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Sound Processor (DASP Applications)

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Reflecting user needs, the car audio market has recently witnessed an increase in expensive products which have greater diversity and individuality than ever before. Ten years have passed since the component stereo system made its debut. During that period, the sound quality of car audio has been remarkably improved by expanding frequency range, smoothing frequency characteristics, and securing maximum sound pressure.
However, these improvements have not completely satisfied those users who desire an extreme power and sound quality that has, up till now, been beyond the limits of conventional car design.
Equipped with a newly developed DSP (FT8801) device, Fujitsu TEN's EQS-1000 sound processor produces the most powerful, most exciting stereo sound field that has ever been experienced in a car.
This report describes Fujitsu TEN's new sound processor, yet another innovation from the people who've made the best sound on wheels a reality.

1. Introduction

Each new model year sees the automobile interior becoming increasingly luxurious. Plush, sculptured, and ergometrically designed seats, doors, and panels and sophisticated electronic controls and entertainment systems are transforming the automobile interior into a posh rolling living room. In fact, many car owners have come to view their machines not as a mere means of transportation, but one of escape – a private home on wheels.

In tune with this trend, automobile audio equipment manufacturers have striven to produce systems with more convenient features, higher power output, more even frequency response, and wider dynamic range.

And yet, in spite of their efforts, auto audio is still far from ideal, mainly because the cramped passenger compartment is a very poor sound field. The dissatisfied listener desires a more natural and spacious sound field such as that of a concert hall or theater.

This need has prompted us to develop the EQS-1000 sound processor to give the listener an "instant escape from the cramped car interior" by producing a sense of space and presence as is experienced in a concert hall. Fujitsu TEN's sound processor uses the new FT8801 digital signal processor (DSP) developed by Fujitsu TEN. This paper is a summary of the development of the EQS-1000.

2. Overview

2.1 Sound fields

A sound field is composed of direct sounds and delayed reflections and echoes of the direct sound. The initial reflection and reverberation characterize the sound field and give a sense of depth and space. Consider now the acoustic characteristics of a concert hall, an example of a spacious sound field. Figure 1 is a cross sectional view of a concert hall. Three classes of sound constitute this sound field. First, there is the direct sound ① which arrives at the listener directly from the sound source. The listener locates the sound source by this sound. Next, we have initial reflections ② from behind the stage and

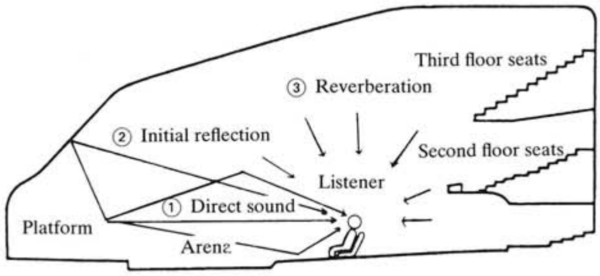

Figure 1. Cross section of a concert hall

walls. These reflections reach the listener with short time lag (50 ms or less) after the direct sound and give a sense of space. Lastly, a sense depth is provided by reverberation ③ resulting from complex and repetitive reflection and attenuation of the original direct sound as it bounces from the ceiling, floor, walls, and other objects. This sound reaches the listener from all directions.

Figure 2 shows the time-base composition of these three classes of arriving sound. The first

