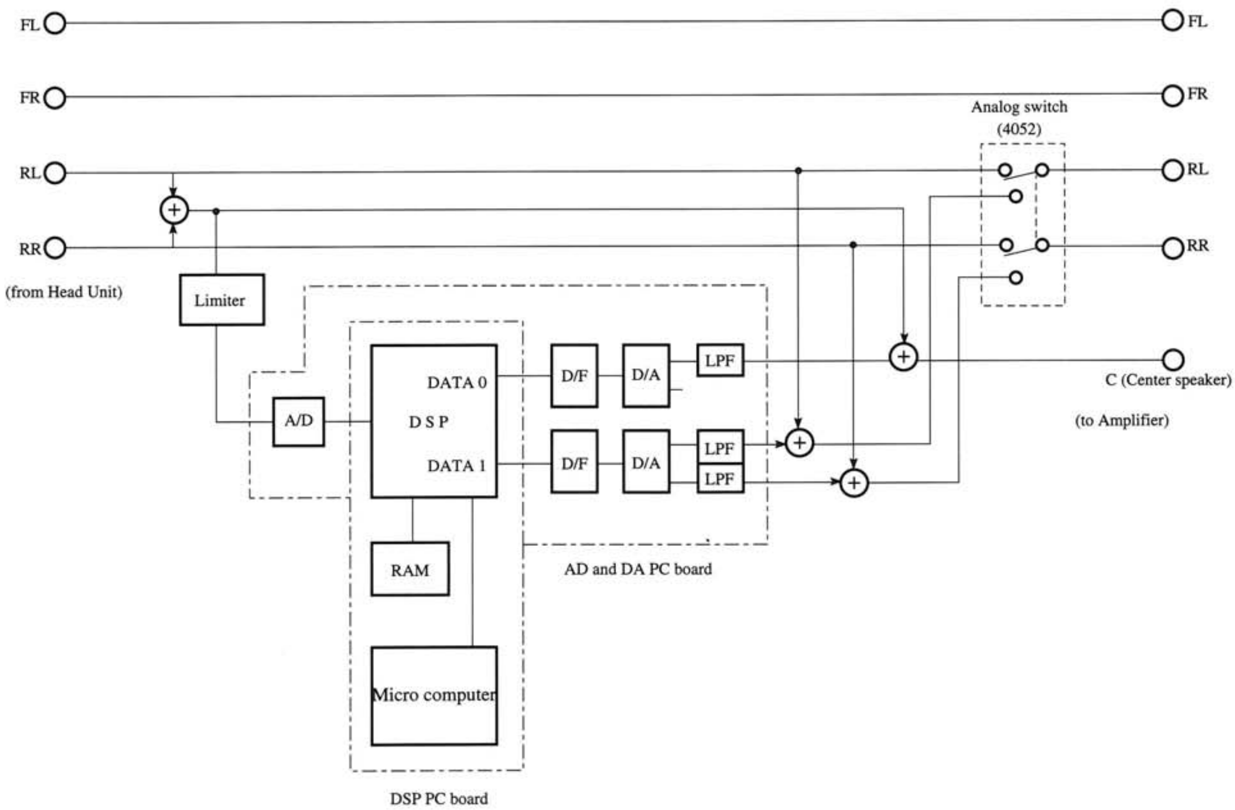


Fig 13. Block diagram of digital signal processing

An additional loudspeaker, for effective sound in the cabin, is installed in the center of the instrument panel. It has the following feautures.

(1) Reproduction of "depth" in front of the listener

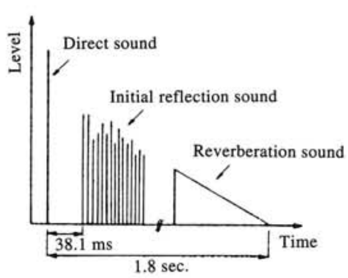
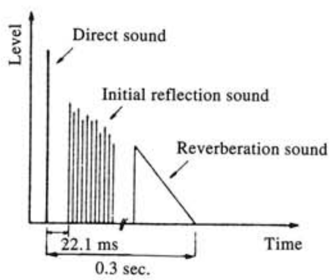
(2) Correction of right and left imbalance caused by listening from a shifted position

4.2 Evaluating the sound field control system

4.2.1 Measurement and evalution

The SERA sound field control effect was measured using the proximity four-point microphone method. This method measures reflected sound direction, strength, and time dependence on the sound reception point. The difference in impulse response measured with four microphones located in close proximity give the required information. This measurement method was developed by the Science and Engineering Research Institute of Waseda University.

