The Development of a High-End Car Audio Product, the *Sound Monitor* Digital Preamplifier

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GOOD DESIGN AWARD

Abstract

In recent years, high-end car audio products that are as competitive in sound quality and design as home audio products have been put on the specialty retail market. However, actually installing these products in vehicles may result in significant degradation in sound quality, depending on the vehicles' conditions, although separate evaluations of them in audition rooms have proven that they provide high-quality sound. Specialty stores have improved their installation techniques year by year, making great progress in the past several years. Nevertheless, the layout limitations of onboard speakers have not allowed ideal, "realistic sound" to be achieved. Therefore, for the Sound Monitor 2000 model, we have set a development goal of reproducing "realistic concert sound" for drivers (listeners) by solving problems related to acoustic characteristics in passenger cabins with high-performance DSPs and the newest digital control technology.

This document introduces the current state of sound-producing technology and its problems in the specialty retail market and focuses on the Sound Monitor 2000 model "Digital Preamplifier (DTA-500X)", which has been developed to solve these problems, describing an overview of the product and the main points of its design.

1. Introduction

Our company developed a high-end car audio sound monitor for specialty stores, and sent the first model to the market in 1993. The product concept involved the pursuit of high performance and high sound quality in a car audio component unit by adjusting vehicle-compartment acoustic characteristics of audio systems actually installed in vehicles in order to "create the world's best sound quality inside a vehicle." Originally, the specialty store market was only at the level of importing audio parts and outfitting vehicles using installation techniques based on information from the U.S. market. During the past few years, however, the installation techniques of Japan's specialty stores have improved dramatically. This fact was proven at last year's IASCA World Finals [*1] (which are held in the United States each year), at which time two vehicles entered from Japan received the highest rating for installation technique. In the area of sound development, however, the "ideal sound stage" has not been achieved.

For high sound quality playback, many high-end car audio systems will adopt a multichannel system. This system uses a channel divider to divide the playback bands so that only superior bands of speakers are used, and it connects dedicated amplifiers to each. As for the sound, hi-fi playback can be expected with minimal distortion in a wide band of frequencies; but the characteristics of the channel divider that joins the bands of each speaker unit will require fine adjustment in each vehicle. Moreover, the peaks and dips of the sound frequency characteristics that are produced according to the speaker mounting angle and shape of the vehicle compartment must be corrected as much as possible to a smooth curve to permit easy listening.

Tuning techniques also play an important role in achieving the "ideal sound stage" inside a vehicle. To faithfully play back the depth and position of each musical instrument recorded by music software, the time of arrival produced by the speaker's mounting position is corrected and each speaker's phase is adjusted. Thus, for vehicles in which advanced installation techniques alone do not solve the acoustic characteristic problems, the recently developed Sound Monitor 2000 model can be utilized, employing the latest digital techniques for solving such problems. This report will describe the aim of the product's development and its major design points, centering on the "digital preamplifier."

*1 The IASCA (International Auto Sound Challenge Association) World Finals is an international competition that focuses on car audio sound quality and installation.

2. Aim of development

The Sound Monitor 2000 model was developed with the following goals in mind :

(1) Playback sound (sound stage)
Reproduce a "concert sound stage" to listeners seated in the driver's seat and passenger's seat.

(2) Design
Enable adults should be able to understand and use the product with ease, and feel pleasure when using it.

(3) Product finish
Adopt high-quality materials that constantly arouse one's interest and that are suitable for a high-quality product, and develop new components for improving visibility and operation feel.

(4) Product specifications
Incorporate the world's most advanced specifications not only for high-quality design but for product electrical performance.

3. Year 2000 model lineup

For this high-grade system, a digital control amp tuner (DTA-500X) with DSP (digital signal processor) was developed in addition to a 1DIN-size in-dash six-CD auto-changer (ICD-500X) that can freely control six CDs.

Also developed as a basic model was a CD tuner (CDT-400X) that can be connected to the ICD-500X via digital-signal input-output expansion terminals.

4. Design concept

The design of the Sound Monitor 2000 model was based on the concept of "an expression of intellect and the future." The phrase "high-end car audio systems for specialty stores" tends to portray a design aim of high quality and reliability, as well as a product that lacks newness and individuality. To strongly emphasize the recreated sound monitor of this model, we implemented a full model change and greatly changed the image of the former model. We believe that this model has established the direction for future high-quality models. Specific design points are listed below.

