

Development of Active Noise Control System for Engine Booming Noise

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ABSTRACT

The critical issue with active noise control technology is the balance between cost and performance, due to the need of expensive high-performance DSP processors for the many convolution operations required. As a result, there are few applications of this type of technology to reduce booming noise.

The purpose of this paper is to propose a new algorithm significantly reducing the amount of arithmetic operations by focusing on the fact that booming noise is a narrow band noise. This algorithm enables us to employ an inexpensive controller even though adaptive signal processing is utilized. The integration of this system with an audio system further improves the balance of cost and performance.

The system developed in this study was applied to a new vehicle model in combination with an active control engine mount system to reduce booming noise generated in three-cylinder operation by a new V6 engine provided with cylinder on demand. The result is the achievement of excellent fuel economy and NV performance.

1. Introduction

The demand for fuel economy of automobiles is increasing, and as a consequence, new technology for achieving better fuel economy for the power plant as well as reducing the weight of the vehicle is being actively studied. The demand for improving the noise and vibration performance so as to improve the marketability of the vehicle has also become strident. However, the conventional passive measures for noise and vibration were found to increase the weight of the vehicle in most cases, and this was in conflict with the demand for fuel economy. Active Noise Control (hereafter ANC) technology achieves a high-level balance of these demands⁽¹⁾. This technology attains noise reduction to a level that cannot be achieved by passive measures. A speaker

is used to create cancellation sound that interferes with the noise, thus making it possible to build a system that keeps the increase in weight to a bare minimum.

In a previous study, the authors implemented a system that reduces fixed frequency, low frequency road noise by using analog circuits, which are cost effective, integrated it with the audio system and used it in mass production vehicles⁽²⁾. The target of this study is engine booming noise, the frequency of which varies with changes in engine rpm. The conventional booming noise control technology⁽³⁾ required a large number of signal processes as it used convolution operations, and also required high-performance and expensive signal processors like Digital Signal Processors (DSP). Thus, cost related to performance became a challenge.

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The objective of this study is to construct a control algorithm that will significantly reduce the number of arithmetic processes by focusing on the fact that the engine booming noise is a narrow band noise. The objective is also to improve the balance between cost and performance by promoting integration of the proposed system with the audio system.

The audio integrated ANC system developed based on this study, was used as a standard feature in new vehicles fitted with the V6 cylinder-on-demand engine to improve the engine booming noise that occurs when the vehicle is running in three-cylinder operation thereby enabling a high-level balance between the fuel consumption and NV performance.

2. Overview

2.1. Characteristics of Engine Booming Noise

Figure 1 shows the noise in the cabin during gradual acceleration. The horizontal axis in the figure is time and the vertical axis is frequency. The peak at the center of the figure is the booming noise. It is confirmed that the frequency of booming noise during acceleration increases synchronizing with the engine rpm and that the booming noise is restricted to a narrow frequency band. As shown in Fig. 2, being a narrow band noise means that the noise waveform is very close to that of a sine wave, and the waveform of the cancellation sound used for noise reduction is also close to that of the sine wave.

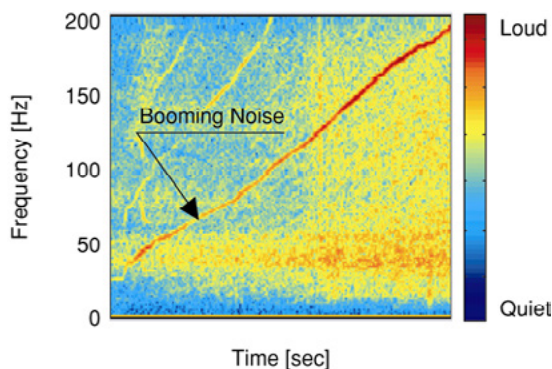


Fig.1 Contour map of noise inside cabin

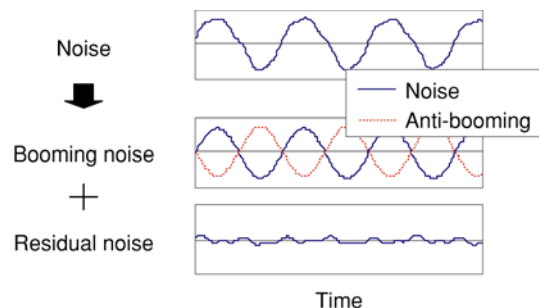


Fig.2 Time domain signal of noise inside cabin

2.2. System Overview

Figure 3 shows the system configuration. As shown in the figure, the system consists of the ANC Control Unit (Electronic Control Unit; hereafter ANC-ECU), located

within the instrument panel, a microphone unit fitted in the rear tray, an audio head unit fitted with the input terminal of the cancellation signal, and speakers that are shared with the audio system. This system has two microphones since the ANC-ECU has a built-in microphone. The input signals to the ANC-ECU consist not only of the power supply, but also the rpm pulse signal (NeP) to detect the engine rpm, and the noise signal from the rear microphone unit around the rear seat. The outputs are signals for driving the speakers located in both the front doors, and the speakers fitted in the rear tray. This system employs an adaptive feed forward control^{(4), (5)} that operates such that the squared value of the noise signal from the two microphones is minimized. That is, it calculates the control frequency based on the engine rpm signal, creates a synchronized reference signal internally, and uses the adaptive digital filter to create the cancellation sound.