Table 3. Sound field control specifications

			Casual mode	Funky mode
Intended sound			• Flat and wide range sound reproduction • Audio balance espeically voice sounds • Audio impression of being at a concert hall with the appropriate echo	• Tight and powerful bass reproduction • Audio balance like being gripped by the sound • Audio impression of being in front of a live-stage
Audio signal processing	Front loudspeaker	Signal	Direct sound	←
		Level	0 dB	←
	Center loudspeaker	Signal	Initial reflected sound + Reverberation sound	Direct sound + Initial reflected sound + Reverberation sound
		Level	—	—
	Rear loudspeaker	Signal	Reverberation sound	Initial reflected sound +Reverberation sound
		Level	–4.5 dB (UP)	–4.5 dB (DOWN)
	Woofer loudspeaker	Signal	LPF: 150 Hz, –12 dB/oct	LPF: 150 Hz, –12 dB/oct
		Level	+6 dB	+10.5 dB
	Initial reflected sound delay time		38.1 ms	22.1 ms
	Reverberation time		1.8 sec.	0.3 sec
Nature of reflected sound				

The following shows the directivity pattern and the impulse response of reflected sound obtained with this measurement at the SERA driver's seat.

Figure 14 shows the results with no digital signal processing.

Figure 15 shows the results in the Casual mode, with long reverberations.

Figrue 16 shows the results in the Funky mode, with short reververations.

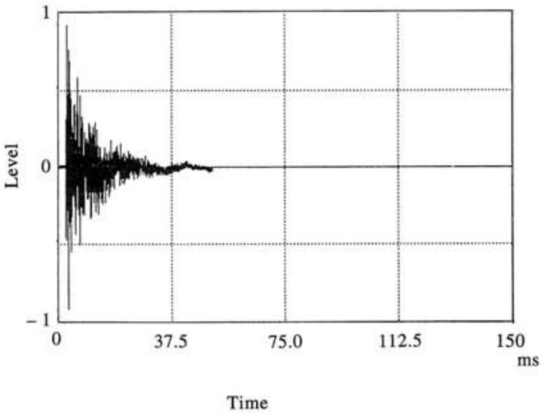


Figure 14. Impulse in the car interior (stereo reproduction) "DEFEAT"

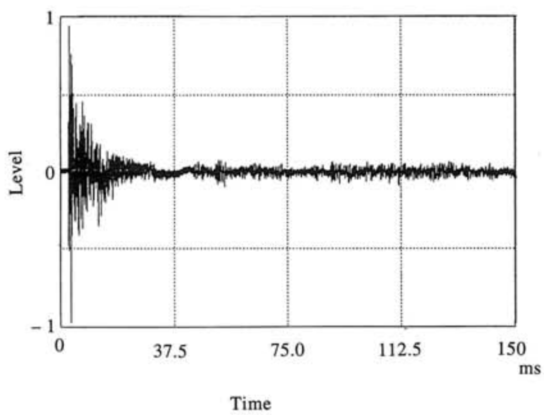


Figure 15. Impulse response in the car interior "CASUAL mode"

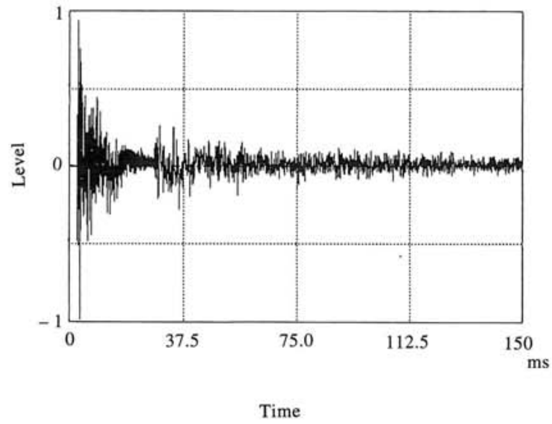


Figure 16. Impulse response in the car interior in "FUNKY mode"

With no sound field control, the impulse response in the car interior converges in about 40 ms (Figure 14), a remarkably short reverberation time. The initial reflection sound comes immedeiately after the direct sound. The directivity pattern shows that reflected sound concentrates at the loudspeakers and is highly unbalanced (Figure 17).

With the sound field control, the inpulse response is as follows:

- Initial reflection sound is added about 40 ms after the direct sound in the Casual mode (Figure 15)
- It is added about 20 ms after the direct sound in the Funky mode (Figure 16)

This clearly shows the effect of the sound field control and the difference between the two modes. Also, the directivity patterns (Figure 18 and 19) show that reflected sounds come from many directions and are better balanced.

These measurements were performed with our initial setting level of effect sounds for the sound field control.