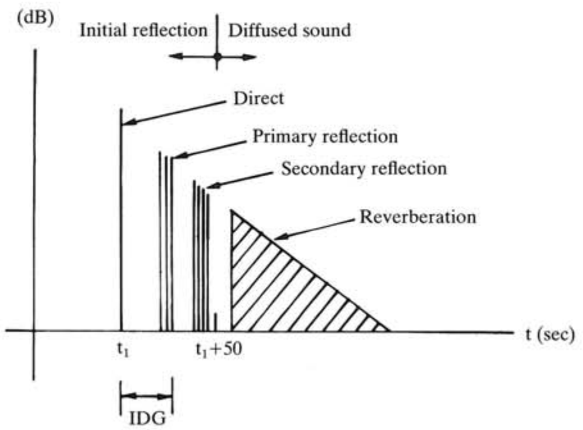

Figure 2. Structure of reflected sound

component is the time it takes for the initial sound to propagate from the source to the listener. This is followed 20 to 50 ms later by the initial reflection, and still later by reverberation. Figure 3 is an analysis of the directions of arrival of reflected sounds with the lapse of time. Frontal components are overwhelmingly dominant for the first 20-50 ms. Reflected sounds begin to reach the listener from all directions thereafter. Duplicating the acoustics of a concert hall in a listening room is not easy. Figure 4 shows the results of comparison of the acoustics of a large and small sound field. The time lag between the initial reflection and reverberation is evident. Given the same sound absorption coefficient, the reverberation time can be determined by *V/S* (*V*: room volume, *S*: room surface area) on the basis of Eyring's equation. That is, the acoustics of a larger sound field can be reproduced in a smaller sound field by electronically correcting the short initial reflection and reverberation times. This is just what our sound processor does.

