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DDL Car Audio System

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In October 1989, Fujitsu Ten produced the first in-car digital sound processor (DSP), the EQS-1000. The DSP creates an impression of "presence," adding a new dimension to car audio not possible without digital techniques. Not all DSP applications have, however, been a success; software and hardware limitations can result in unnatural sound, and sound processing is highly complex and difficult to optimize.
In February 1992, Fujitsu Ten released the DDL series, with the EQS-2000 digital sound processor at its core. This series reaches new levels of easy operation and accurate, natural sound field reproduction. The DDL series features many new technologies and functions, including transmitting digital sound via optical fiber.

1. Introduction

During the ten or so years since the audio compact disc made its debut, digital audio technology such as digital audio tape (DAT), digital radio broadcasts by satellite, and sound field control digital sound processor (DSP) are now commonplace. The advent of the mini disc (MD) and digital compact cassette (DCC) should increase the scope of digital implementation, making a fully digital car audio system a possibility in the near future.

As part of a program to develop digital audio equipment, we at Fujitsu Ten produced the world's first in-car CD player, jointly with Toyota. We also developed DSP sound processors which delighted car audio customers.

Fujitsu Ten use a philosophy of maximum digital implementation to develop the DDL (Direct-Digital Link) car audio systems in February 1992 for luxury cars. The DDL system features include auto-equalization for true hi-fi sound reproduction, and auto-tuning which automatically optimizes the effects of initially reflected sound and reverberations.

This paper briefly reports the development aims of the DDL series of car audio systems, their features, and the principal technical concepts behind them.

2. Development aims

The development aims for the DDL series were as follows:

2.1 Emphasis on sound quality

Improve sound quality on the system level by improving the sound quality of the individual components that make up each system. Also, minimize the adverse effects on overall sound quality of mixing components.

Improve the acoustics of the car interior.

2.2 System diversity

Make it possible to upgrade each system step-by-step and to connect conventional analog equipment to meet a variety of user needs.

2.3 Ease of operation

Make control panels easy to use so that sophisticated functions are controlled by simple operations.

The system we designed to encompass all of these development aims is described below.

3. System overview

3.1 System configuration

Figure 1 shows the configuration of the DDL digital car audio system.

The hideaway type DSP sound processor (EQS-2000) provides conventional sound field control, plus parametric equalization. A DSP expansion cartridge (EQU-8040) is available for the processor to add extra functions including increasing the number of bands in the parametric equalizer and changing the equalizer to a graphic equalizer. The commander (EQR-2140) controls the CD auto-changer, tuner, and audio unit, as well as all operations of the DSP sound processor.

The double thin lines in Fig. 1 are F-bus communication lines, and the double dotted lines are fiber optic transmission lines.

3.2 System features

3.2.1 Digital optical signal transmission

The DDL series has a new method of digital transmission using a fiber optic link (optical digital link) which improves sound quality.

If a CD and a DSP are combined using the conventional method of analog signal transmission through an RCA cable, the digital signal from the CD source must be converted to an analog signal before it is transmitted. The transmitted signal must then be converted back to a digital signal in the DSP unit for DSP processing.

Using an optical link allows the digital signal from the CD source to be directly transmitted to the DSP unit without incurring a conversion loss.

Digital optical signal transmission also makes it possible to reproduce sound with a very high degree of linearity, completely free from the effects of electrical noise in the car.

Table 1 compares analog and digital methods of signal transmission.

3.2.2 Improved acoustics

In addition to sound field control, the DSP sound processor provides parametric equalization to correct for peaks and dips in the car interior frequency response.

We have also changed the reverberation density under the sound field control to produce a more natural sound. Rear sound mix corrects for the unusual acoustic environment around the rear seat passengers.

3.2.3 Automatic adjustment of control effect

It is important to adjust sound field control and parametric equalization to produce the best possible sound.