(1) Improved sense of comfort and operability
The overall form has a symmetrical design that evokes an image of stability; and with the adoption of 2-axis knobs in the main control unit, clear operation differentiation was created. The principal operations are centralized in the knobs that are laid out to the left and right, which are manipulated by turning and twisting action. It is an orthodox technique, but is an effective way to manage the multifunctional controls.

(2) Representation of status
The thickish hard acrylic and the aluminum plate

arranged to the rear project a feeling of quality, quantity, and depth in the surface composition. Moreover, placing the acrylic in the front and making use of the metallic characteristics of the aluminum provide nighttime lighting to the panel. The color of the aluminum is a European smoked silver. Combined with the display explained later, the overall color is a monotone that expresses "intellect and the future."

Meticulous care is taken not only with regard to the front panel design but with regard to the plug material/shape, code color, and case finish. The objective was to project a feeling of status from the overall product.

(3) Display that projects feeling of precision

For the display element, a full-dot VFD (vacuum fluorescent display) was adopted. To match the exterior, provide better visibility, and express a sense of advancement, white was selected as the color. The advantage of using a dot matrix display is that a parametric equalizer Q curve and level expression is utilized, and digital control information is transmitted visually to the user.

(4) Consideration for installation

For installation in the vehicle compartment, a plate has been added to conceal the gap with the console panel. This plate can be divided and gap-free installation is possible with 2DIN-size installation involving the DTA-500X and the ICD-500X in-dash changer.

5. Mechanism design

During the design of this product, the developers created a concept after repeatedly debating what kind of quality was truly expected of a high-end car audio system. When designing the mechanism, engineers constantly thought of ways to produce quality that would be equivalent to that of high-end home audio systems while working with a limited space of 178 mm 50 mm. An overview is provided below.

5.1 Display unit

In order to obtain quality that is appropriate for a sound-field correction parameter display, equalizer curve display, or high-end car audio system display, a full dot matrix VFD display was adopted, providing a detailed display capability. The VFD lays out phosphor directly in a 1616 (pitch of approximately 0.3 mm) dot matrix placed on a silicon semiconductor chip that is referred to as an active matrix. This display is capable of producing ultra-high brightness and high-density graphics that a conventional VFD could not achieve (Fig. 1).

The actual product secures a display area of 59.19.7 (total dots = 6,144) by laying out 24 modules. And since a static drive is employed, it produces the advantages of low power, low noise, and minimal effect on sound quality.

Originally, the light emitted from the VFD is green. To produce white, which is the luminescent color emitted by the design image, a polarizing color filter was used. Normally, an existing product is used, but exclusive colors were blended to provide a subtle tone and texture.

This has created a subtle feeling that the texture will not be harmed as a result of the permeability being too high, exposing the interior to view; nor will the permeability be too low, diminishing the brightness.

Furthermore, as much depth as possible is provided from the surface of the front panel to the surface of the display; and when the power is turned on, the dark interior is lit up for visibility. These steps improve the feel of the product.

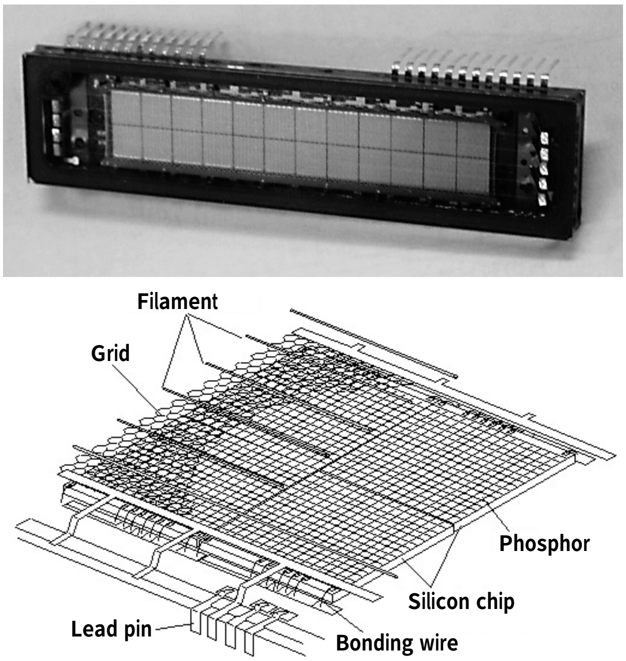

Fig.1 Outline View and Structure of VFD

5.2 Front panel

Fig. 2 shows a structural drawing of the front.

