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For Enthusiasts of High-quality Car Audio "Sound Monitor"

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The combination of compact disc audio and very quiet car interiors has brought us close to the dream of perfect in-car audio.
Our Sound Monitor series aims to be the best in terms of performance and quality of sound, to achieve the goal of "creating the ultimate in-car sound quality." To realize the full potential of the Sound Monitor series, we have developed a Sound Field Configuration Measuring Instrument (SFC-1) and a parametric Equalizer (FCX-3) so car audio workshops can create the optimum sound environment in every car.

1. Introduction

The popularity of CDs in the car audio market has prompted car audio makers to introduce their own high-quality hi-tech car audio to at tract more customers.

In 1989, Fujitsu Ten introduced a range of highly functional high-quality car audio products, called the α–series, designed for the diverse needs of our customers. Car audio in this field is aimed mainly at younger drivers, and is therefore very sensitive to the latest trends in functionality, design, and sound. Manufacturers must be quick to respond to this shifting market. The market size, however, does not justify high development costs, so it is sensible to design a small number of basic long-lived products which will win a following among audiophiles.

It is important for the real high-quality audio sought by audiophiles to reflect a consistent design philosophy.

This was our approach to developing a high-quality audio system to realize the audio engineer's dream of listening to the ultimate in sound quality in one's car to take the opportunity of Fujitsu Ten's 20th anniversary.

What follows is a brief description of the Sound Monitor system, including our development aims and design concepts.

2. Development aims

The system's development concept is to clarify the fundamental objects emphasis on sound for car audio. Car audio must provide better sound quality for the driver.

We have been working jointly with auto manufacturers towards creating sound quality for car-specific audio systems for many years. Sound field correction in the car has been one of the most difficult aspects of this task. There is no aftermarket audio system that guarantees defined characteristics in the end user's car. High-quality sound hardware must therefore have comprehensive field support to be successful.

We collected data on cars owned by purchasers of high-quality audio in order to customize our systems to their cars. This improves our coordination with dealers and helps us give better service to our customers.

Development was a combination of both hardware and software. New hardware technology improved physical properties. New circuits and components gave higher sound quality, and new materials and structures heightened the sense of texture and feeling. Software development included developing a sound field configuration measuring instrument and a parametric equalizer, and also developing installation and sound tuning techniques.

The main features provided by the sound field configuration measuring instrument (SFC-1) for sound tuning are as follows:

① Real-time 1/3 octave analysis
② Electrical frequency response measurement by sweep signals
③ Single-tone abnormal sound inspection at installation
④ Four-pen plotter data output
⑤ RS-232C interface to transmit results

3. System summary

3.1 System configuration

Figure 1 shows the system configuration. The system has a tuner deck (TCX-1), a control amplifier (CAX-1), a parametric equalizer (FCX-3), a CD auto-changer (CDX-1), and a power amplifier (PAX-3).

The control amplifier has analog and digital input terminals on the front panel to connect to various types of music source.

The F-bus communication line, shown as the double line(=) in the figure, allows the tuner deck to control the CD auto-changer. The dotted line (...) is a coaxial-cable digital audio signal transmission line.

3.2 Parametric equalizer (FCX-3)

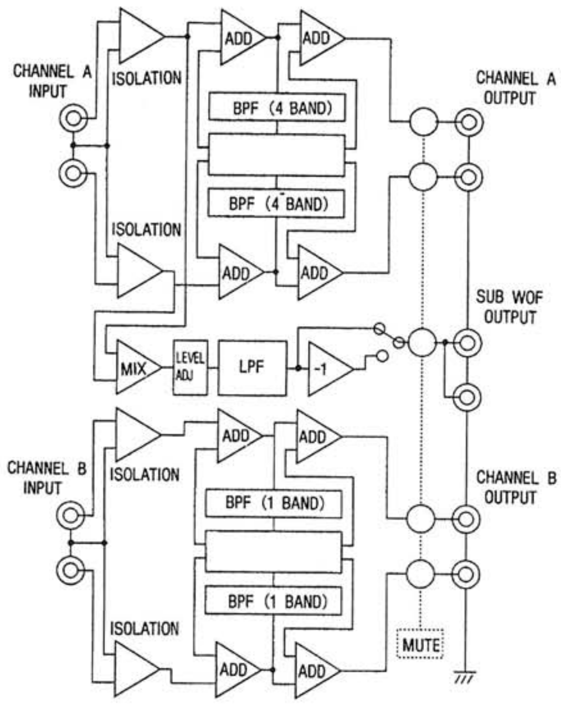
We developed a new parametric equalizer to adjust the center frequency, Q, and the gain to compensate the complex acoustic frequency responses measured by the sound field configuration measuring instrument (SFC-1).

3.2.1 Internal construction

Figure 2 is a block diagram of the parametric equalizer. The equalizer is made up of channel A with four bands and channel B with one band. Each band-pass filter (BPF) is a state variable filter so that the center frequency (F_0) and sharpness (Q) can be continuously tuned.

The subwoofer control converts the signal input from channel A to monaural, then feeds it to a low-pass filter (LPF). This low-pass filter is also a state variable filter which can match the cutoff frequency to the characteristics of the loudspeakers.