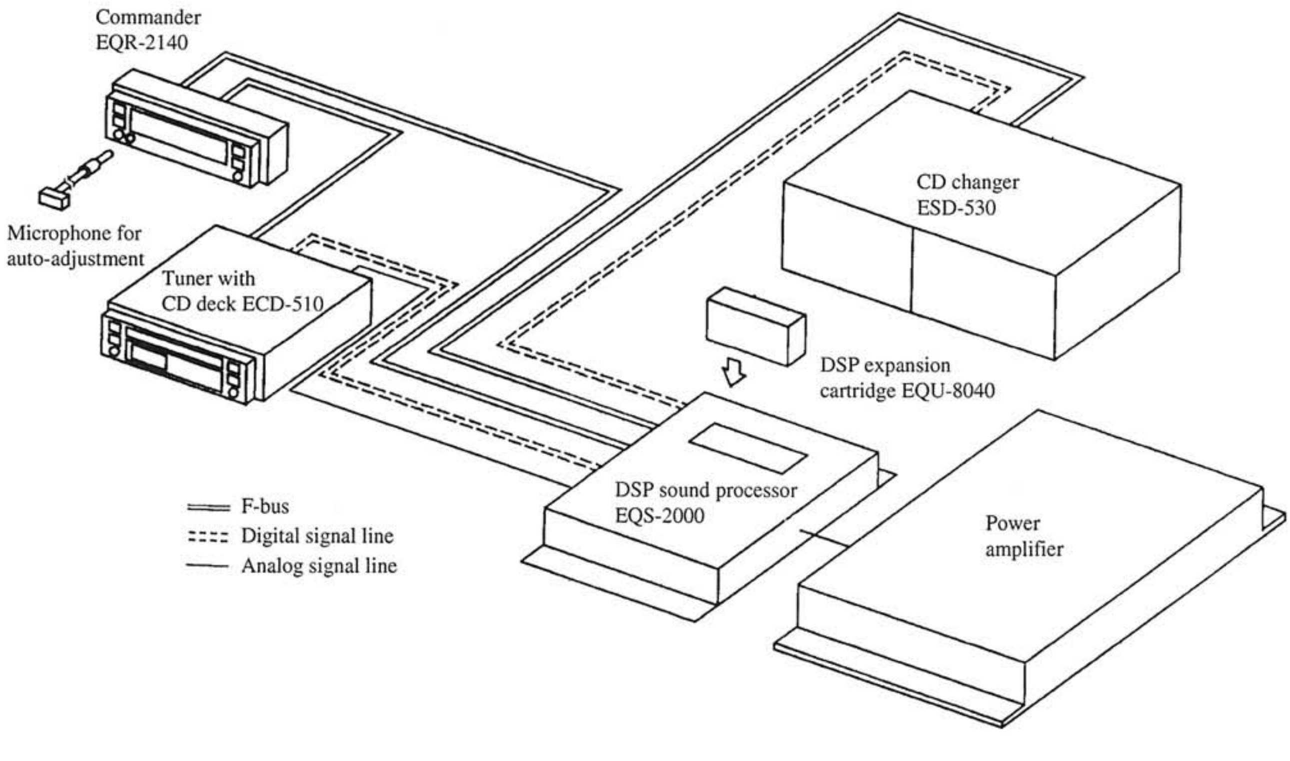
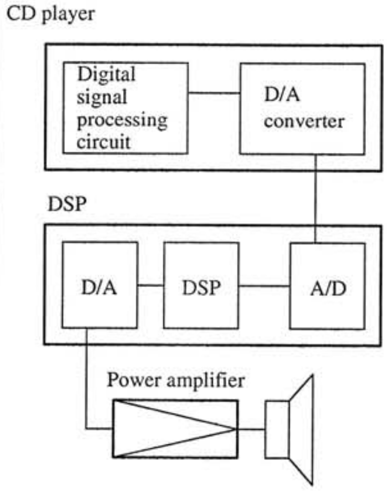
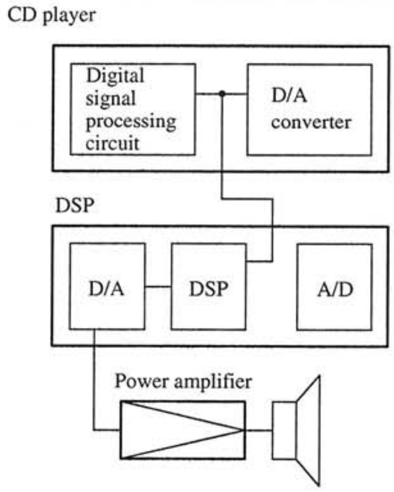


Figure 1. DDL series system components

Table 1. Comparison of analog and digital signal transmission

		Conventional method (Analog signal transmission)	New method (Fiber optic digital signal transmission)
Parameters	Overall harmonic distortion	0.08%	0.006%
	S/N ratio	90 dB	108 dB
	Frequency response	20 Hz to 20 kHz ±1 dB	20 Hz to 20 kHz ±1 dB
	Noise immunity	Not good	Very good
System configuration			

Notes: 1. These values assume that a CD is being played back.
2. Based on comparisons of two systems made by FUJITSU TEN

With the DDL series, you can adjust sound field control and parametric equalization with a single keytouch. Sound in the car is fed back via the microphone built in the commander, or an external microphone, and the system automatically adjusts its sound field control and equalization.