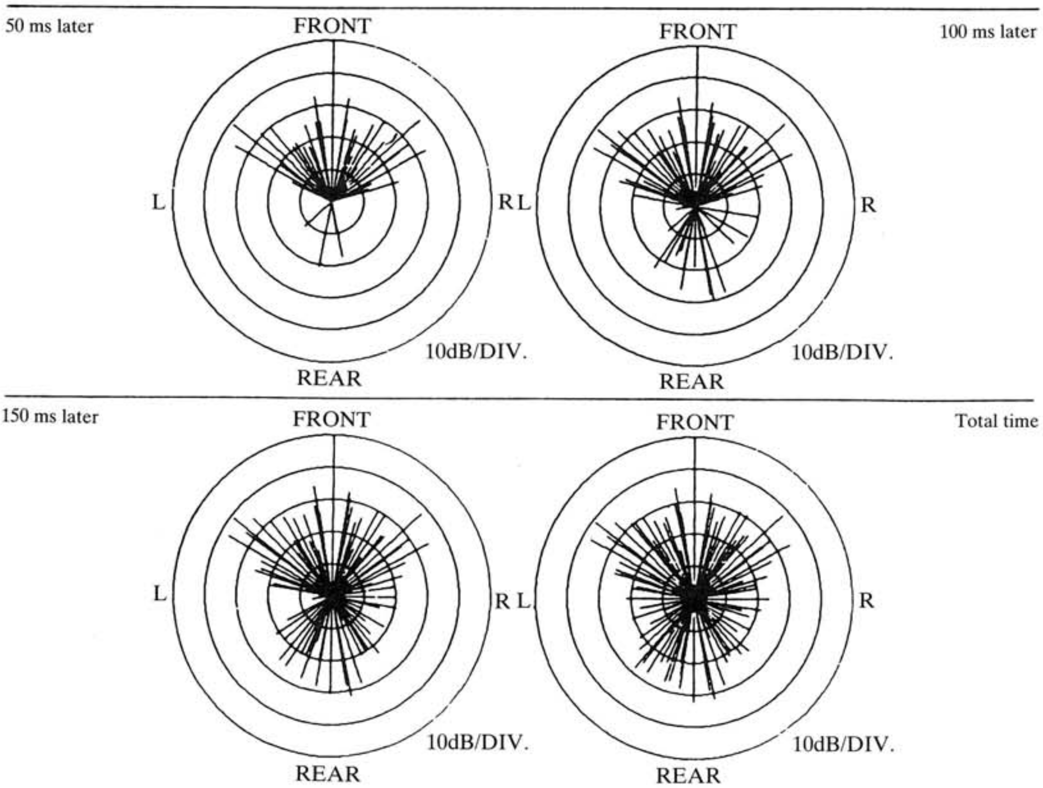

Figure 3. Directivity diagram of reflected sound

Parameter	Concert hall	Listening room
Room volume, V (m^3)	5,800	50
Room surface area, S (m^2)	2,000	90
V/S	2.9	0.56
Average free path, $4V/S$ (m)	11.6	2.24
Reverberation time (s)	1.3	0.25
Average sound absorption coefficient	0.30	0.30
Room constant (m^2)	857	39
Critical distance (m)	5.8	1.2
Sound source and listening point distance (m)	12	3
Initial reflection sound arrival time (ms)	50	9
Direct sound attenuation (dB)	−30	−18
Reverberation sound contribution (dB)	23	10

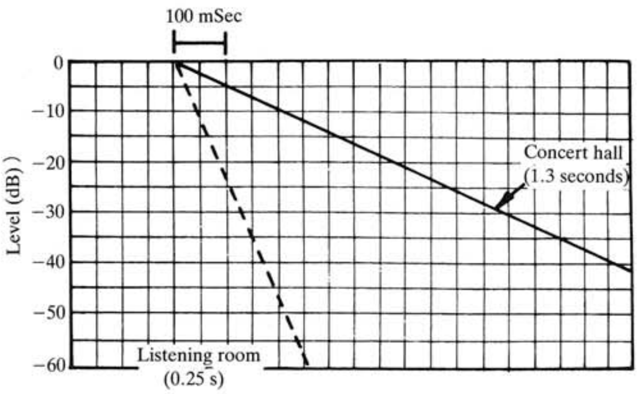

Figure 4. Reverberation characteristics in a large or small sound field

3. Sound processor system

The sound processor system supplies presence when reproducing music in a listening room or similar space. Among the many ways of achieving presence, addition of reverberation with a delay circuit is most commonly used.

3.1 Delay circuit

A block diagram of a typical delay circuit is shown Figure 5.

In operation, the input signal (direct sound) is delayed (τ) by a delay circuit. The resultant signal is the initial reflection. This signal is then attenuated and mixed with the direct sound to add reverberation. The attenuation factor (g) of the attenuator is a value less than 1. Small sound fields can be corrected to provide the acoustics of a larger sound field by controlling the delay time and attenuation factor.

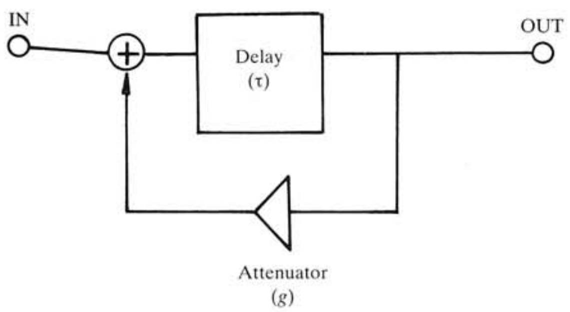

Figure 5. Delay circuit

3.2 Reproduction method

In a large sound field, reverberation reaches the listener from all directions. In a small sound field, however, in which electrically corrected reverberation and initial reflection sounds are reproduced by a limited set of speakers, the reverberation tends to exhibit directivity. The solution to this problem should allow for the following factors:

① Location of speakers
② Timbre of speaker units
③ Number of speakers
④ Speaker frequency response

First, speakers with identical or equivalent timbre and frequency response need be chosen for both the main speakers (for direct sounds) and auxiliary speakers (for reverberation sounds). Omnidirectional speakers are best for reverberation sound output. It is also important that the orientation of the reverberation speakers be symmetrical in terms of height and angle. Slight differences in height or angle could overemphasize the sound from either speaker, and produce an unnatural timbre. If the speaker used to reproduce initial reflections is directed at the listener, the sound source would be highlighted rather than diffused. The angle of the speakers must be adjusted to create natural presence.

