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Characteristics Tuning System for Automobile Audio System Components

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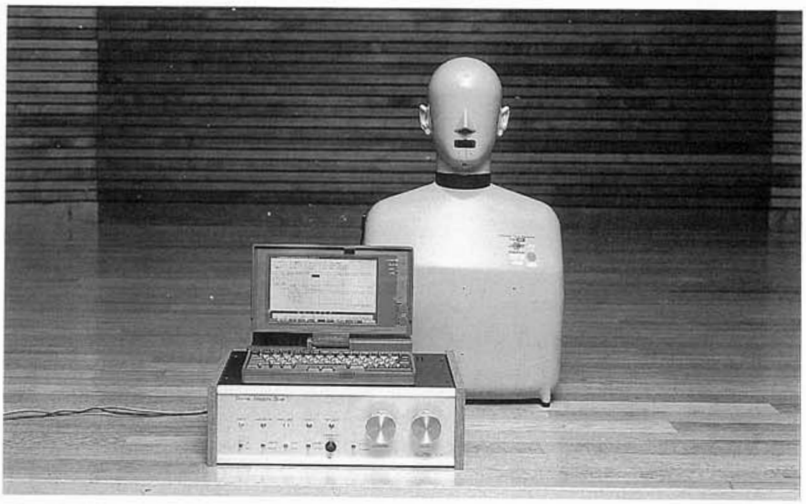

To increase the efficiency of acoustic testing and evaluation of car audio systems, we developed a new type of characteristics tuning system for automobile audio system components. This system integrates the functions that have been conventionally implemented with several analog devices into a single unit. By applying Fujitsu Ten's digital signal processor (DSP) technology, we changed signal processing in the system from analog to digital. This system enables many acoustic experiments of high-quality sound. With an equalizer and other adjustments of acoustic characteristics and a power amplifier, this system also serves as a kind of performance simulator. Acoustic measurement and analysis functions are also built in. This system has the following features:

(1) Can be almost totally controlled from a personal computer.

(2) Processes signals digitally to minimize degradation of the information quality. In acoustic evaluation, the digital signal processing enhances the sound quality.

This paper discusses the background and purposes of development and the outline and functions of the system.

1. Introduction

With the growing popularity of CDs, the percentage of car audio systems containing CD players is increasing as is the percentage in home audio systems. Car audio systems now have much higher sound quality than those of just a years ago. Each year, the designers of car audio systems are expected to produce systems featuring increased sound quality.

Automobile manufacturers produce cars in several price ranges by changing the development concept according to the intended users and their age groups. For car audio systems, the manufacturers are expected to design and supply unique products having a variety of acoustic characteristics.

To quickly develop high-quality audio systems that meet all requirements, we must use the experimental data and improve the level of sound quality adjustment.

We therefore developed a new characteristics tuning system for car audio system components consisting of a variable characteristics equalizer and a measurement and analysis system. The system was developed for the following reasons:

1) To raise the efficiency of sound quality adjustments using a personal computer
2) To achieve a higher level of sound quality, giving more functions and increasing the data processing speed with a DSP

The next sections describe the background and purposes of development, the outline and functions of the system, and the effects of development.

2. Background and purposes of development

2.1 Development and design of car audio system

If we check the passenger compartment of a car from an acoustic point of view, we find window panes and other sound reflecting items as well as seats and other sound absorbing items, both existing together in a small space. For listeners, several speakers are positioned asymmetrically. Compared to an ordinary listening room, an automobile presents a very different acoustic environment. Since the acoustic characteristics in this space are far from optimum, an equalizer is generally used to correct the characteristics.

When a car audio system is designed as an integral part of a specific car model, the acoustic characteristics of the interior can be monitored during development. By adding an acoustic characteristics correction function to the specifications of the audio system, an optimum acoustic environment can be provided.