All of these features are discussed in more detail later on.

4. Digital signal transmission technology

4.1 Signal transmission system

The digital signal transmission system used by the DDL series conforms to the CP340 digital audio interface format standards type 2 specified by the Electronic Industries Association of Japan (EIAJ). The same standard covers CD players, DAT, and BS tuners. The audio interface format defines subframes, formats, and blocks as shown in Figures 2 and 3.

Signals are transmitted in a bi-phase mark signal system, in which the source signal is modulated as shown in Figure 4. The clock frequency is 2.8224 MHz for CD and 3.072 MHz or about 3 megabits per second (in NRZ code, 6 megabits per second) for DAT. This assumes that audio signals are sampled at 48 kHz giving 48 KHz × 32 bits × 2 ch = 3.072 MHz.

In the DDL car audio systems, these high-speed digital signals are converted to "0" or "1" on off signals of LEDs in the optical fiber link. An optical fiber link is ideal for method signal transmission in a car because it is largely immune to external noise.

Figure 5 shows the structure of the optical fiber cable. The cable has a core 0.97 mm in diameter, with a high refractive index, covered with a cladding material, 2.0 mm in diameter, with a low refractive index. The cladding is covered by a sheath for protection. Figure 6 illustrates how light propagates through the optical fiber by total internal reflection. Signal transmission using optical fiber is, however, more prone to timebase jitter than transmission through a coaxial cable.

4.2 Source of jitter and remedies

There are two major causes for jitter in optical fiber transmission.

1) Jitter due to misalignment of the connector axes

The optical fiber connectors that are generally used in cars and in household applications are locked together by a boss on one connector (Fig. 7). This structure is not, however, always secure. As the cable vibrates, the

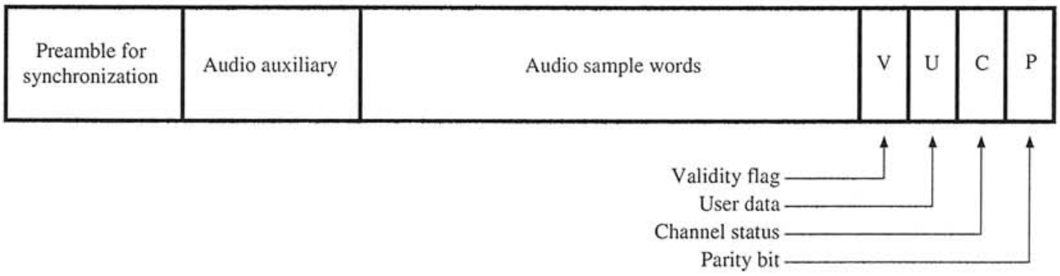


Figure 2. Frame data format

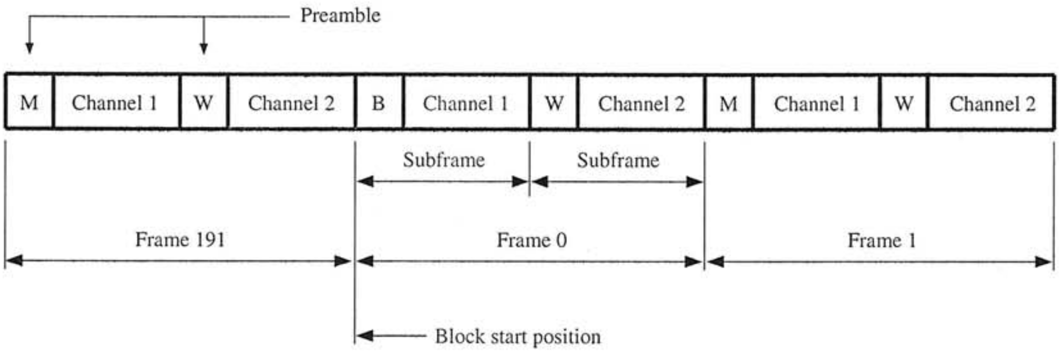


Figure 3. Data format of subframe

connector can move and cause misalignment of the optical axes of the LED and cable core (Fig. 8), producing modulation jitters.

2) Jitter due to optical fiber bending
Ideally optical fibers should be straight, but some bending is unavoidable in a car. Light propagation in the optical fiber depends on total internal reflection and this only happens if light hits the boundary of the core and cladding material at less than the critical angle. Bending the cable may allow some light to escape from the core to the cladding layer, reducing the quantity of light received. If you now add cable vibration to this, the bending radius will oscillate, causing the quantity of light received to also oscillate and cause modulation jitter.
Any jitter will appear in the clock pulses for D/A converters, upsetting D/A conversion timing and seriously impairing sound quality.