3.3 Digital audio signal processing (DASP)

As stated earlier, a delay circuit is essential for generating reverberation. Numerous delay circuits and adders would be needed to achieve natural reverberation. Implementing these circuits with analog bucket-brigade delay devices and operational amplifiers would not only entail a huge circuit geometry but also would be liable to certain performance problems, such as lower S/N ratios and higher distortion. We used digital audio signal processing (DASP) introduced to minimize these problems.

DASP offers the following benefits:

① Precise and uniform characteristics
② Easy modification (software)
③ Single processor implementation of different hardware functions

Thus, DASP makes possible a high-performance, high-quality sound processor better suited to market needs than the analog circuit techniques previously used.

4. Passenger compartment sound field control

4.1 Acoustics condition in a car

One advantage the passenger compartment has over the listening room in the home is that the focal point of the sound system is more easily adjusted. Moreover, four speakers can be laid out to surround the listener. The compartment thus offers unique conditions for sound field control, but it is also subject to certain acoustics-related problems:

(1) Natural-mode resonant frequencies resulting from the volume and geometry of the compartment range from several dozens to several hundreds of Hertz and disturb the system frequency response.
(2) The listener is not equidistant from the left and right speakers and so does not experience the optimum stereophonic effect.
(3) The limited mounting space for speakers may limit speaker diameter, resulting in a lack of bass.
(4) The reverberation time is short, with little echo.
(5) Rigid and soft interior materials are mixed, making the arrival of reflected sounds unnatural.

Problems (1) through (3) can be solved by the use of a fix equalizer, a center speaker, and a subwoofer. Problems (4) and (5) can be solved by controlling the durations of the initial reflection and reverberation.

4.2 Sound field reproduction in a car

In the home, speaker orientation and placement is relatively flexible. This is not true in a car, where adequate and unobstructed spaces to serve as speaker baffles are limited. If a tray is available, it should provide a fairly generous amount of mounting space for the rear speakers. The front speakers are subject to more critical restraints, limiting the size of the speaker's effective piston area. A smaller speaker produces less bass. Therefore, a 3-D system with a subwoofer is used. This system consists of a large 20-cm diameter subwoofer and two smaller full-range units (10-20 cm dia.) mounted in the rear tray. This 3-D system supplements the frequency response of the main speakers (for direct sound) and subspeakers (for reverberation). The use the subwoofer for bass enhancement allows smaller diameter speakers to be used, broadening the choice mounting locations. Figure 6 shows the speaker layout.

The main speakers for direct sound reproduction of the left and right channels are mounted on the door panels. The direct sounds from the left and right speakers, the initial reflection sound, and reverberation are summed and reproduced by the center speaker. The center speaker is not directed at the listener but is tilted towards the windshield which acts as a deflector. The rear speakers for reverberation reproduction are mounted on both sides of the rear tray. The subwoofer is mounted in the center of the rear fray.

With this system, a sound stage forms in front of the listener and at eye level. Direct sound reaches the listener first. Then, initial reflection and reverberation arrive from the front with slight time delays, creating presence.

4.3 Sound processor (EQS-1000)

Figure 7 shows the components that make up the surround processor system. The surround processor has presets which simulate four room sizes: concert hall, large club, church, and stadium. These presets are key-selectable. Separate controls adjust the initial reflection and reverberation times and the speaker levels of the center speaker (initial reflection