The front panel does not simply use colorless, transparent acrylic for the main parts; rather, it uses an acrylic panel that has been carved from a light-green-colored plate and that projects a glass-like image. Thus, it succeeds in producing a clear texture. And since the cut surfaces are buffed and mirror-finished, its features are utilized even further.

The next item that catches the viewer's eyes is the aluminum sheet. Normally, to provide the texture of aluminum, aluminum foil is often affixed. Initially, these parts were also manufactured with aluminum foil;

however, demand grew for a texture that approached that of actual aluminum, so a quick change was made to 0.5-mm aluminum sheet. The unique lightness of foil disappeared, giving way to a heavier, more solid appearance.

The designers instructed that clearance be provided between the acrylic and aluminum. This was difficult to do from a design aspect, but after some debate, a clearance of 1.5 mm was provided. This was a rather large-scale modification, but when the finished products were compared, it was clear that this contributed to a much better feeling of depth and to the improvement of the texture.

With priority given to operability, the knobs were symmetrically arranged to the left and right and were made from aluminum machining material similar to that used in high-end home audio systems. A diamond cut was given to the end for accent. Along with the soft feel that is unique to metal, the product produces a feeling of high quality not found in mass-produced goods.

5.3 Structure of case

Meticulous attention was given to the case, too. The structure is simple, with split upper and lower sections, to achieve better rigidity and improved sound quality.

Furthermore, the case was copper-plated and a clear coat was added to prevent oxidation. The application of copper plating, a nonmagnetic material, minimizes the effects of external noise from magnetic bonding as well as the effects of radiation noise produced from internal digital circuits.

The RCA cable for this product (for voice output) is 6N line (wire with at least 99.9999% copper content); while the plug was manufactured from cut aluminum using our company's original design rather than using a general-purpose formed-resin product. As a result, not only is the appearance improved but the sound quality is too, since audible distortion from the cable is greatly reduced.

6. Electrical design

An important element of in-car audio equipment development is the achievement of higher performance in the hardware (music playback equipment). But a vehicle compartment, as a music playback environment, has complicated acoustic characteristics that make it difficult to produce 100% of the original performance, even when high-performance hardware is installed.

During the electrical design phase, emphasis was placed on two points: ① correction of vehicle compartment acoustic characteristics and ② development of higher performance hardware. An overview is provided hereinafter.

6.1 Correction of vehicle compartment acoustic characteristics

When a vehicle compartment is compared to an ordinary listening room, there are certain differences, which are listed below. Improving these characteristics is an important element for vehicle-installed audio systems.

① The distributions of characteristic frequencies resulting from the vehicle compartment shape and size extend from several tens of hertz to several hundred hertz, and peaks and dips occur in the acoustic frequency characteristics.

② The listening positions and distances to the speaker vary; and a sound image is oriented to the speaker near the listening position, diminishing the stereo sensation.

③ There are many limitations on a speaker's installation location; thus, to achieve broadband playback, a multichannel system (such as a 3-way system) is often constructed.

To solve these problems, the following were installed as sound-field correction functions: for (①), a "parametric equalizer" that can separately control the