3) Preventing Jitter
In the DDL series we have taken steps to prevent jitter by using cable connectors in the high-speed optical links and by adding vibration resistance to optical fiber connectors.
The optical connectors in the DDL series are of the lever-lock type. This connector type ensures that the engagement between a reception module and an opti-

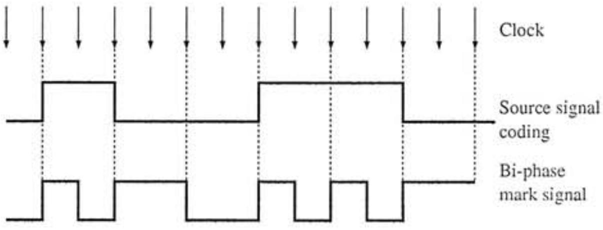


Figure 4. Bi-phase mark signal system

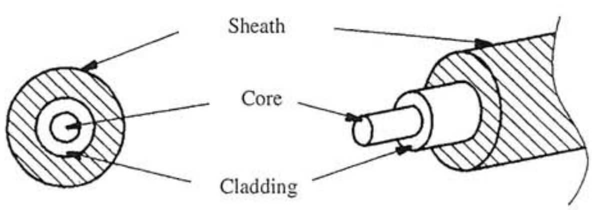


Figure 5. Structure of optical fiber cable

cal connector is secure. The lever-lock structure prevents jitter caused by misalignment of connector optical axes, and helps prevent loosening of the optical connector due to car vibration. A further step was to use an EIAJ connector for the high-speed optical link on the DSP processor's sound module.

These new connectors that reduce jitter are compatible with the friction lock type optical connectors widely used in home audio. They are also compatible with portable equipment that has EIAj optical digital output, such as the new MD and DCC, as well as the established CD and DAT.

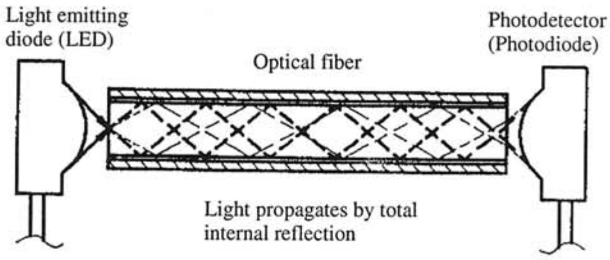


Figure 6. Light signal transmission inside optical fiber

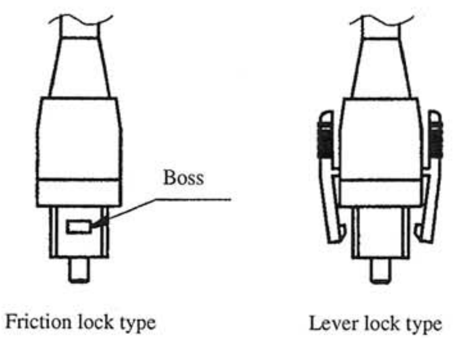


Figure 7. Connector structure

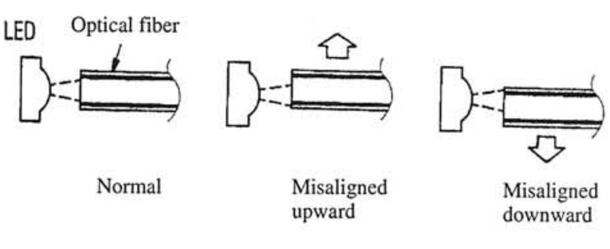


Figure 8. Misalignment of optical axes in a connector

5. Improving car acoustics

5.1 Problems with car acoustics

A car interior has the following problems with its acoustics:

① Optimal stereo effect is not possible because listeners are not centered between the left and right loudspeakers.

② Short reverberation time, and little reverberation

③ The mixture of soft and hard materials in the car interior makes reverberation unnatural.

④ Constraints on mounting positions make it difficult to install large loudspeakers, so the sound lacks low frequencies.

⑤ The natural frequencies, dependent on the volume and geometry of the car space, range from several tens to several hundreds of Hertz, with peaks and dips in the frequency response.

Problems ① to ③ have been solved by adding a center loudspeaker to reduce asymmetry, as is already done with DSP sound processors, and by sound field control to add initial reflection and reverberation. Problem ④ has been solved by a 3D system with subwoofers.