Figure 1 shows Fujitsu Ten's design and development flow of a car audio system for a specific car model. This flow can roughly be divided into three steps

1) Planning specifications
   The specifications and performance of an audio system (including the speakers) are determined after the development concept of the car is understood.
2) Developing system components
   Speakers, amplifiers, and other system components are developed and their individual performance is measured and evaluated
3) Determining the specifications of the component characteristics tuning and audio system in an actual car

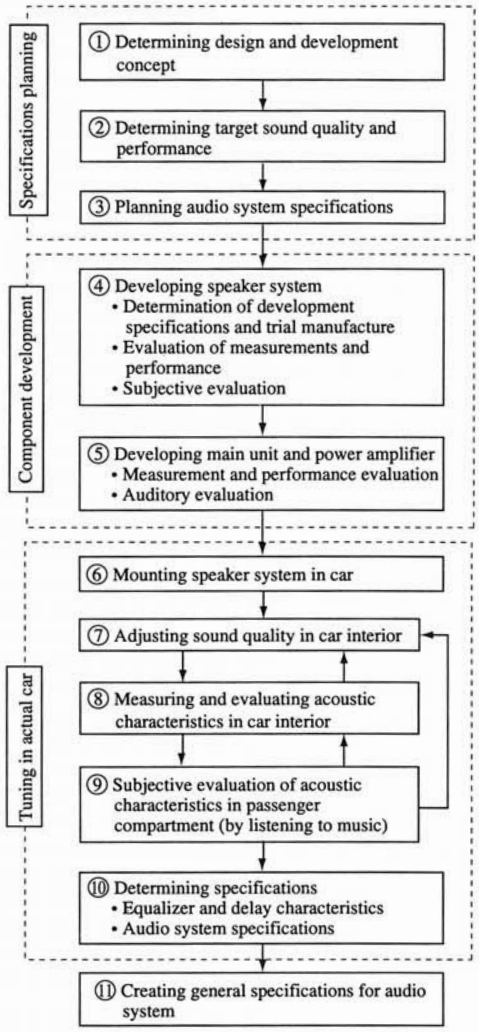

Figure 1. Flow of car-audio system design and development

The speakers and other components are mounted in a car and the sound quality and acoustic characteristics in the interior cabin are repeatedly measured and evaluated. From this data, the audio system specifications are determined.

In the flow of design and development, the simulation and evaluation (tuning) of optimum acoustic characteristics of the components (steps ⑦ to ⑨ in Figure 1) greatly affect the sound quality of the overall system. Therefore, we spend a great deal of time on these steps. Tuning corrects the acoustic characteristics of the passenger compartment by adjusting the balance of speaker output and the equalizer characteristics and other parameters. Tuning produces acoustic characteristics that conform to the development concept of the car.

Data obtained by measuring the interior is used for tuning. However, since the actual sound quality cannot be evaluated with only data based on physical measurement,

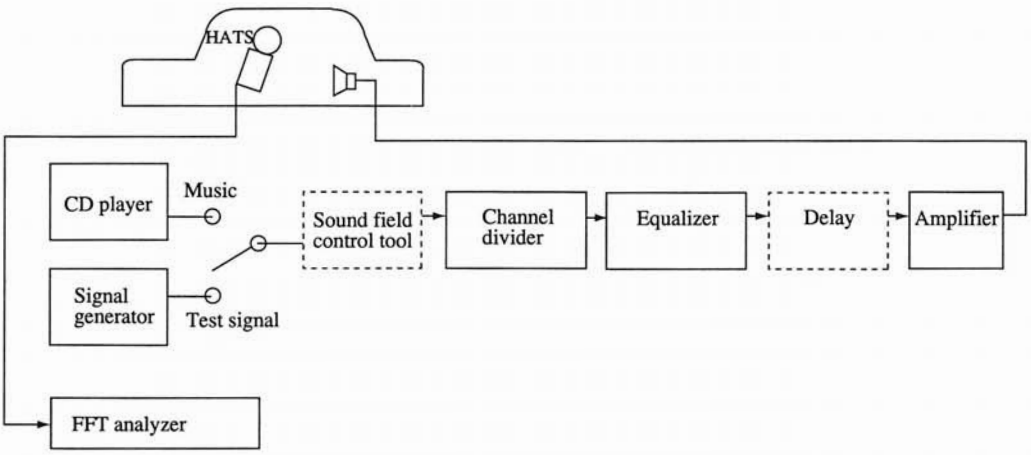

Figure 2. Block diagram of conventional tuning system

music sources are also listened to. By repeating these two types of measurements, the specifications of the equalizer and of the other functions are determined for an audio system.