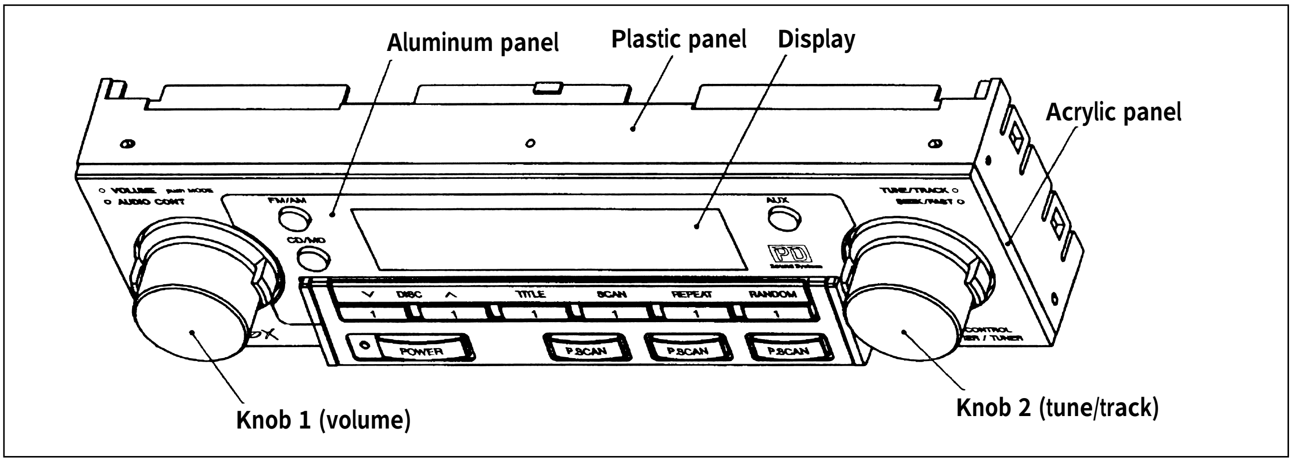

Fig.2 Front Panel Structure

equalizer's central frequency, Q, and gain; for (②), a "time alignment device" that adjusts the arrival time from each speaker to the listening position; and for (③), a "channel divider" that can control each speaker's playback band.

Installing each of these sound-field correction functions, however, leads to hardware complications and obstructs the achievement of higher performance. For that reason, this product has simplified the hardware circuit configuration by fully digitalizing the preamplifier circuits, including the various sound-field correction functions.

6.2 Internal circuit configuration

Fig. 3 shows a block diagram.

Digital signals that are compatible with the CP-1200 of the EIAJ (Electronic Industries Association of Japan) standards are capable of being received,[*2] are demodulated by a DIR (digital audio interface receiver), and are input to the DSP (digital signal processor). In contrast, AM/FM tuner analog signals are converted to digital signals by a 20-bit Δ-Σ-type A/D converter. The DSP digitally processes all functions, including volume adjustment; a 24-bit Δ-Σ-type D/A converter converts the signals to analog signals; then the signals are output after passing through an LPF circuit, which eliminates unnecessary harmonic signals. Thus, the circuit configuration is simple.

*2 The playback-capable sampling frequency is limited to 44.1 kHz.

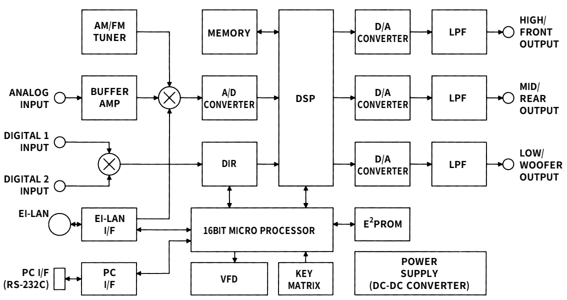

Fig.3 Block Diagram of Digital Pre-amplifier

6.3 DSP circuit

During the selection of devices, emphasis was placed on reducing the size of the circuits and minimizing interference to the tuner receiving circuit. Thus, a semiconductor manufacturer was requested to develop a 24-MHz/24-bit fixed-decimal-point DSP integrated circuit with built-in high-performance 20-bit A/D converter and 24-bit D/A converter.

Fig. 4 shows a block diagram of the DSP's internal signal processor.

The parametric equalizer and channel divider use IIR filters. As shown by the signal flow in Fig. 5, a feature of

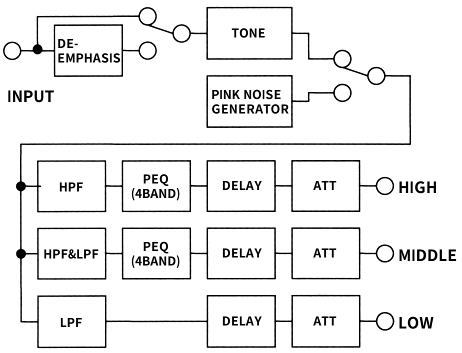

Fig.4 DSP Internal Signal Processing Block Diagram

the IIR filter is that it minimizes the DSP operation steps.

A disadvantage, on the other hand, is that characteristic deterioration due to operation errors can easily occur since the calculation results are also used in subsequent calculations. For this reason, characteristic deterioration has been reduced to a minimum through 24-bit×31-bit double-precision operations. Fig. 6 shows the differences between 24-bit×16-bit operations and 24-bit ×31-bit operations processed with a parametric equalizer (Fo = 100 Hz, Q = 5, and gain = 10 dB).