Adding in an equalizer generally degrades sound quality somewhat. We minimized this effect with a higher dynamic range given by a DC-DC converter, a high-performance operational amplifier, and an oxygen free copper (OFC) PC board.



Note: Only the audio section is shown.

Figure 2. Parametric EQ block diagram

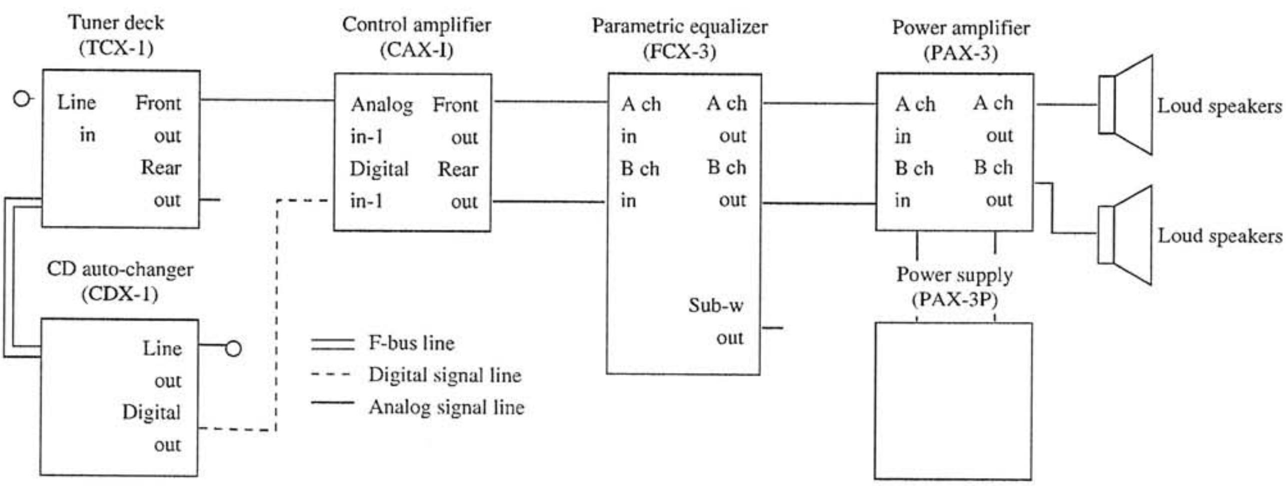


Figure 1. System block diagram

3.2.2 Parametric equalizer circuit

As mentioned earlier, the parametric equalizer uses state variable filters. These filters give LPF, HPF, and BPF output signals at the same time. The state variable filter is also used by its application to adjust the cutoff frequency with an LPF, as in the subwoofer control.

Figure 3 is a block diagram of the state variable filter. The transfer function of the BPF output is given by solving the equation:

$$H=\frac{(j\omega/\omega_o)(1/Q)}{(j\omega/\omega_o)^2+(j\omega/\omega_o)(1/Q)+1}$$

$$\omega_o=\frac{\sqrt{m}}{\sqrt{C_1C_2R_1R_2}}\ ,\quad Q=\frac{\sqrt{m}}{n}\cdot\sqrt{\frac{C_1R_1}{C_2R_2}}$$

The center frequency, ω_0 of the BPF can be adjusted while keeping Q constant by changing R_1C_1 or R_2C_2 while keeping the ratio R_1C_1:R_2C_2 constant.

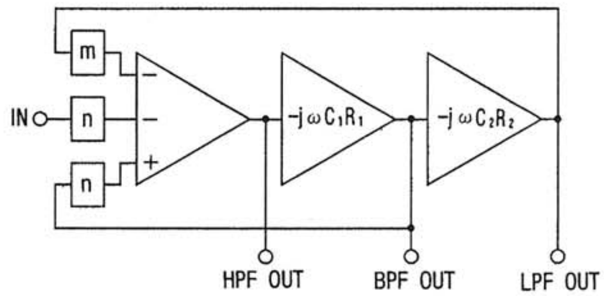


Figure 3. Theoretical block diagram of state variable filter

Q can also be adjusted while keeping ω_0 constant by changing n, the mixing constants, in the input addition/subtraction circuit while keeping ratio between mixing constants.

In the practical circuit implementation (Figure 4), this BPF is inserted in the feedback circuit of the operational amplifier so that F_0 and Q are continuously adjusted by changing mixing resistances R_1 and R_2, and R_3 R_4 at the same time. Gain is varied by controlling feedback to the operational amplifier.

Figure 5 shows the frequency response as these three parameters are varied.

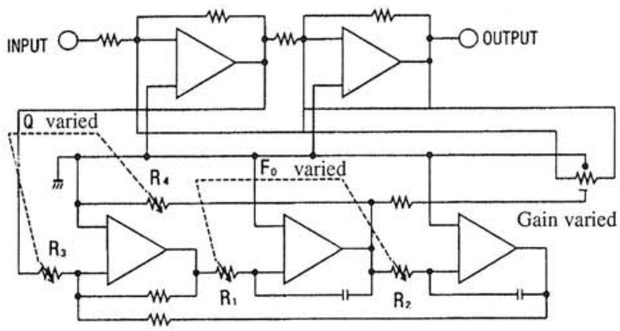


Figure 4. Parametric EQ circuit diagram

3.3 Control amplifier (CAX-1)

Digital source playback circuitry available up to now has had inherent problems. There was not enough space in the CD playback section for a high-performance D/A converter, the servo circuit and the D/A converter were too close together and susceptible to noise effects, and even with a high-performance D/A converter, the signal connection head unit designed for cassette deck playback degraded performance. To solve these problems, we developed a control amplifier which combines a high-performance D/A converter and an audio circuit.