2.2 Problems with a conventional tuning system

Conventional tuning system have several problems, as follows:

1) Unsatisfactory tuning efficiency
   ① Since many hardware devices are connected, it takes a great amount of time to prepare for and conduct experiments. (Figure 2)
   ② Since the tuning system has large-scale components, setup outside the passenger compartment is required, necessitating external system operations. Frequently, a worker must get out of the car for parameter adjustment, then get inside to listen.
   ③ The transfer characteristics (gain frequency response and phase characteristic) of an equalizer cannot be checked when the equalizer is connected to the tuning system. Adjustment of the equalizer parameters requires disconnection of the equalizer from the system each time to measure the transfer characteristics. Therefore, it is difficult to check the transfer characteristics as required during daily operations.
   ④ As most signal processors in car audio systems are now digitized, it has become necessary to tune new factors, such as sound field control and signal delay processing. Controlling the new factors along with the conventional ones, however, increases the scale of the tuning system and complicates operation.
2) Sound quality of the tuning system difficult to maintain
   Conventional tuning systems use only analog devices for hardware components. Each device has a satisfactory sound quality. However, since CDs have enhanced sound source quality, deterioration in the overall sound quality of a tuning system employing several hardware devices can no longer be ignored. This possible deterioration adversely affects the ability to evaluate the sound characteristics of high-quality acoustic system products.

2.3 Purposes of development

To solve the problems discussed above, we proposed a system to meet the following goals:

1) Enhancing tuning efficiency
2) Enhancing the sound quality of the tuning system and improving the sound quality evaluation techniques of Fujitsu Ten
   To achieve the aforementioned goals, we took the following measures:
   ① Integrated the hardware components of the tuning system into one unit, making the hardware compact and lightweight and simplifying hardware connections
   ② Put all the functions of the main unit under control of a personal computer to enhance operability (remote system control using a notebook personal computer)
   ③ Displayed the transfer characteristics of the equalizer on a personal computer for easy checking

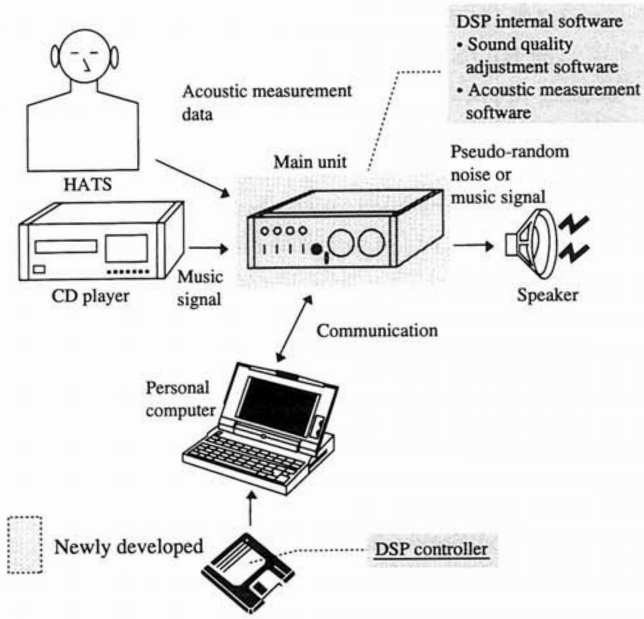

Figure 3. Setup of new tuning system

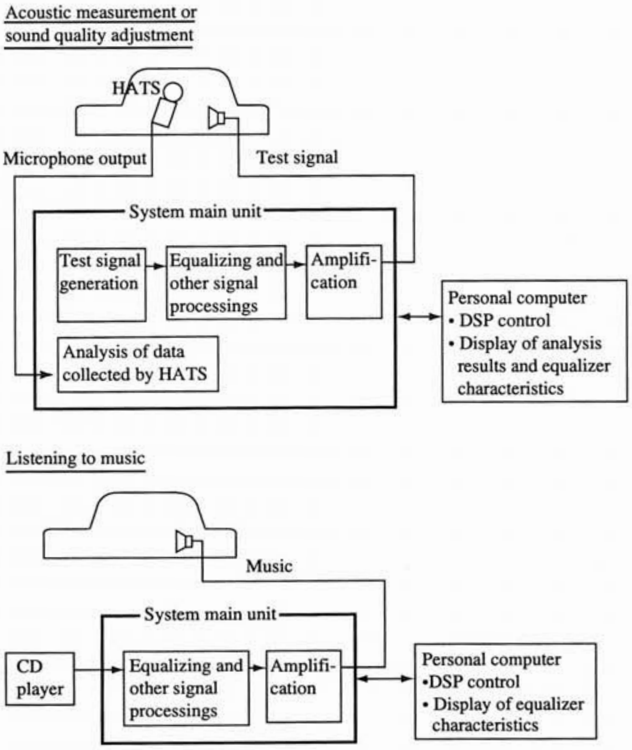

Figure 4. Block diagram of new tuning system

④ Processed all signals digitally in the tuning system to minimize signal deterioration and enhance the system sound quality