To solve problem ⑤, however, we must correct for the acoustic transmission frequency response in the car compartment where the system is installed. Because we expect our commercial models to be used in a wide variety of cars and with a wide variety of loudspeaker systems, we developed a DSP parametric equalizer, installed in the EQS-2000, which corrects for variations in frequency response.

5.2 Parametric equalizer

5.2.1 Overview

The acoustic transmission frequency response in a car has extremely complex peaks and dips.

Conventionally, a graphic equalizer has been used to correct for the response, but the effect is limited since the center frequencies (F_0) and rolloffs (Q) are fixed.

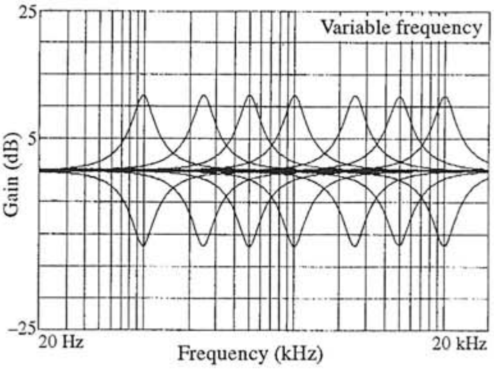
Unlike a graphic equalizer, the parametric equalizer can set any center frequency, rolloff, and gain. This is shown in Figs. 9 (a), (b), and (c).

5.2.2 Designing the parametric equalizer

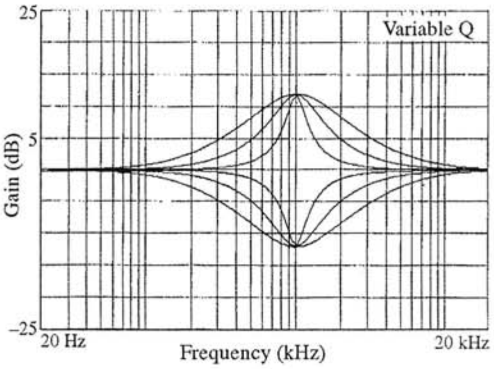
The DSP does the processing needed for the parametric equalizer using second-order IIR filters with cascade connections to get the needed EQ bands (Fig. 10).

Each coefficient of the IIR filter can be calculated from the required values for center frequency, rolloff, and gain by solving the equations below.

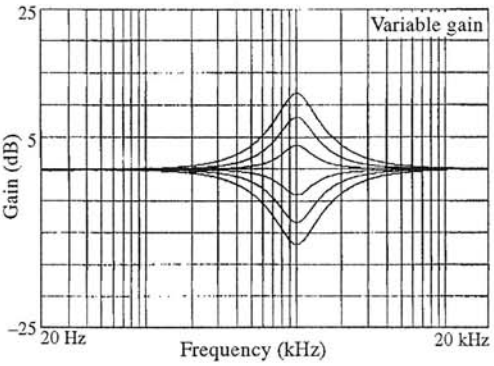
Coefficients calculated using these equations are transferred from the microprocessor to the DSP to give the required equalizer characteristic.



(a) Variable center frequency F_0



(b) Variable rolloff Q



(c) Variable gain G

Figure 9. Parametric equalizer characteristics

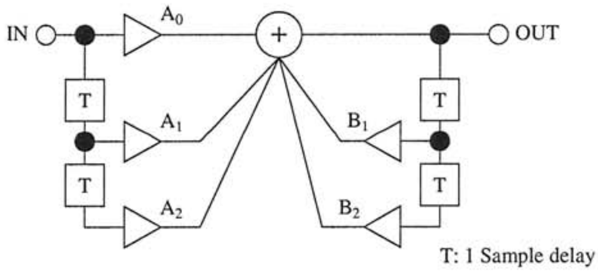


Figure 10. Second-order IIR filter

$$G_p = \sqrt{G}, \quad G_m = 1/G_p$$
$$\omega_o = \tan(\pi \times F_o/F_s)$$
$$P = 1 + G_m \times \omega_o/Q + \omega_o^2$$
$$A_0 = 1 + (G_p - G_m) \times \omega_o/(P \times Q)$$
$$A_1 = (-2 + 2 \times \omega_o^2)/P$$
$$A_2 = 1 - (G_p + G_m) \times \omega_o/(P \times Q)$$
$$B_1 = (-2 + 2 \times \omega_o^2)/P$$
$$B_2 = -(1 - 2 \times G_m \times \omega_o)/(P \times Q)$$

5.3 Sound field control

The sound field control system was introduced in previous issues of this journal, and we will now focus on its features.