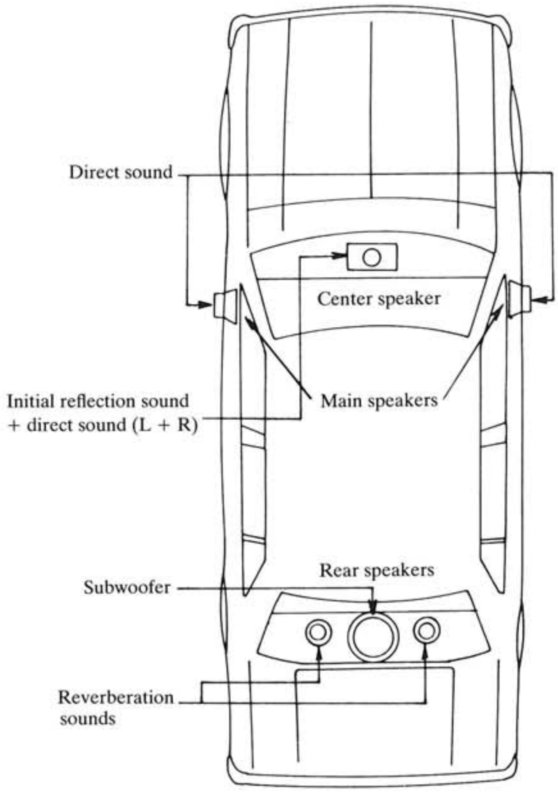

Figure 6. Speaker layout

sound), rear speaker (reverberation sound), and woofer for greater presence. A large color LCD provides a positive visual indication of the effects of these controls. Figure 8 shows the EQS-1000 and principal controls.

4.4 Control microcomputer

The microcomputer used in the surround processor functions as a user interface. The microcomputer transmits room size preset information (filtering coefficient and delay time) to the DASP when the preset keys are operated.