Next, the pink noise generator is a block that produces test tones, which are needed to measure the sound frequency characteristics inside the vehicle compartment. Pink noise is defined as "random noise of equivalent energy per unit octave." It is generated when white noise, which is digitally produced in the DSP, is passed through a low-pass filter (LPF) having the characteristic of -3 dB/oct. Thus, when sound frequency characteristics are being measured, there is no need to play back a CD on which pink noise is recorded. With a single button operation, test tones will be produced.

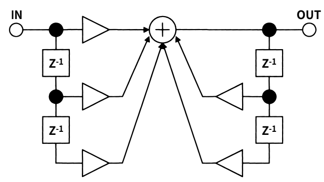

Fig.5 IIR Filter

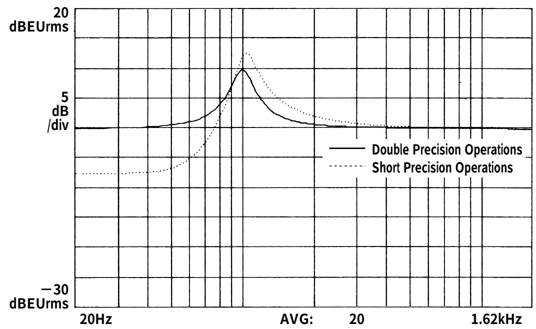

Fig.6 Differences between Frequency Characteristics with Short Precision Operations and with Double Precision Operations

6.4 Volume control

Conventionally, volume control devices have included mechanical controls, which operate based on sliding resistance; and electronic controls, which utilize an analog switch to change the resistance values. With a mechanical control, sliding noise and gang errors (variation in resistance between controls) may increase as time passes and the environment inside the vehicle compartment worsens.

Furthermore, to prevent digital clips from occurring when an equalizer is variable in systems that use a DSP (digital signal processor), volume-included control is often implemented and microcomputer-controllable electronic controls are often adopted. When an electronic control is used, however, ① circuit size increases and sound quality worsens, ② noise and distortion increase since semiconductors are used, ③ separation worsens, and ④ switching noise is produced when the control is adjusted.

On the other hand, a digital control, which adjusts the sound volume at the digital stage, can solve the aforementioned problems. But to reduce the quantization noise that becomes conspicuous when the volume is reduced, a 24-bit advanced multibit ΔΣ-type D/A converter that exceeds an S/N ratio of 113 dBDR = 113 dB was adopted.

Fig. 7 shows a signal spectrum for "digital volume operation that uses a D/A converter to play back a 40-dB attenuated signal inside a DSP" and "analog operation that attenuates D/A converter output 40 dB with built-in volume control," using a 24-bit advanced multibit ΔΣ-type D/A converter. As is evident, the difference in noise level between the two systems is not large, but it is clear that the analog system produces harmonic distortion in the vicinity of 2 kHz or 3 kHz.

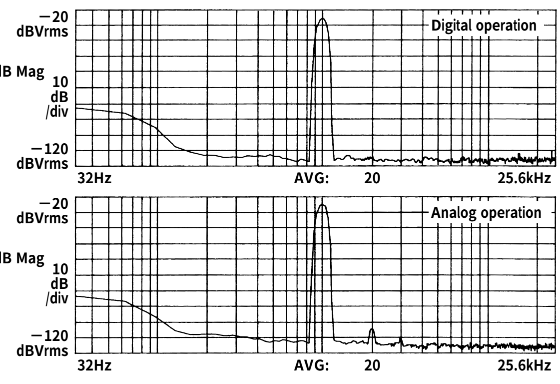

Fig.7 Comparison between the digital and the analog attenuated spectrum

6.5 Power circuit

As shown in Fig. 8, the power supplies for each circuit block are separated when there are mixed analog and digital products. This action prevents interference between circuits, which can adversely affect sound quality.

Initially, a compact three-terminal regulator that obtained highly stable voltage was adopted for the 5-volt power supply of the D/A converter, which has a great effect on sound quality; however, the results of a listening evaluation revealed that there was a "lack of divergence and speed." It is difficult to find a simple connection between the results of the listening evaluation and the correlation between the physical characteristics; however, the regulator has a built-in feedback circuit for maintaining a constant voltage when there are fluctuations in the output voltage, and the response time of this circuit and the noise generated by the device itself are thought to have an effect.