3. Outline of the tuning system

3.1 System configuration

Figure 3 shows the configuration of the tuning system. This system consists of the following three hardware sections:

1) Main unit
2) Personal computer
3) Head and torso simulator (HATS)

our system has two primary functions:

① Acoustic measurement (e.g. gain vs. frequency characteristics in the passenger compartment)

② Acoustic characteristics adjustment (e.g. equalizer for adjusting sound quality)

The main unit of the system processes signals to implement the above functions. The internal circuitry in the main unit can be switched according to purpose. Figure 4 shows the functions and connections of the main unit for tuning. The main unit has functions that correspond to the Fast Fourier Transform (FFT) analyzer and equalizer of the conventional analog system.

Since the main unit also has a power amplifier, connections between hardware devices are greatly simplified. The personal computer shown in Figure 4 can be taken inside the passenger compartment to adjust the parameters while the technician listens to the music. For acoustic measurement, a sound-gathering HATS is used. The HATS consists of a dummy head with a torso which has a microphone at (the entry of the external auditory canal on each side) each ear.

3.2 Outline of Individual System Components

3.2.1 Main unit

The main unit contains a DSP, an A-to-D converter, a D-to-A converter, and a power amplifier. Table 1 lists the principal specifications of the main unit. The DSP processes signals and operations digitally for the acoustic measurement and other functions. This digital processing nearly eliminates signal deterioration and produces a much higher sound quality than conventional systems.

We designed the main unit to increase the efficiency of acoustic experimentation and evaluation. In addition, we established general-purpose specifications to increase the versatility of the system for additional acoustical experiments by doing the following:

Table 1 Major specifications of system main unit

Input	Analog	One system (2ch)
	(A-to-D converter)	Resolution: 16 bits Fourth-order $\Delta\Sigma$ system 64-times over-sampling output
	Digital	One system (2ch)
Output	Analog • Preamplifier output	Three systems (6ch) (4.1Vrms)
	• Power amplifier output	6ch (30W×6) (With 4Ω load)
	(D-to-A converter)	Resolution: 16 bits
	Digital	Three systems (6ch)
DSP	Operation precision	18E6 × 18E6 = 24E6
	Cycle time	60ns
	Input data length (bits)	16
	Output data length (bits)	18
	Quantity	3
Electrical characteristics	Full-harmonic distortion rate	
	• Preamplifier output	0.005% max. (0 dB, 1 kHz)
	• Power amplifier output	0.05% max. (1 dB, 1 kHz)
	Dynamic range	
	• Digital output	110 dB
	• Preamplifier output	104 dB
	• Power amplifier output	84 dB
Power supply	Two-way system (100 VAC, 13 VDC)	
Dimensions	430 (W) × 140 (H) × 300 (D) mm	
Weight	About 7.2 kg	

1) Building all necessary functions of the tuning system, except the CD player, other sound equipment, and speakers, in the main unit to reduce the number of system components
2) Providing a digital I-O interface to connect an external digital system
3) Designing the system to run on both 100 VAC and 13 VDC to make the main unit available both on the test bench and in the passenger compartment.
4) Designing the main unit so the system is almost completely operable from a personal computer

3.2.2 Personal computer

The personal computer in our system has software to control the DSP and display the acoustic analysis results and equalizer characteristics. DSP control includes changing the measuring conditions and equalizer parameters, and requires complicated processings. Each time a parameter is changed, coefficients must be calculated and written into the DSP internal memory. The DSP control software built into the personal computer automatically executes complicated calculations and processes. This system can easily be operated from a personal computer. Even a new user can operate the system with knowledge of only the cursor, Enter, and numeric keys.

3.2.3 HATS

The HATS is used in a wide range of acoustic engineering applications, such as sound analysis in concert halls, engine noise evaluation, and binaural recording and reproduction. Since the HATS simulates the sound transfer functions at a simulated human head and torso, physical data closer to human hearing than microphone input can be obtained. The HATS reduces the hearing fluctuation.

4. System functions

This section details the acoustic measurement and sound quality adjustment functions.