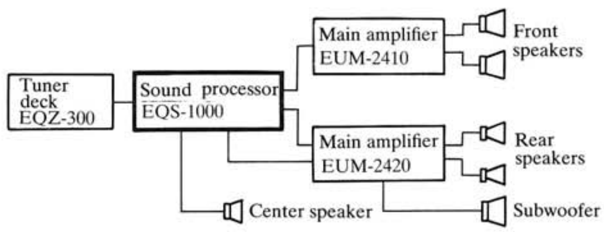

Figure 7. The DASP system

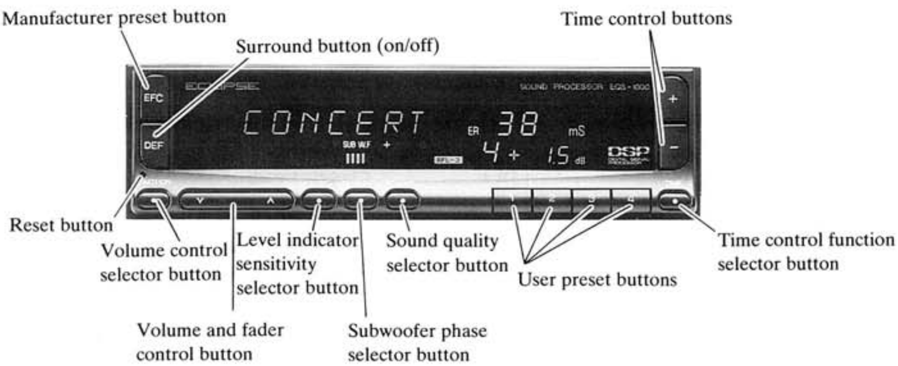

Figure 8. Display and main function of EQS-1000

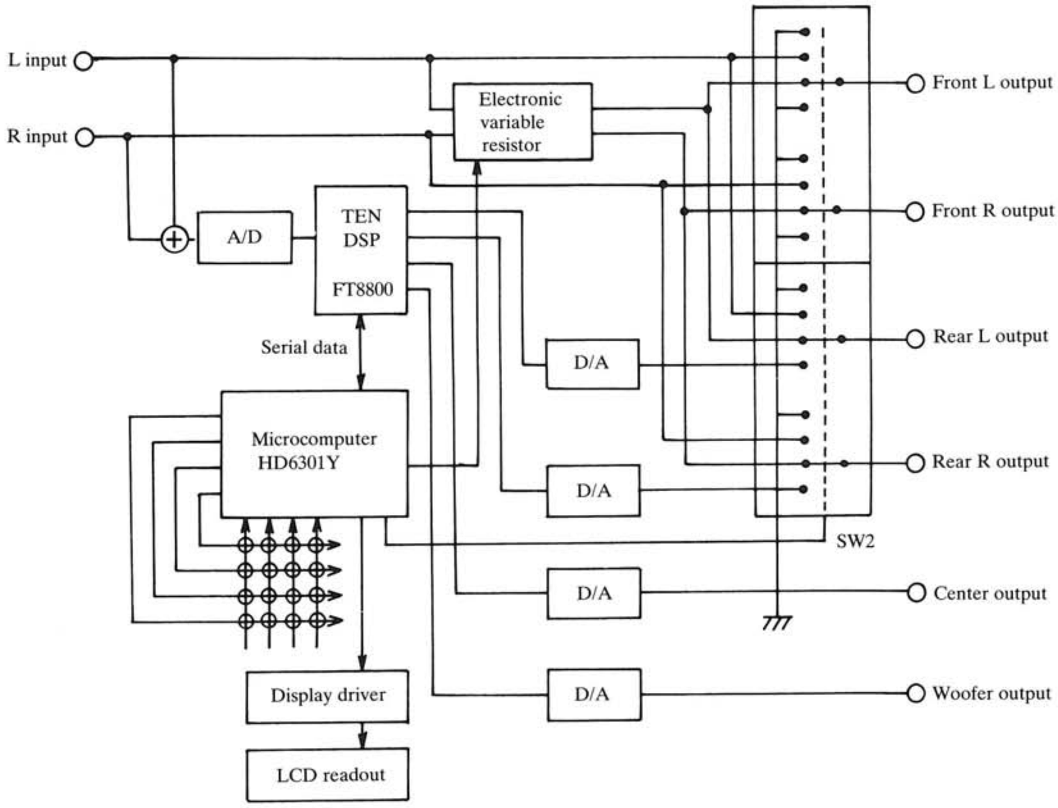

Figure 9. Sound processor

Table 1. Controls and displays

Functions			Remarks
Major category	Subcategory	Key name	
Control functions	Time controls Initial delay/reverberation	+ −	Single-step when pressed shorter than a specified time interval; Speeds up when pressed for a longer time interval.
	Level controls ER/LR/woofer	UP DOWN	Single-step when pressed shorter than a specified time interval; Speeds up when pressed for a longer time interval.
	User memory (read/write)	1−4	Program in memory when pressed longer than a predetermined time interval; initially set to the manufacturer's presets.
	Woofer phase Normal/reverse	PH. SHIFT	Toggle switch
	Input signal bandwidth limiter RFL1−3	RF. LIMIT	① RFL2 → RFL1 → RFL2 → RFL3 → ①
Selection functions	Sound field effect (on/off)	DEF	Toggle switch
	Manufacturer's presets (read only)	EFC	① Concert hall → Large club → Church → Stadium → ①
	Initial delay/reverberation time Mode selector	TIME	Toggle switch The selected mode is adjustable with the + and the − key.
	ER/LR/woofer Woofer/fader Mode selector	FUNCTION	Cyclic selection (sound field effect on) Toggle switch (sound field effect on) The selected mode is adjustable with the UP and DOWN keys.
	Level indicator sensitivity level selector	LEVEL	① HIGH → LOW → ① : toggle selection
	Woofer cutoff frequency	FC	WOFfc = HIGH/LOW
	Woofer defeat	WOF	WOF = On/Off
Readouts			

Figure 9 is the internal block diagram of the sound processor.

The microcomputer provides key entry, LCD output, DASP control, electronic volume control, analog switch control, preamplifier and power amplifier power on/off, and muting control.

Table 1 describes the controls and the readouts of the surround processor.