Although the size of the circuit increases, the power circuit that has been adopted combines shunt regulators and transistors. Optimizing each constant solves the "lack of divergence and speed" problem, without diminishing the stability of the power supply.

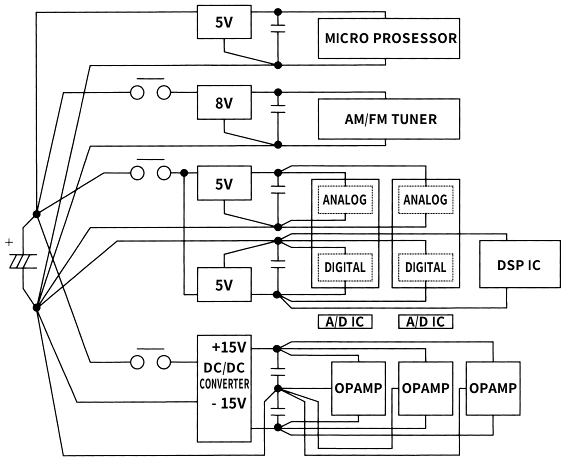

Fig.8 Block Diagram of Power Circuit

6.6 Sound development

It is often said that you will never get the sound quality you want (concert sound stage) if you focus only on the physical characteristics of the hardware. During the process of creating a product, the task of "confirming by actually listening to the music" is extremely important.

For this reason, we started by checking the sound of a component point by point, from the component to the circuit block to the overall circuit, in order to develop the sound. Moreover, the circuits were designed in a way that minimized the number of components existing in the audio signal line (Fig. 9).

The concept of sound development with a sound monitor involves the faithful playback of sound that has been recorded on a CD, and is based on the idea of adding nothing.

Thus, for the circuit board pattern, the power supplies for each circuit and ground, which are the foundation for signal playback, were first planned; then the pattern of the signal system was designed after the current flow was laid out in an ideal manner (patterned).
The effect of the power supply pattern design appears in the structure of the sound. Since the prototype stage, this product has had no major design changes attributable to an evaluation of sound. And during sound evaluations, differences in component-level sound played back remarkably.

Components, including integrated circuits, transistors, resistors, and condensers, were adopted after careful examination of each item, model (manufacturer), and rated value, while comparisons were made with other companies' products and conventional models, particularly centering on the power supplies.

These sound evaluations were conducted at the evaluation laboratory of our company's Sound Development Center and by using sound monitor demo cars. Using home audio systems, car amplifiers, and speakers for reference, we were able to faithfully play back sound, which is the foundation of the sound monitor.

7. Vehicle used to demonstrate this system

7.1 System overview

A 2000 Alfa Romeo 156, a small Italian sports car, was chosen as a demo car. Sound tuning was performed.

Fig. 9 shows the system installed in the demo car.

This system consisted of a digital control amp tuner (DTA-500X); in-dash six-CD auto changer (ICD-500X); and separate-power-supply amplifier (PAX), which was developed as a first-generation sound monitor.

To ensure that the speaker system reproduces a "sound stage" in front of the listener, the front speakers utilize a front two-way system (SG-5200), which is equipped with 2.5-cm-diameter soft dome tweeters and 13-cm-diameter mid-range speakers. The first topic of discussion was the installation position. For the mid-range speaker, a special baffle board was attached to the kick panel so that sound can be heard from the front, regardless of the number of passengers. Tweeters were installed in two locations: inside the side mirror (triangular corner) and at the front pillar (pillar A). Switches are installed to make it possible to conduct listening tests of the sound stage differences that result from installation position.

Initially, a large woofer unit was installed in a special box and placed in the trunk space. But because of a poor relationship in tone and frequency characteristics with the mid-range speaker, a small-unit 13-cm-diameter quadruple woofer with excellent response (SFX-5400) was installed on the rear tray, producing parallel drive with a total of eight channels arranged to the left and right.

Using the aforementioned system configuration, a frontward-oriented three-way multi-system was installed in a demo car and was able to produce a natural and pure sound stage.

7.2 Tuning with a personal computer

After the installation of each product is completed, it is necessary to adjust the sound-field correction parameters according to the vehicle compartment acoustic characteristics.