4.1 Acoustics measurements function

The acoustics measurements function is used to analyze the acoustic characteristics at the HATS location. The DSP internal program generates an acoustic test signal, then measures and analyzes the signal. Figure 5 shows the flow of program execution. The DSP improves the speed of operation especially the FFT analysis. The acoustic test signal and measurement items are described below.

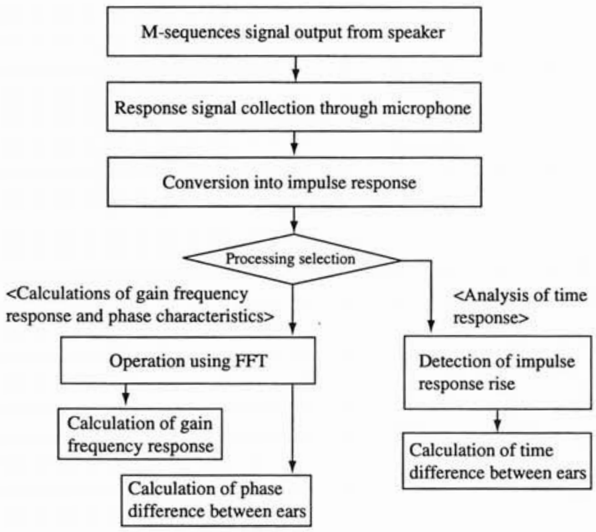

Figure 5. Flowchart of acoustic measurement program installed in DSP

4.1.1 Acoustic test signal

This system uses a maximum-length linearly recurring sequences signal (M-sequences signal) as the acoustic test signal. The M-sequences signal is a pseudo-random signal. Therefore, an impulse response in a sound field can be obtained from the relationship between the M-sequences regenerated signal and the sound response in the sound locale (car interior). The M-sequences signal is advantageous vis-a-vis the S/N ratio because it is not affected by noise as badly as a conventional characteristics test signal (white noise). The DSP in the main unit of our system generates the M-sequences signal, so an external signal generator is not required.

4.1.2 Analysis of acoustic-level frequency characteristic

The acoustic-level frequency is the primary characteristic to be evaluated in the car interior. Our new system measures and displays four data items related to this characteristic (Figure 6):

① 1/3-octave band frequency response
② 1/3-octave band sweep frequency response
③ Level balance in low-frequency, medium-frequency, and high-frequency bands
④ Phase difference of sound waves between the right and left ears

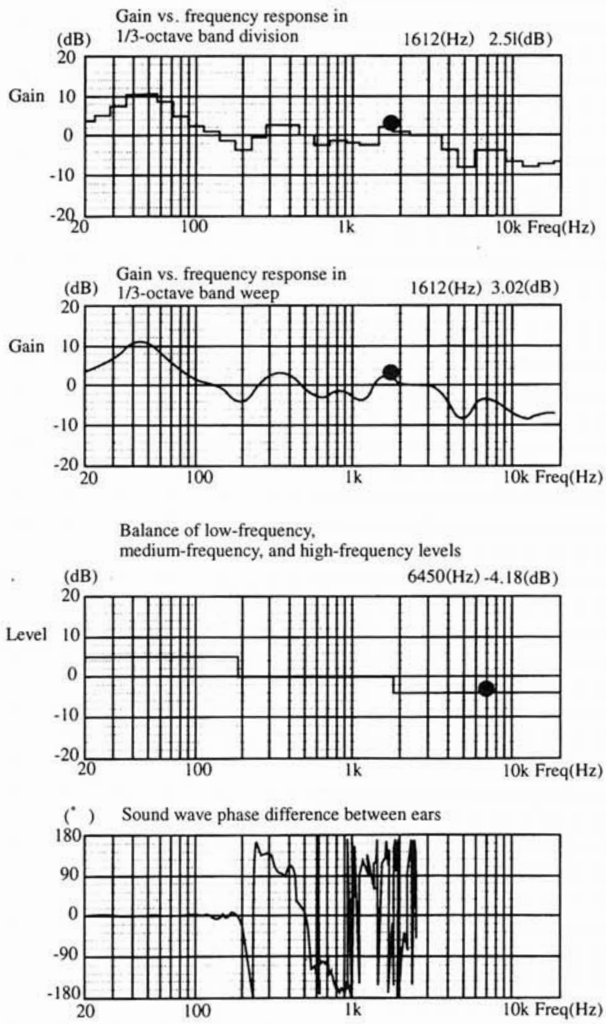

Figure 6. Example of measurement data display

② and ③ are unique analysis items of this system, and are not supported by FFT analyzers on the market. ① is generally used to evaluate acoustic-level frequency response. To calculate the frequency response in ②, a bandpass filter of 1/3-octave band is used to sweep from 20 Hz to 20 kHz. This new technique enhances the analysis precision and enables the equalizer parameters (central frequency and Q) to be determined more accurately. As for ③, the level balance can be checked easily in all bands because the total bandwidth is divided into three, as shown in Figure 6.