Among the principal keys are the EFC key to select the sound field, the + and − keys to adjust the initial delay and the reverberation time, and 1 to 4 keys to store the adjusted sound fields in memory.

The microcomputer uses a low-speed serial interface to modify constants.

Figure 10 shows the low-speed serial interface connects the microcomputer and the DASP.

Figure 11 shows the serial data structure.
The command word directs the microcomputer to read from or write to the DASP.

The address word specifies the address of the RAM in the DASP.

Figure 12 shows a communication timing chart.

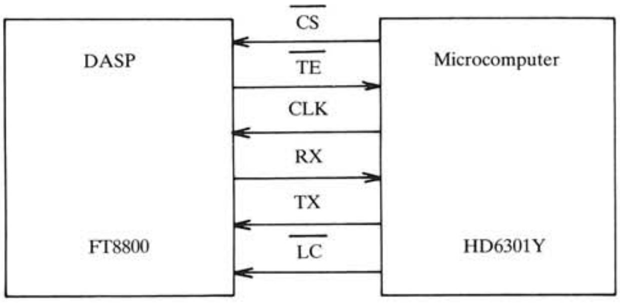

Figure 10. Low speed serial interface

Command: 7 bits	Address: 9 bits	Data: 24 bits

Commands select reading from the DASP (0001000B) and writing to it (0010000B).

Figure 11. Serial data structure

In communication, commands and addresses are latched before data I/O takes place.

Figure 13 shows the main flowchart of the microcomputer.
The microcomputer scans keys periodically and, after key processing, displays the control status on the LCD.

ACC processing controls the preamplifier, A/D and D/A converters and the DASP power supply. CS processing controls the power amplifier and DASP reset.