The sound-field parameters can all be adjusted by knobs and buttons on the product's main unit. Of course, it is also necessary to operate from inside the vehicle; and it takes time and effort to make adjustments while verifying the results of measurements taken by sound-field measuring instruments that are installed on the exterior.

Thus, the product was given an interface that can be controlled by outside computer, and computer-controlled software that can be used to graphically adjust sound-field correction parameters was developed.

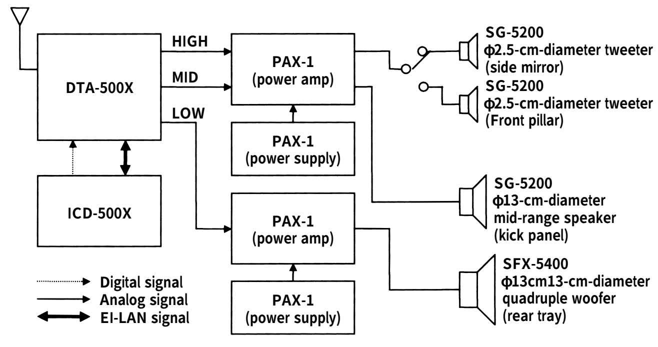

Fig.9 Demo-Vehicle-Mounted System

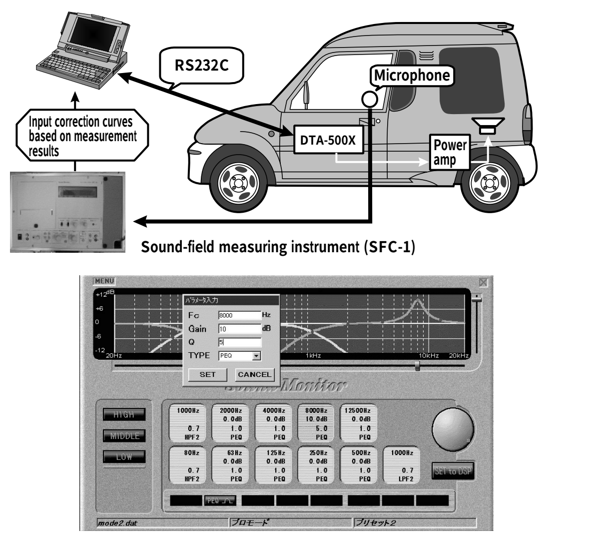

Fig.10 Parameter tuning system and a Example of Display Screen

(See Fig. 10.)

Thus, when the microphone of a sound-field measuring instrument (SFC-1, for instance) has been installed in a vehicle compartment, you can output pink noise from the product simply by using the pink noise ON setting from the personal computer. Next, while confirming the sound frequency characteristics with the sound-field measuring instrument, you can adjust the parametric equalizer, time alignment, and channel divider by using the computer screen and mouse. This is extremely convenient in that it frees you from the tedious work of having to open the door, make adjustments, close the door, and take measurements.

Furthermore, the product is designed such that, within its range of adjustment, the vehicle compartment acoustic characteristics can be satisfactorily adjusted. However, the results of listening verification showed that when it became necessary to slightly reduce the cutoff frequency of the channel divider, the product supported a custom function that enabled sound-field correction parameters to be set in units of 1 Hz. Fig. 11 shows an actual process flowchart. Based on parameter input, a personal computer calculates the DSP coefficients, and only required enumeration data is transferred to the product. Enumeration data that was sent to the product is transferred to the DSP; and at the same time, is written into nonvolatile memory. In this way, a custom function has been created without increasing the microcomputer load.

Settings can be called by button operation from the product side, and "Custom" appears on the product display.

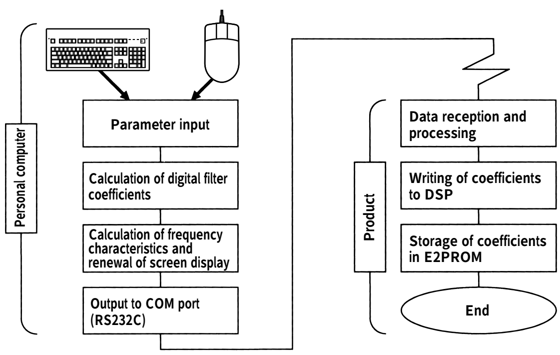

Fig.11 Parameter tuning processes

7.3 Acoustic characteristics

Fig. 12 shows sound frequency characteristics that were measured from the driver position, and Table 1 shows the sound-field correction parameters for that time.