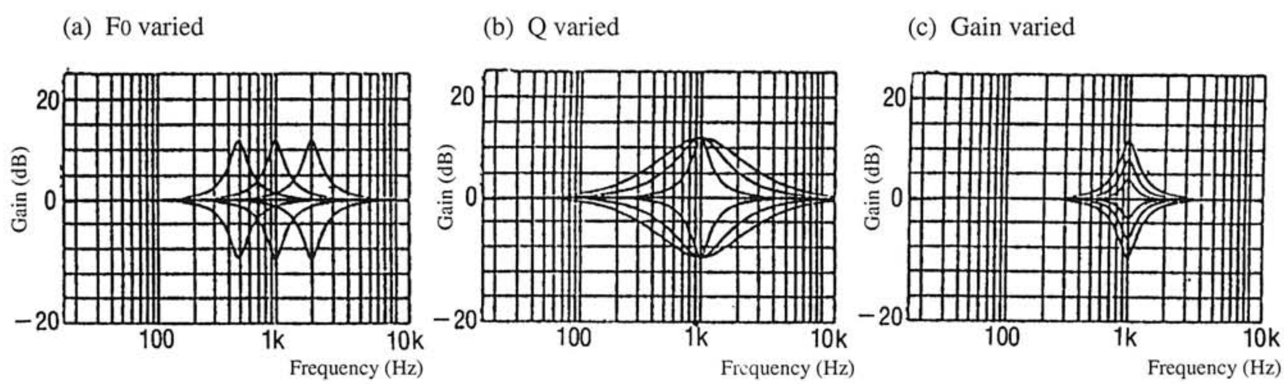


Figure 5. Parametric EQ curve

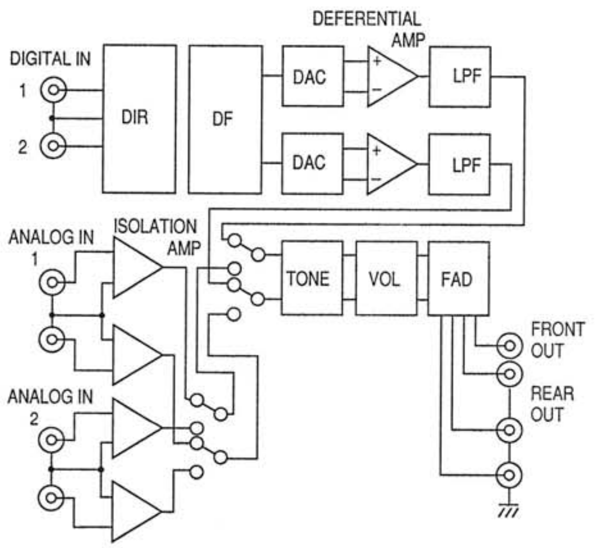
3.3.1 Internal construction

As shown in the block diagram in Figure 6, the control amplifier is broadly divided into a D/A converter, which decodes digital signals into analog signals, and a preamplifier, which switches input signals to adjust the sound tone and volume.

The preamplifier section switches inputs using high-quality relays. For sound quality control we added a middle frequency control (MID) to adjust vocal frequency gain. The D/A converter combines four interactive DACs for improved low-level signal reproduction. The second-stage LPF is a Bessel filter for minimum waveform distortion.

A DC-DC converter steps up the power supply to ± 9 V to ensure a broad signal dynamic range.

The preamplifier section and the D/A converter section are isolated from each other by a shielding plate which prevents digital noise from radiation.



Note: Only the audio section is shown.

Figure 6. Control AMP block diagram

3.3.2 Preamplifier section

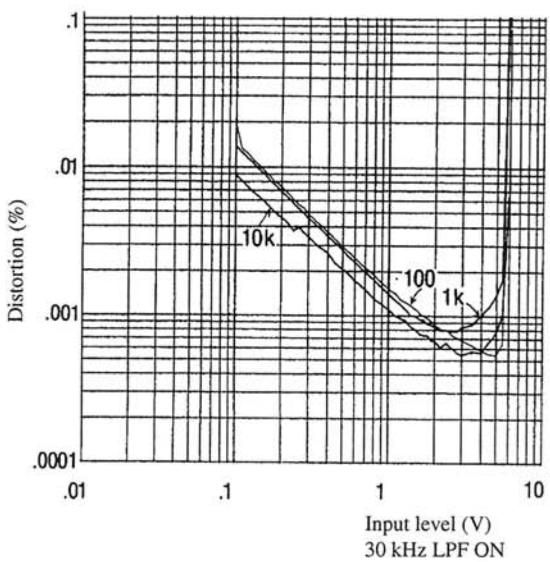
Our aim to achieve a new level of audio performance led us to review components and connection patterns as follows:

① Use relays for signal switching
To correct harmonic distortion
② Use low-distortion operational amplifiers
③ Eliminate chip components
To improve separation
④ Place guide patterns
To improve maximum input level
⑤ Higher dynamic range by using a DC-DC converter

The resulting preamplifier maintains the digital source quality, with a harmonic distortion of 0.0013% (1 kHz, 1 V) and a separation of 91 dB (1 kHz) as shown in Figure 7 (a) and (b).