5.3.1 Improved reverberation sound quality

Reverberation sounds are generated by connecting comb filters in parallel as shown in Fig. 11.

Here, the delay (τ_n) and feedback coefficient (G_n) of each comb filter are optimized to compensate for loss of reverberation density and coloration.

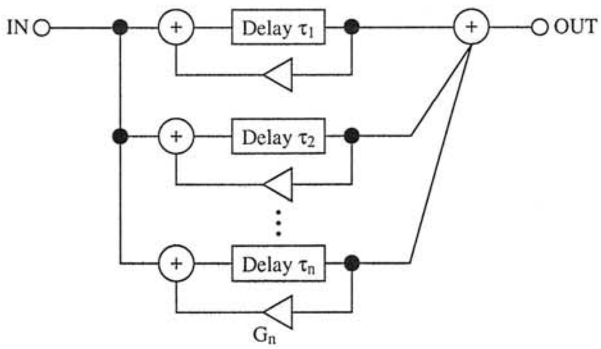


Figure 11. Comb filter block diagram

Figure 12 shows simulations of impulse responses of the comb filter block. Reverberation density is higher than in the conventional method, which means a more natural sound.

5.3.2 Addition of "rear mix"

Previous in-car sound field reproduction fed direct, unprocessed sound to the front and center loudspeakers, initial reflection mainly to the center loudspeakers, and reverberations to the rear loudspeakers. Reverberation tended to dominate sound in the rear seat, so the rear seat passengers hear unclear and unnatural sound.

The DDL series solves this problem by feeding direct sound to the rear loudspeakers too. Based on our listening tests, we set a direct sound level of about 10 dB below the reverberation level. This level is adjustable between -70 dB and +10 dB relative to its standard value.

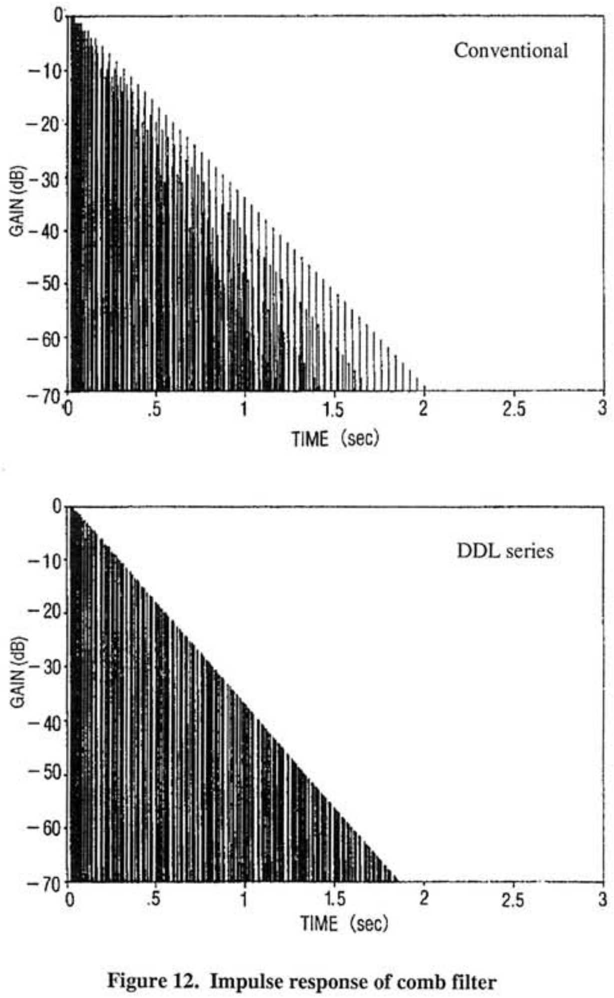


Figure 12. Impulse response of comb filter

6. Automatic equalization

6.1 Overview

The automatic equalization system automates the following processes that were done manually before:

① Optimizing the levels of initial reflection and reverberation under the sound field control.

② Correcting for the acoustic transmission frequency response in the car.

The automatic equalization system can be used as a tool which automatically corrects for the acoustics of the car interior. It can also be used to customize the acoustics to personal taste with some manual tuning.

Figure 13 is a block diagram of the automatic equalization system. The DSP generates a reference signal to test the acoustic characteristics in the car. This reference signal passes through the D/A converter, the amplifier, and then to the loudspeakers. This is the same path as that taken by normal audio signals.