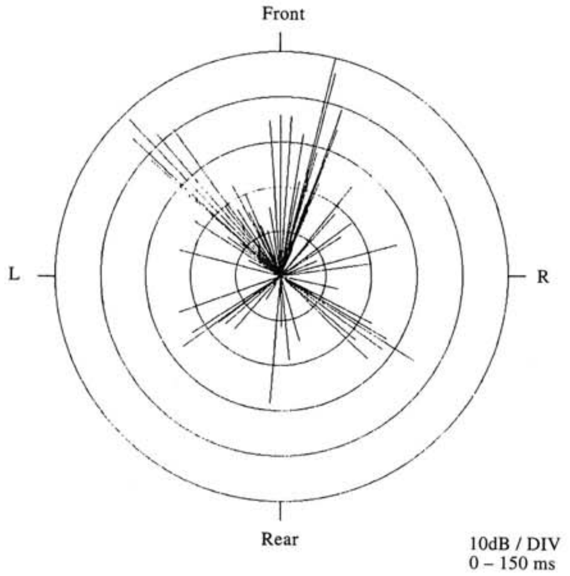


Figure 17. Directivity pattern in the "Defeat mode"

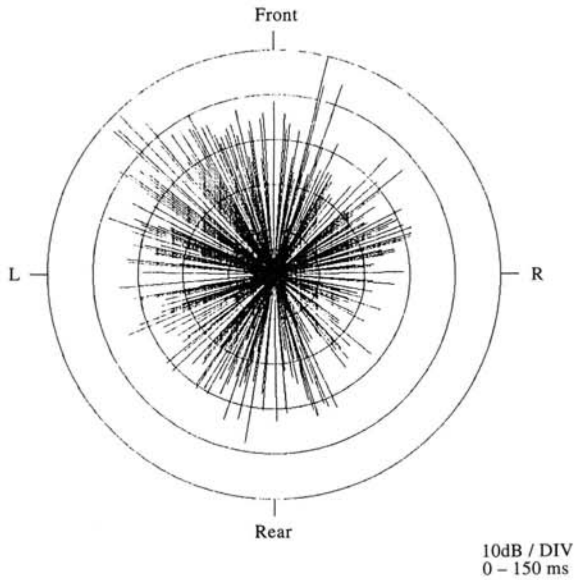


Figure 19. Directivity pattern in the "FUNKY mode"

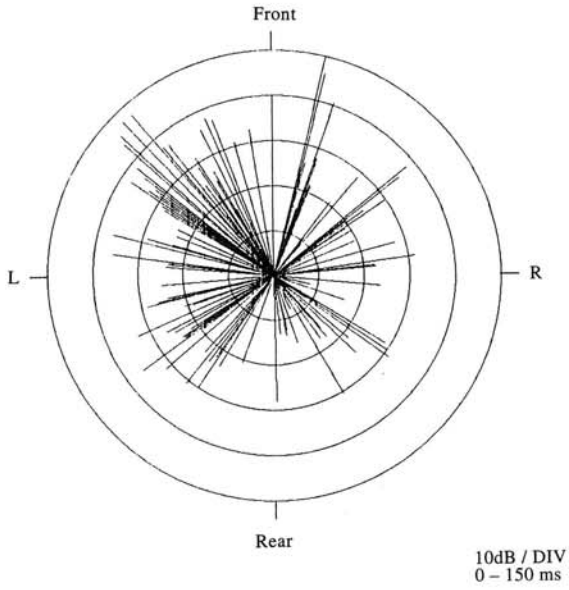


Figure 18. Directivity pattern in the "CASUAL mode"

4.2.2 Evaluation by listening

The optimum levels of effect sounds for the sound field control depend on music. With the optimum adjusted conditions, the original music is greatly enhanced, and sometimes sounds as if it extends beyond the car windows.

The Casual mode best suits music with a beautiful melody. It can reproduce music as natural as that of a modern concert hall with "presence" and "depth."

The Funky mode best suits music with a beat rhyrhm. It can reproduce music with a feeling of "presence". Reproducing a live-music recording will give the feeling of being present at water-front concert.

Listening to an audio signal without sound field control will give the listener the lack of "presence."

5. Conclusion

We are sure this car audio system has become very popular with young people since the release of the SERA.

We are convinced that car audio systems with sound field control using digital signal processing will, in future, be used throughout the car audio equipment world. The study of sound field control and car interior acoustics has just begun. We will further develop sound field control systems by studying the optimization of initial reflection sounds and the improvement of reverberation sound quality.

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Entered the company in 1979. He has been engaged in the research and development of car audio systems, and currently works in the 1st Engineering Department of the 1st Audio Products Division.

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Masaaki Hamai

Entered the company in 1977. He has been engaged in the development of car audio system, and currently works in the 1st Engineering Department of the 1st Audio Products Division.

Keijiro Katsumaru

Entered the company in 1968. He has been engaged in the development of car audio equipment, and is now in the 1st Engineering Department of the 1st Audio Products Division.