The digital circuit is deactivated during analog input to minimize noise in the analog signal.

(a) Harmonic distortion characteistics



(b) Separation

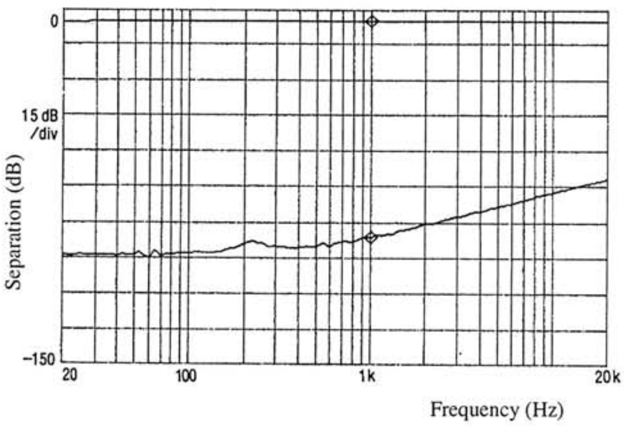


Figure 7. Separation and total harmonic distortion at the preamp stage

3.3.3 D/A converter

The D/A converter is made up of a digital decoder(DIR), a digital filter (DF), a D/A converter, a differential circuit, and an LPF. Features of the circuits that make up the D/A converter section are summarized below.

1) D/A converters

Figure 8 is a block diagram of a D/A converter. The digital input signal is split in the input circuit into the high-order 10 bits and the low-order 8 bits, and the high-order 10 bits are fed to the 10-bit resolution current output DAC via a digital offset circuit.

The dynamic range DR (10) of the DAC is:

$$\mathrm{DR}\ (10) = 6.02 \times 10 = 1.78 = 61.98\ (\mathrm{dB})$$

The low-order 8 bits are fed to a digital feedback circuit equivalent to a first-order noise shaper circuit for noise shaping.

The dynamic range (DR) of the noise shaper is given by the equation:

$$\mathrm{DR}(8) = 10\log\left\{\frac{9}{2\pi^2}\left(2^n - 1\right)^2 \bullet M^3\right\}$$

$$= 10\log\left(\frac{9}{2\pi^2}384^3\right) \fallingdotseq 74.2\ (\mathrm{dB})$$

where

n: Number of quantized bits ($n = 1$)

M: Over-sampling multiplier ($M = 384$)

The total dynamic range DR (18) is theoretically:

$$\mathrm{DR}\ (18) = 61.98 + 74.2 = 136.1\ (\mathrm{dB})$$

The noise-shaped signal is input to a 1-bit DAC. The 1-bit DAC calculates its output using a constant current and pulses from the noise-shaped signal. The constant current corresponds to 10-bit ($1/2^{10}$) in the 10-bit DAC. The D/A converter thus has less noise level rise and jitter than a general-purpose high-order noise shaping DAC.

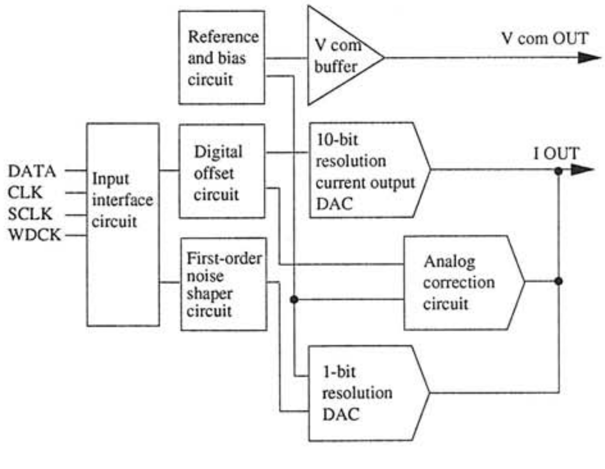


Figure 8. Block diagram of D/A converter

The DAC has a digital offset circuit followed by an analog correction circuit to prevent the zero-cross distortion generally found with the on/off states of the MSB of a 10-bit DAC. When the DAC input falls below $1/2^9$ of the highest input level, the part of the input signal on the more positive side on the bipolar zero (BPZ) is shifted to the negative side by switching bit 9, the MSB. This prevents the usual change in the MSB as the input passes the BPZ. The output of the DAC is restored to the positive side of the BPZ by a constant current source with the same weight as bit 9 (see Figure 9).

This feature removes the zero-cross distortion inherent in any general-purpose multibit DAC.