The signal from the microphone in the command unit is converted by an A/D converter, then passed to the DSP which uses it to adjust equalization.

6.1.1 Generating a reference signal

The DSP generates white noise. A number of pink noise filters, implemented with fourth-order IIR filters, convert the white noise to pink noise with a known amount of energy in each octave band. This is the reference signal.

6.1.2 Setting the reference input level

In automatic equalization, it is possible for the desired S/N ratio to be too low because the reference signal level is too low. This can be caused by variations in the performance of the external test equipment used or signal saturation (data overflow) in the measurement loop.

To avoid this problem, a reference signal level is preset in the system's microprocessor sound level control.

The reference signal level is gradually increased until the microphone input reaches a predetermined level. A flowchart of the process is shown in Figure 14.

6.1.3 Averaging

Although the measured data read from the DSP has been averaged over a very short time by a low-pass filter, it is averaged again because it is instantaneous data.

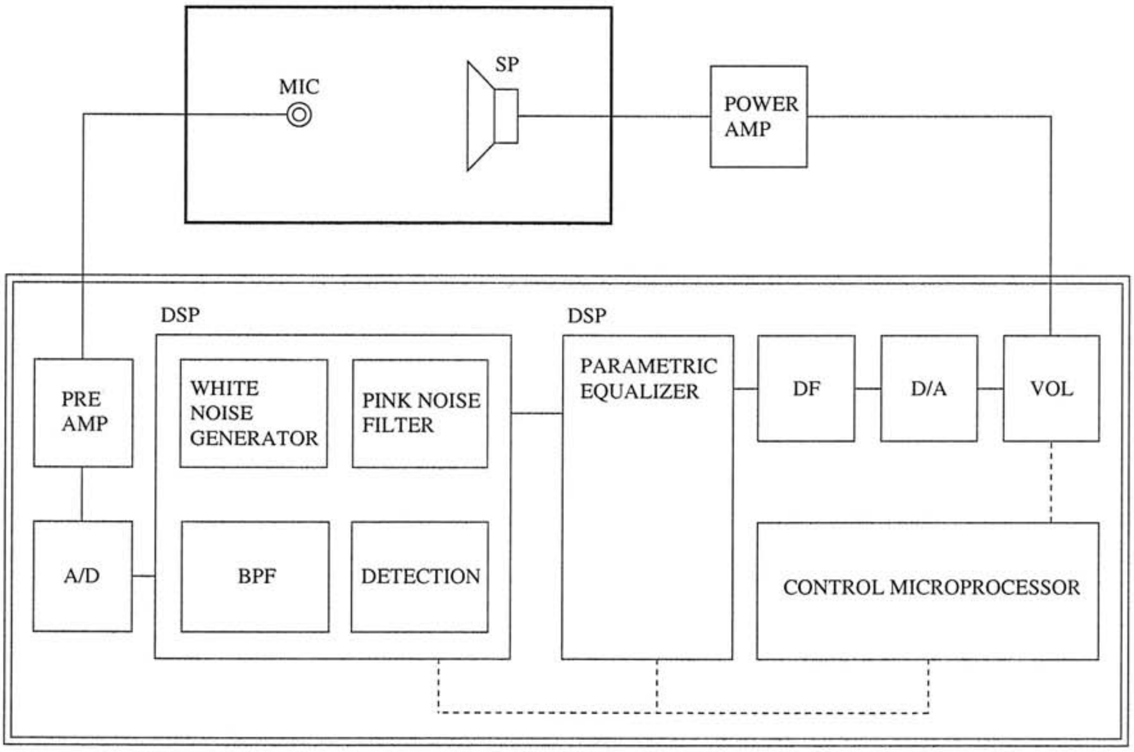


Figure 13. Block diagram of automatic equalization

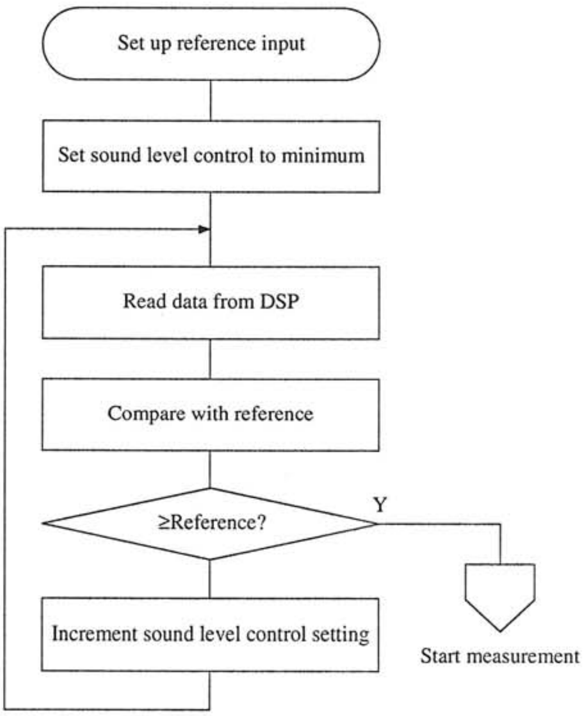


Figure 14. Flowchart of setting reference input level

This system takes sample measurements 16 times at 50-ms intervals, and the average of the 16 sampled values becomes the measured data. Figure 15 is a flowchart of the averaging process.

6.2 Automatic balance correction

The balance between loudspeakers has a significant effect on sound field reproduction. Balance correction automatically optimizes the balance between the initial reflection and reverberation.

The correction procedure is described below.

① Switches the system to automatic correction mode.
② Outputs a reference signal to the front loudspeakers.
③ Sets the reference signal level as described above and stores that level.
④ Outputs a reference signal from the rear loudspeakers.
⑤ Sets the reference signal level as described above and stores that level.