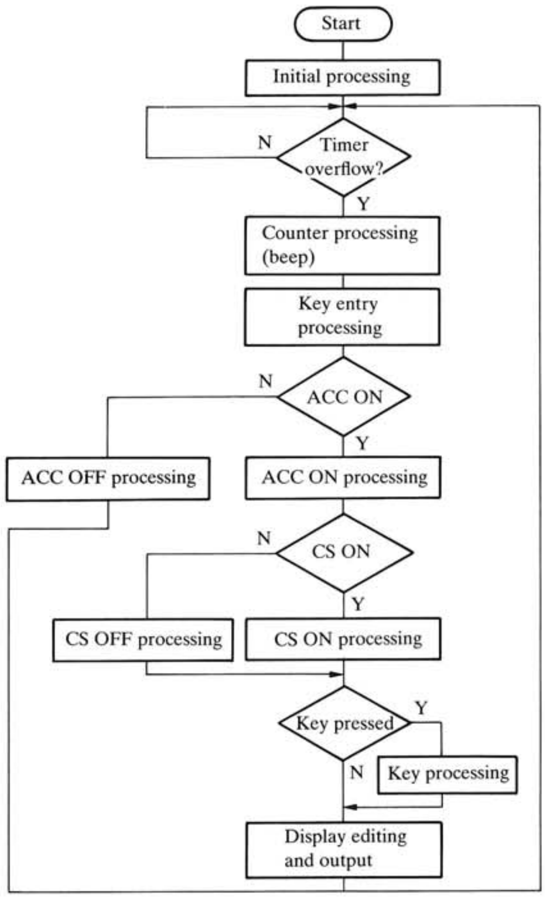

Figure 13. Main flowchart

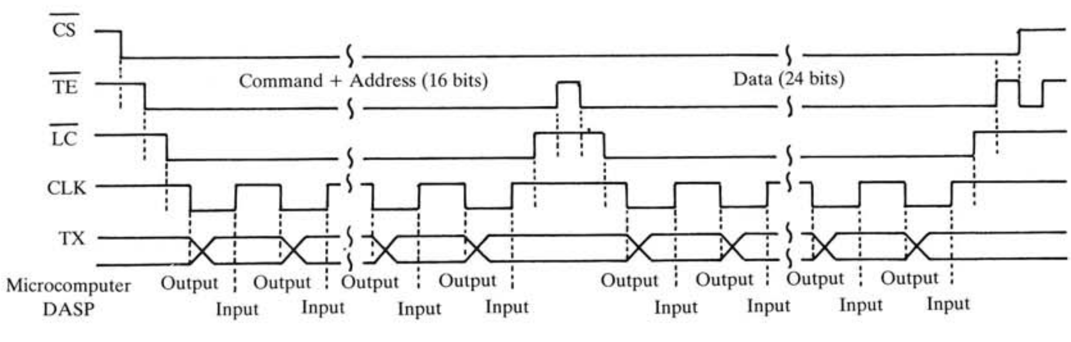

Figure 12. DASP write timing

Figure 14 is a flowchart of EFC key processing.

EFC key processing starts DASP processing by turning the soft mute on.

The mode counter is the incremented. If the counter equals 4, it is reset to 0 for the concert hall sound field. Likewise, a large club, a church, and a stadium sound field are represented by counter values of 1, 2, and 3.

The correct coefficients are set for each sound field selected and the soft mute is thus turned off.

The DASP then gives a beep which is turned off after a predetermined time interval.

Since the soft mute process, beep, and sound field coefficients are preset in a ROM table, the microcomputer proceeds with processing by incrementing the ROM address.

5. Specifications

Table 2 summarizes the sound processor specifications.

6. Conclusion

The preceding pages have outlined a sound processor capable of simulating the acoustics of four types of sound fields within an automobile passenger compartment. We will continue to investigate new applications for our state-of-the-art DASP and analyze market appraisal information on the system just released to further refine its performance. We hope that our DASP will find popular acceptance in systems which transform the car interior into a "mobile listening room" having equivalent or superior sound quality as that of the home listening room.

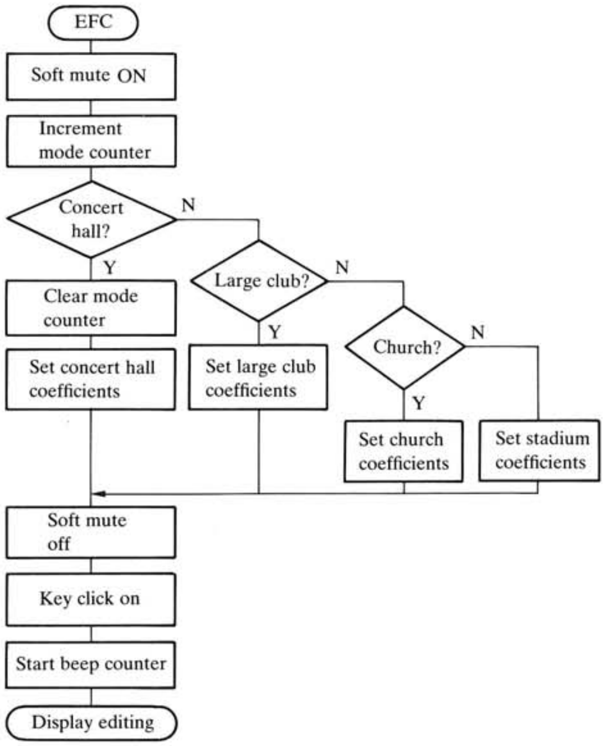

Figure 14. Flowchart for EFC key processing

Table 2. EQL-1000 specifications

Parameter	Specification		Remarks
Frequency response	20 Hz	-1 ± 2 dB	300 mV, 1 kHz reference, with surround components up to 8 kHz
	20 kHz	0 ± 2 dB	
Distortion	0.08% or less		300 mV, 1 kHz reference
Residual noise level	50 μV or less		Input shorted
Initial reflection time	Variable range: 0–100 ms		Concert hall environment
Reverberation time	Variable range: 0–480 ms		Concert hall environment

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