4.1.3 Analysis of time response

To observe how a sound reflected or refracted in the car interior reaches the ear over time, we analyze the following items related to the time response of a sound:

① Impulse response
② Time required for sounds to reach the ear

In this system, we added ② to the analysis items for the first time. This addition produces the advantages explained in the next paragraph.

Since the distance from the listening position to each speaker differs in the passenger compartment, the sound image is displaced and the sound wave phase is distorted. The influences of these symptoms cannot be corrected in a conventional analog system but can be in a digital system by adjusting the time delay of the signal. After the impulse response is measured, the time from when an impulse occurs to the time the sound reaches the listening position is measured and displayed in ②. With the time difference data, sound image displacement can be easily corrected using the time delay adjustment function explained in Section 4.2.

4.2 Acoustic characteristics adjustment function

The acoustic characteristics adjustment function is used to adjust the acoustic characteristics of an audio system including the passenger compartment according to the aforementioned results of acoustic measurement analysis. For this tuning, the main unit has an equalizer and delay functions. We used DSPs to obtain these functions.

Figure 7 outlines the signal processing using the DSP, including measurement the acoustic characteristics.

In this measurement, test signals go through the signal processing section of the equalizer. In sound reproduction from a CD or other source, output signals from the source go through the same signal processing section.

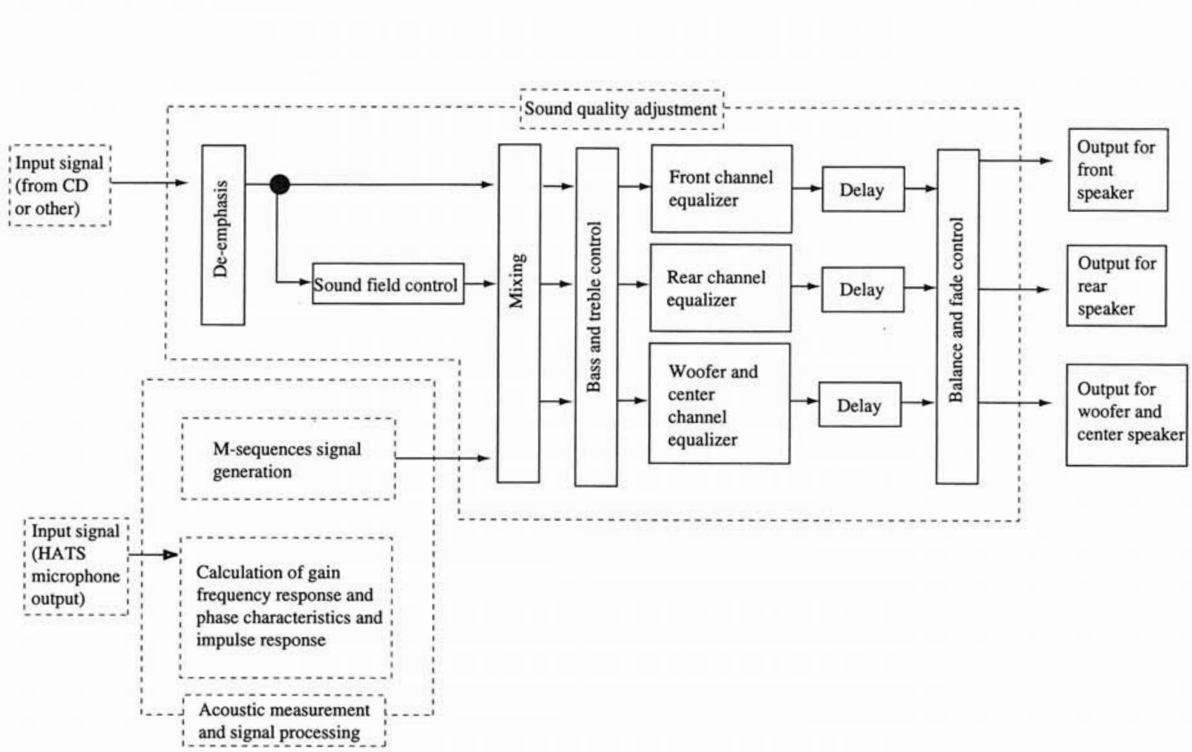

Figure 7. Outline of signal processing using DSP

Table 2 lists the processing contents of each function block shown in Figure 7. The recursive filter explained in Table 2 is called an infinite impulsive response (IIR) filter. By changing the coefficients at a0 to b2, equalizing characteristics curves of various shapes can be drawn. The DSP we developed for this system provides the signal processing or equalizing characteristics of the types listed in Table 3 for each recursive filter.