⑥ Calculates the optimal reverberation sound level from the difference between the front and rear loudspeaker levels.

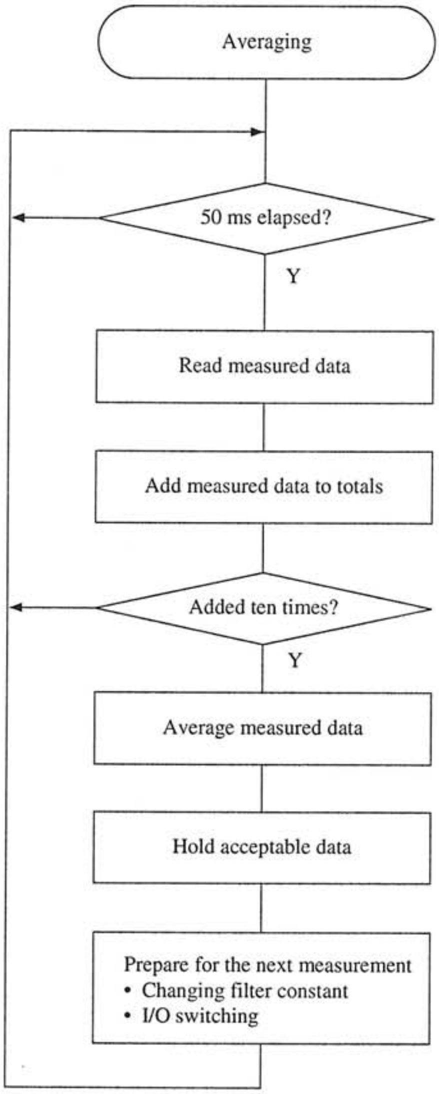


Figure 15. Averaging the microphone input level

⑦ Outputs a reference signal from the center loudspeakers.
⑧ Sets the reference signal level as described above and stores that level.
⑨ Calculates the optimal initial reflection level from the difference between the front and center loudspeaker output levels.
⑩ Outputs reference signals from all loudspeakers before completing the correction process.

6.3 Automatic correction of frequency response

A parametric equalizer is used to automatically correct for variations in the acoustic transmission frequency response in the car. The result is a close-to-ideal frequency response.

The measurement band between 63 Hz and 20 kHz is divided every one-third octave to make 25 bands.

The correction process is described below.

① Switch the system to automatic correction mode.
② Output a reference signal from the front loudspeakers.
③ Set the reference signal level as explained above.
④ Calculate the target reference signal level for the frequency and store that level.
⑤ Divide up the measurement band and measure the sound level at each frequency.
⑥ Search for peaks and dips in the frequency response measured.
⑦ Calculate the correction required for each peak or dip according to a prescribed priority order.
⑧ Fine adjust at each correction frequency according to fuzzy rules.
⑨ Calculate the correction necessary for gain and rolloff according to fuzzy rules.
⑩ Write the correction values into the parametric equalizer.
⑪ Repeat steps ④ to ⑩ for the rear loudspeaker channel.
⑫ Outputs corrected reference signals from the front and rear loudspeakers before completing the correction process.

7. Conclusions

The DDL car audio systems has realized new levels of functionality and sound quality, and met with an excellent reception in the car audio world. We are committed to continuing our product development using suggestions from our customers and building on our knowledge and experience.

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