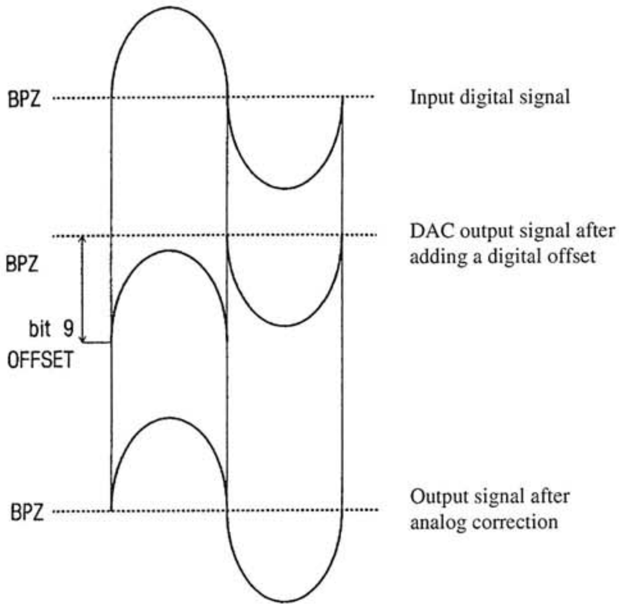


Figure 9. Circuit function of digital offset and analog EQ

2) Differential circuit

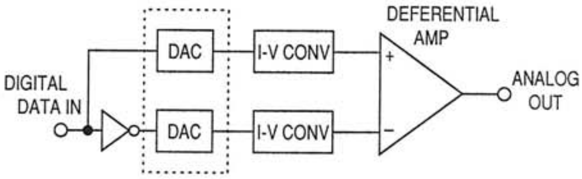
Figure 10 is a block diagram of the differential circuit. The differential circuit generates separate in-phase and antiphase signals for left (L) and right (R) in the digital signal stage. These are then converted by D/A converters to analog signals before they are summed by a differential amplifier.

The differential circuit:

① Improves the total S/N ratio by 3 dB increasing the output signal by 6 dB while raising the noise level by 3 dB.

② Cancels even-harmonic distortion.

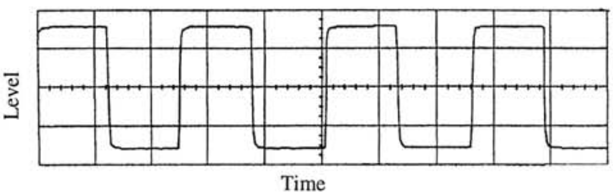
③ Cancels in-phase noise components generated by the digital circuitry.



3) LPF circuit

The LPF circuit removes unwanted signals, i.e., loopback signals, from high frequencies. A Butterworth filter is most commonly used as an LPF, but we used a Bessel filter for lower passband signal waveform distortion. Square wave responses of the two types of filters are shown in Figure 11 (a) and (b). A Butterworth filter has peaks at the rise and fall of each pulse, distorting the waveform.

(a) Bessel filter



(b) Butterworth filter

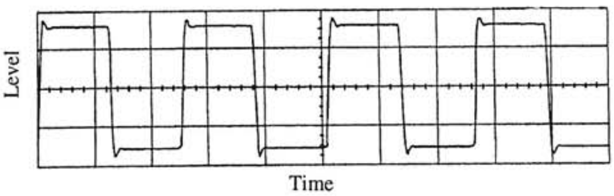


Figure 11. Square ware at LPF

3.3.4 D/A converter performance

The design techniques we used have improved various characteristics. These improvements are illustrated below using the example of measuring monotonicity and reproducing small signals.

1) Monotonicity

Monotonicity is the measurement of the analog signal output as the input increases in LSB steps from 0. This property is important to reproduction of small signals, and the presence or absence of zero-cross distortion. Figure 12 compares characteristics of our latest design with our previous model. The previous model was unable to reproduce input signals of +1 LSB and + 2 LSB. Our new D/A converter, however, reproduced these signals with no zero-cross distortion.

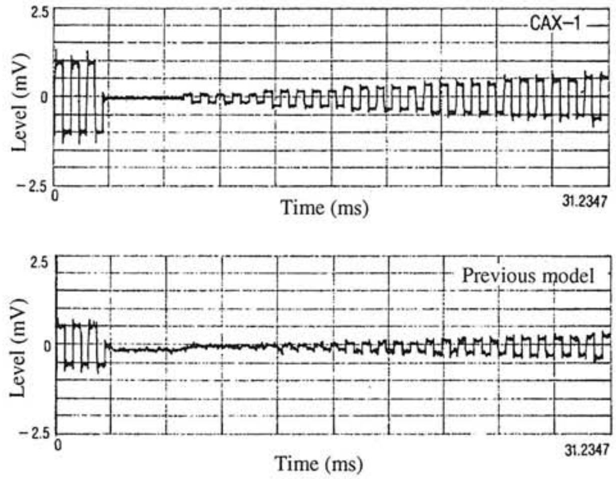


FIgure 12. Monotonicity characteristic curve

2) Low-level signal reproduction

We measured reproduction of a -60 dB signal at 1 kHz. Figure 13 compares performance with our previous model. The previous model had significant harmonic distortion at 1 kHz, while this D/A converter had a low harmonic distortion of 0.72%.