This DSP enables easy sound field control, sound signal delay, and other desired signal processing. The processing program can also be changed easily. By using the DSP, we were able to make the tuning system easier to operate and more efficient than conventional tuning systems.

The personal computer display supports seven adjustment functions for acoustic characteristics. Each function can be controlled on a single screen:

① Initial system setup
② Constant file loading
③ Constant file saving
④ Parameter adjustment (channel level and mixing)
⑤ Gain frequency response adjustment
⑥ Crosstalk adjustment between right and left channel signals
⑦ Sound field control (adjustment of initial reflection and reverberation)

Figure 9 is the gain frequency response adjustment screen. The system supports a graph display function for this screen so that the gain frequency response and phase characteristics can be checked visually. With this function, we can check the equalizer transfer characteristics without measuring them as in conventional systems (Figure 10).

Table 2 Main acoustic signal processing functions of DSP

Acoustic characteristics adjustment	Basic audio function	Primary recursive filter: 3 elements × 2 channels
	Gain vs. frequency characteristics adjustment	Primary recursive filter: 1 element × 6 channels Secondary recursive filter: 8 elements × 2 channels 5 elements × 2 channels 1 element × 1 channel
	Sound field control	Initial reflection and reverberation Primary recursive filter, 2 elements (for reverberative sound quality adjustment) Primary recursive filter, 2 elements (for initial reflection adjustment)
	Delay	6 channels (maximum delay time: 5 ms)

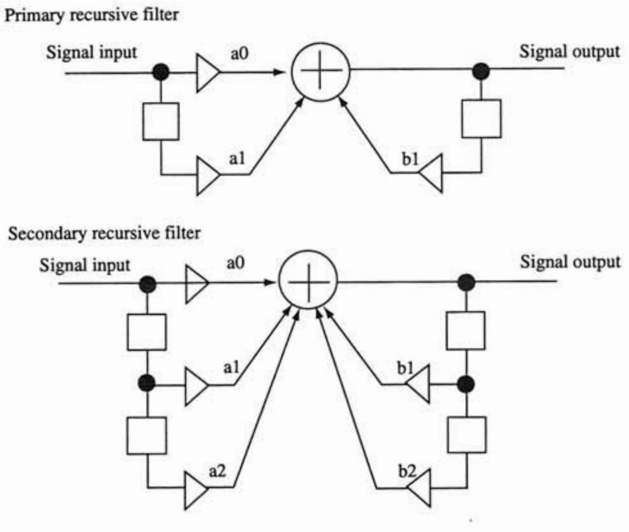

Figure 8. Recursive filters

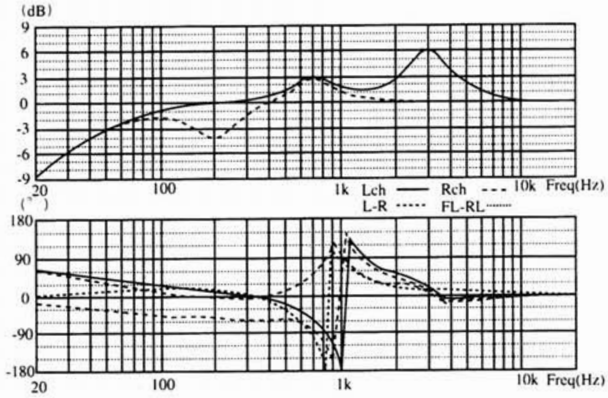

Figure 10. Gain frequency response and phase characteristics display screen

Table 3 Acoustic signal processing with recursive filters

	Acoustic signal processing (adjustable parameters)
Primary recursive filter	Primary low-pass filter (frequency) Primary high-pass filter (frequency) Primary phase shifter (frequency) Bass (frequency and gain) Treble (frequency and gain)
Secondary recursive filter	Equalizer (frequency, gain, Q) Secondary low-pass filter (frequency, gain, Q) Secondary high-pass filter (frequency, gain, Q) Secondary phase shifter (frequency Q) Processing with primary recursive filter