A time alignment adjustment is extremely difficult to do by using a measuring instrument. Actually, the distance from each speaker to the listening point is measured with a tape measure, the delay time given to each channel is calculated, and the parameters are set. Then monophonically recorded music signals including vocal signals are played back; and while the position of the vocal sound image is verified, the delay time is fine-tuned. Adjustments were thus made using this procedure.

8. Conclusion

This report has explained the major design considerations and development aims of the 2000 Sound Monitor System.

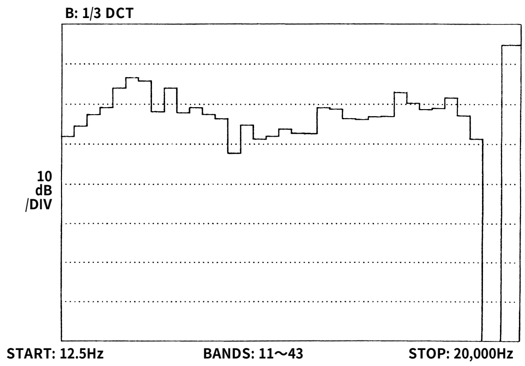

Fig.12 Acoustic Frequency Characteristics

Table 1 Field Adjustable Parameters

Channel divider				
Output channel	Filter characteristics	Cutoff frequency	Slope	Phase
HIGH	HPF	3.15KHz	12dB/oct	
MIDDLE	LPF	3.15KHz	12dB/oct	
	HPF	80Hz	12dB/oct	
LOW	LPF	80Hz	18dB/oct	NORMAL
Time alignment				
Output channel		Level	Delay	
HIGH-Lch		0dB	1.4msec	
HIGH-Rch		0dB	2.0msec	
MIDDLE-Lch		-1dB	0.5msec	
MIDDLE-Rch		-2dB	0.5msec	
LOW-Lch		0dB	0msec	
LOW-Rch		0dB	0msec	
Parametric equalizer				
Output channel	Band No.	Central frequency	Q	Gain
HIGH	1	5KHz	5	-2dB
	2	16KHz	3	+1dB
	3	—	—	—
	4	—	—	—
MIDDLE	1	1.6KHz	4	-3dB
	2	—	—	—
	3	—	—	—
	4	—	—	—

Table 2 Specifications

Preamp unit		DSP unit	
Frequency characteristics	20Hz~20KHz(-1dB)	Parametric equalizer	High/mid 4 bands each
Harmonic distortion	0.004%		Level: ±10dB
S/N ratio	113dB(IHF-A)		Central frequency: 63Hz~16kHz (1/3 oct)
Separation	90dB(at 1KHz)		Q: 1-5 (5 stages)
Dynamic range	105dB	Channel divider	Crossover (low-mid): 63Hz~160kHz (1/3 oct)
Output level	5.6Vrms		Crossover (mid-high): 1KHz~10kHz (1/3 oct)
			Slope: 6・12・18/oct
		Time alignment	0~5msec(0.1msec)
Tuner	FM	AM	
Received frequency	76.0~90.0MHz	522~1629KHz	
Practical sensitivity	12dBf(S/N 30dB)	22uV(S/N 20dB)	
Frequency characteristics	30Hz~15KHz	—	

Even before being officially announced, this system has been discussed greatly in car audio magazines and in the audio sections of car magazines, where it has received high praise. An Alfa Romeo 156 that was equipped with the system has also been popular at high-end audio events, and the number of people wanting to test the system never diminishes. The original aim of product development, which was "the feel of a concert sound stage," has just about been achieved.

To reproduce this "sound stage" to a greater degree before the listener in the future, a compact broadband speaker and high-sound-quality, high-efficiency amplifier will be developed, with the aim being to "produce the best possible music playback that can be achieved inside a vehicle."

The product concept and design have been highly rated by newspapers and magazines, and the Sound Monitor 2000 has been selected as a "good design" product.

References

• Ise Electronic Corporation "Features of Application Notebook CL Series"

Profiles of Writers

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Isao Yokoyama

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Takashi Ota

Joined company in 1984. Since that time, has engaged in development of car audio equipment. Currently works in Design Section of Engineering Department at AVC Headquarters, Division No. 2.