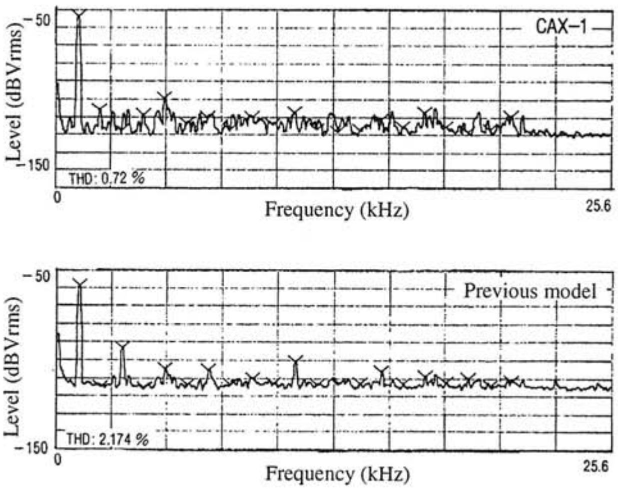


Figure 13. Low level signal (1kHz-60dB) play spectrum

3.3.5 Structural design

Before developing our latest series of audio equipment, we asked users for their opinions of existing products. Typical feedback from dealers was that car audio gear lacked a sense of luxury compared with home audio; front panels lacked stiffness because of use of resin materials; and controls were not pleasing to the touch. To improve the appearance and feeling of controls, our design included the following:

1) Aluminum front panel
   Car audio, both popular and luxury models, uses resin-molded front plates. To give products a sense of luxury, the molding is typically treated to give a surface texture with hairlining, special coating, or plating.
   Our sound monitor is fitted with an extrusion-molded aluminum front panel, 1.5 mm thick, in a resin subpanel. The panel surface is treated with alumite, in the same way as the radiating plate in the power amplifier, giving it a texture not possible with painting or coating.
   We are going to extend this to using the same aluminum extrusion dies in the manufacture of the control amplifier (CAX-1) and the tuner deck (TCX-1), and also when developing other Sound Monitor models.
   Figure 14 shows the structure of the front panel.
2) Improved feeling of controls
   Variable controls for car audio are extremely small with low resistor and slider torques. Resin controls which apply rotational torque directly are very light to the touch and lack a sense of luxury.
   Typical reasons for this are:
   ① Controls used for car audio equipment are smaller than those used for home audio.
   ② Resin controls are very light to the touch, or feel poor.
   ③ Variable resistors have a low rotational torque.
   ④ Controls often have multi-functions.
   Various measures have been taken to rectify these problems. For ①, the front panel controls are as large as possible. For ②, controls have a two-piece structure with brass inside resin, as shown in Figure 15. For ③, the torque of each control has been increased. For ④, we used a separate knob for volume control.

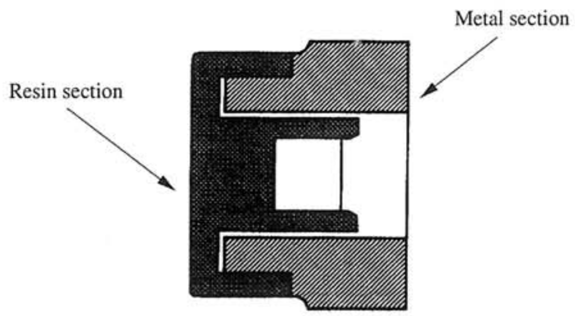


Figure 15. Structure of main volume

   The new main controls are pleasing to the touch with a sense of weight, and the brass surround for the resin section adds to the sense of luxury.
3) Front-panel RCA input terminal arrangement
   The control amplifier has RCA terminals on the front panel. Previous equipment incorporated minijacks; this is the first control amplifier with conspicuous terminals. The use of RCA terminals did, however, present some problems. The RCA terminals detract from the appearance when they are not in use, they are vulnerable to scratches and foreign matter, and circuits may be damaged by bodily static electricity. We overcame the first two problems by fitting protective caps for when the terminals are not in use. We also inserted a protective resistor in the internal circuit to protect against static electricity discharges of up to 20 kV.