Frequency-Specification Control

DEEMPHASYS:OFF

Graph display: Display start (Press Enter to start graph display.)
Channel: FRONT (Press Enter to select the front or rear channel.)
LR operation: Independent (Press Enter to select independent or output mode.)

	Parameter	Bass	Trebl	Mid	EQ1	EQ2	EQ3	EQ4	EQ5	EQ6	EQ7	EQ8
L ch	fc (Hz)					700	3000		50	1000		
	Gain (dB)					3.0	6.0					
	Q					1.5	2.0			2.0		
	EQ-TYPE						1		6	4		

L and R common (EQ: 4 to 9 only) At 44 kHz operation only Front only

	Parameter	Mid	EQ1	EQ2	EQ3	EQ4	EQ5	EQ6	EQ7	EQ8
R ch	fc (Hz)		200	700			50	800		
	Gain (dB)		4.0	3.0						
	Q		1.0	1.5				2.0		
	EQ-TYPE						6	4		

(Blank for type 0)

0	1	2	3	4	5	6	7	8	9	10
GEQ	PEQ	*1	*2	*3	*4	*5	*6	*7	*8	THROUGH *9

*1: Secondary LPF
*2: Secondary HPF
*3: Secondary PS
*4: Primary LPF
*5: Primary HPF
*6: Primary PS
*7: Primary BSS
*8: Primary TREB
*9: Lowercase

Figure 9. Frequency response adjustment screen

Table 4. Comparison of conventional and new tuning systems

	Item	Conventional system	New system
Hardware	Component	Equalizer, channel divider, amplifier, FFT analyzer, signal generator, and HATS (sound field control tool and delay)	Main system unit, personal computer, and HATS
	Signal processing system	Analog	Digital
	S/N	90dB	Analog output: 104 dB Digital output: 110 dB
	Total weight (excluding HATS)	About 42 kg	About 10 kg
	Total volume (excluding HATS)	About 0.12 m^3	About 0.02 m^3
Software	Function control method	Individual hardware control	Integrated hardware control using personal computer
	(Number of measuring points between 20 Hz to 20 kHz)	31 (at 1/3-octave band division)	90 (at 1/3-octave band sweep)
	Other		New functions: • Display of equalizer characteristics • Automatic measurement of sound travel time from speaker to listening position
Tuning method		Repetitive execution of listening to music in car interior and adjusting equalizer and delay parameters from outside the car	Adjusting equalizer and delay parameters while listening to music inside the car

5. Development effects

Our development of a digital tuning system has had the effects listed below.

1) Efficient tuning

   Acoustic measurements and sound quality adjustments can all be controlled with a personal computer to enhance system operation. This has reduced the tuning time about 30% and produced the following effects:

   ① A personal computer can be taken inside the car to change the sound quality parameters while the operator listens to music. Slight sound differences and sound quality caused by changes in the parameter can easily be checked.

   ② The display of equalizer characteristics on a personal computer makes it easy to check the transfer characteristics and set the equalizer parameters while tuning the audio system.

   Table 4 compares conventional tuning systems with our digital tuning system.

2) High sound quality

   Since this system digitally processes all signals related to adjusting the acoustic characteristics, signal deterioration is minimal. The digital signal input terminal, which is part of the system, has enhanced the sound quality compared to conventional systems without impairing the quality of CD output information. The precision of sound quality evaluation is also enhanced.

3) Compact and lightweight tuning system

   The equalizer, amplifier, channel, divider, and other discrete components in conventional systems have been integrated to make system hardware that is compact and lightweight (1/4 the weight and 1/6 the size).

This compact and lightweight system facilitates acoustic experimentation in an actual car.

6. Conclusion

The new digital tuning system epitomizes Fujitsu Ten's DSP and sound quality evaluation technologies. However, the semiconductor devices and sound quality evaluation technologies used will continue to progress. By utilizing related technologies, we will enhance system performance.

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