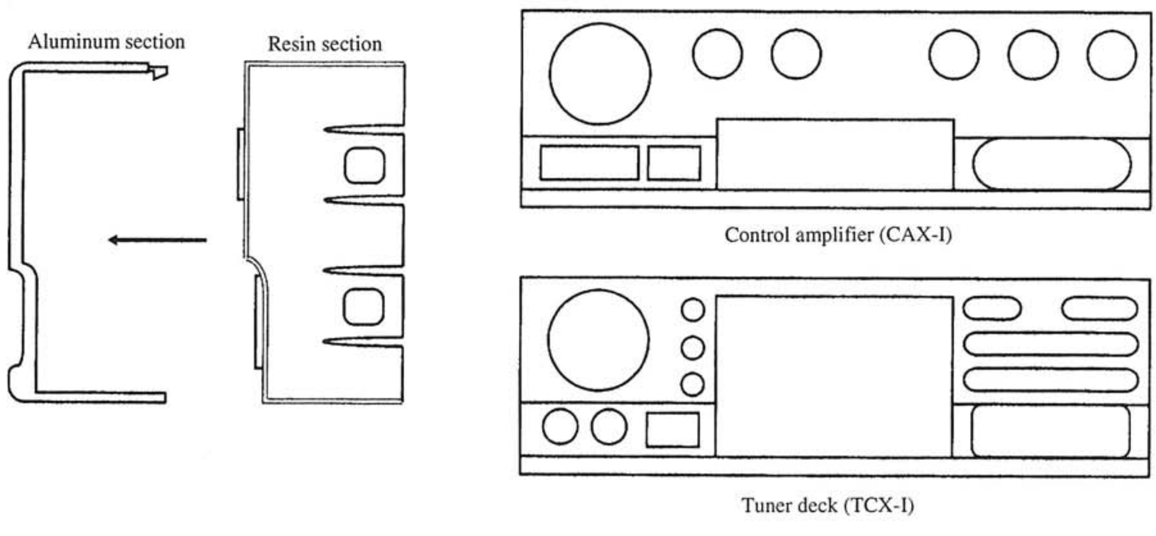


Figure 14. Structure of front panel

3.4 Tuner deck (TCX-1)

3.4.1 Outline

The tuner deck controls the CD auto-changer. The DK-76 mechanism in the deck section provides smooth operation. The tuner section has numerous features including auto-station memory and 12 AM and 12 PM presets. The high sound quality design supports digital sources such as CDs.

3.4.2 High sound quality design

In designing this model, we isolated possible factors of poor sound quality in the previous models. We then repeated sessions of sound quality evaluation to review circuit, compoment, and pattern connections.

1) Relays

Semiconductor analog switches used in previous models were prone to significant loss of sound quality because of complex signal routes. We used relays in this model to simplify signal rounting and for higher sound quality.

Table 1 compares the performance of semiconductor switches and relay switches.

Table 1. Performance of semiconductor and relay switches

	Semiconductor switches	Relay switches	Test conditions
Harmonic distortion	0.06 %	0.0006 %	Input: 1 kHz. 1 Vrms Filter: 1MF-A
S/N ratio	105 dB	105 dB	
Separation	105 dB	105 dB	

2) Low-noise power supply

Figure 16 is a block diagram of the power supply for the audio circuitry. This circuit regulates the current through zener diode D1 to increase stability. We achieved satisfactory regulation characteristics and reduced noise.

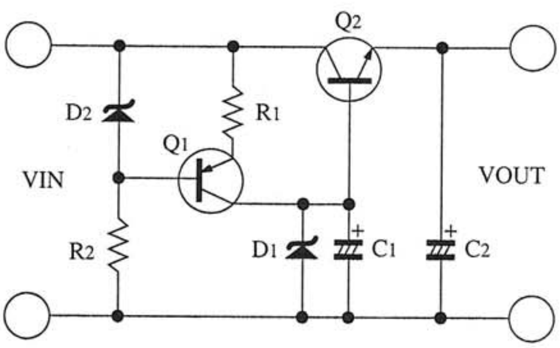


Figure 16. Power supply of audio circuit

3) Careful choice of components

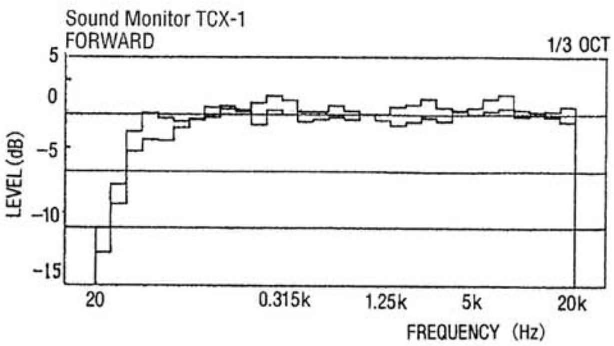
For high sound quality, we used audio-grade electrolytic capacitors, film capacitors, low-noise semiconductors, and discrete resistors rather than chip resistors, in the audio signal circuitry.

There are switches to bypass the VOL and BASS/TREBLE circuits so that the tuner deck can be used simply as a source deck when combined with a control amplifier.

3.4.3 Quality management

The components of the sound monitor system are individually tested to ensure their quality. We measure frequency response using a pink noise tape and data from floppy disks.

Figure 17 shows an example of results.



Fgure 17. Measurement data of frequency response

3.5 Power amplifier

We have worked on making the α 5000M, the top of our range of α models, better than the best our competitors can offer. The α 5000M has a high S/N ratio and low distortion and has impressed both audio critics and dealers.

The exceptional performance is a result of minimizing the interference between the switching noise from the power supply and the amplifier circuits. We used a separator amplifier unit, as used in high-end home audio, to separate the amplifier and power supply sections.

3.5.1 Amplifier section (PAX-3A)

The amplifier incorporates circuits and high-quality components not used in previous car audio equipment to faithfully reproduce digital music sources, from the faintest sound to the most powerful.

① The powerful integrated direct-coupled DC amplifier with push-pull operation, features an FET differential input, a cascade bootstrap, and a three-stage Darlington single-ended push-pull (SEPP) stage.

② A DC servo circuit eliminates capacitors from the negative feedback loop to improve low-frequency phase jitter characteristics.

③ A nonswitching bias circuit in the predriver stage gives the smooth sound of a class A amplifier with the efficiency of a class B amplifier.

④ The final output stage is a three-parallel push-pull connection capable of driving even low-impedance loudspeakers. Figure 18 is a block diagram of the power amplifier.

3.5.2 Power supply (PAX-3P)

A high-performance vehicle-mount amplifier with high output and low distortion needs high-voltage supply rails. The primary 12 V is boosted and high positive and negative voltages drive the amplifier circuit. The DC-DC converter in our power supply uses pulse width modulation (PWM). Power MOSFETs in a three-parallel arrangement switch currents of large values to ensure an efficient and stable power supply. The power supply unit has two predriver stages and two final output stages, as secondary outputs, to supply two power amplifiers. The power supply uses Cannon connectors for their resistance to external vibration and strength, and also because of their proven performance in professional equipment.

The exacting design considerations have given the Sound Monitor amplifier the highest level of performance and sound quality available for a power amplifier with a separate power supply for use in vehicles.

Table 2. Power amplifier performance comparisons

	Conventional models	PAK-1A	Test conditions
Frequency response	5 Hz to 100 kHz	5Hz to 100kHz	–1 dB POINT
Harmonic distortion	0.004%	0.0015%	1 kHz, 30 W output
S/N ratio (Figure 19)	105dB	113dB	Rated output (IHF-A filter)
Separation	80dB	80dB	

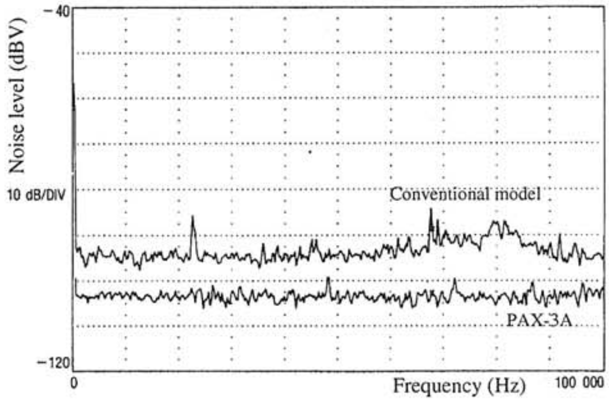


Figure 19. Noise spectrum

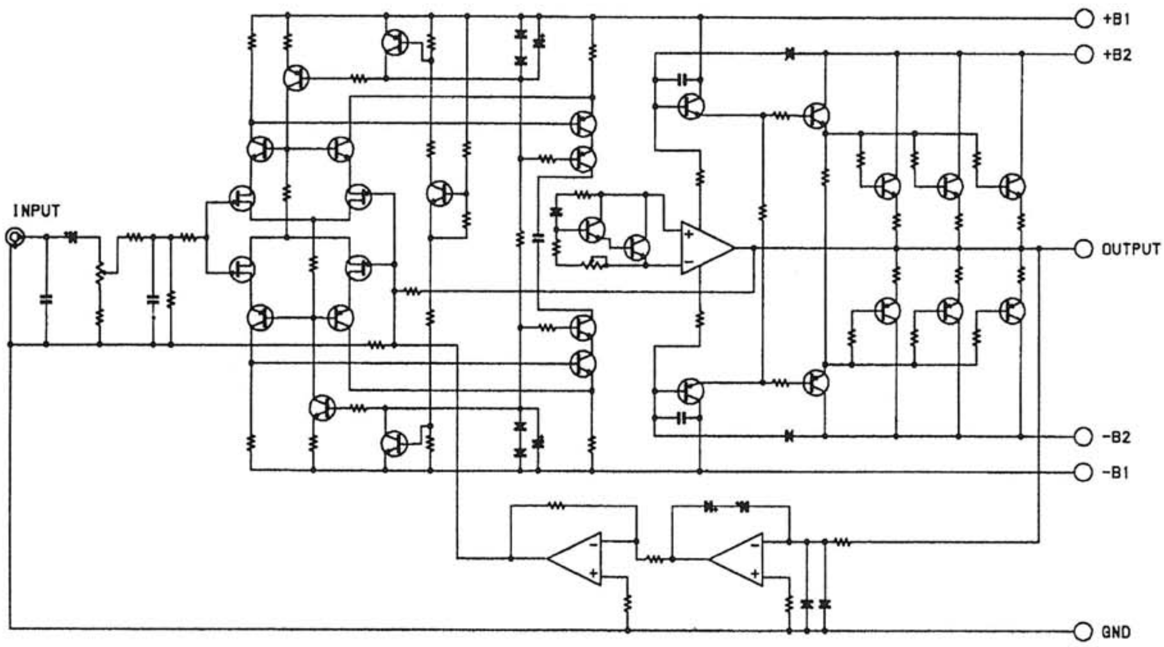


Figure 18. Circuit block diagram of power AMP

4. Conclusions

We have described the development aims of the Sound Monitor system and the principal technological concepts it is based on.

The Sound Monitor system has received high praise in a number of professional journals. The authors are working on yet better acoustic characteristics, and developing the car audio equipment that discerning drivers want. We will continue to give our customers the ultimate in music quality in their cars.

Sound Monitor

Sound field configuration measuring instrument (SFC-1)

Tuner deck (TCX-1)

Control amplifier (CAX-1)

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