The integration of this system with the audio system was designed such that the main amplifier and the speakers were shared. Since this system is to be used in the new V6 engine with the cylinder-on-demand system, the function to communicate with the audio system for operational checks and the input circuit of Stop Cylinder Pulse (hereafter SCP) were added.

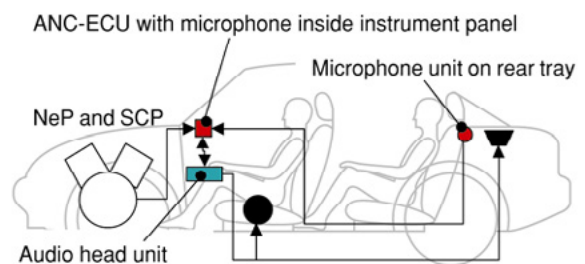
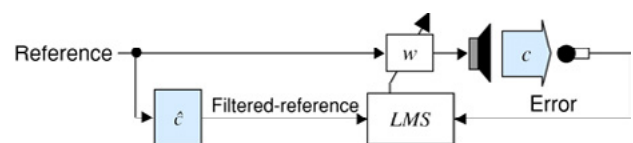


Fig.3 On-board system configuration

3. Control Technology

3.1. Issues in the Conventional Control Algorithm

Figure 4 shows the block diagram for controlling booming noise using the Filtered-Reference LMS (hereafter F-R LMS)^{(4), (5)}, which has been widely used in past work⁽³⁾. This control algorithm generates sine waves synchronized with the engine rpm as reference signals, converts the phase and amplitudes of the signal using the adaptive FIR (Finite Impulse Response) filter w , and then outputs the signals. The adaptive FIR filter w is updated recursively so that the



w : Adaptive FIR filter

c : Characteristic of acoustic cavity

$\hat{c}$: FIR filter estimating characteristic of acoustic cavity

Fig.4 Block diagram of filtered-reference LMS

squared value of e becomes minimal. It uses the error signal e detected by the microphone provided in the cabin and the FIR filter $\hat{c}$ modeled by the acoustic characteristics in the cabin, which is obtained as a parameter beforehand. Given below are the issues with this algorithm.

(1) Expensive signal processors are necessary

The control method uses many convolution operations (FIR filter calculation) equivalent to multiplication of vectors, which increases the number of arithmetic processes and necessitates a high performance processor. For instance, if the acoustic characteristic $\hat{c}$ is modeled by 128 filter length, multiplication needs to be performed 128 times and addition needs to be performed 127 times in each sample interval. Moreover, the convolution operations of the adaptive filter are also necessary. Furthermore, if a system with multiple speakers and microphones is to be built, then the number of arithmetic processes in the control algorithm would become abnormally high. In addition, to track the engine rpm at high speeds, the sampling frequency must be as high as possible, which necessitates high performance, high cost signal processors such as DSP.

(2) Issues related to modeling precision in the low frequency region

Since the acoustic characteristic $\hat{c}$ is modeled using the FIR filter, if the filter length is short for the actual impulse response, the precision of modeling in the low frequency region becomes poor. For instance, in a system that operates at the sampling frequency of 3 kHz, if the acoustic characteristics are modeled at 128 filter length, the frequency resolution Δf of the model becomes 23.4 Hz, the precision of the sound model for low frequencies such as 40-50 Hz decreases, and the control effect diminishes. Consequently, it is necessary to reduce the sampling frequency or to increase the number of filter length for modeling when controlling the low frequency noise. However, the problem with the former method is that it reduces the ability to track the engine rpm, and the problem with the latter is that the calculation load increases.

3.2. Reduction in the Number of Arithmetic Processes

3.2.1. Fundamental logic for reduction in the number of arithmetic processes

Figure 5 shows the booming noise shown on the complex plane. As mentioned in section 2.2, since the booming noise is a sine wave event, the cancellation sound of the booming noise is expressed as two values, A and B , on the complex plane. This is equivalent to saying that if the RX signal and the RY signal, which are reference signals perpendicular to each other and corresponding to the real axis and the imaginary axis respectively, are used, the cancellation sound is generated by taking the amplitude of the RX signal as A and the amplitude of the RY signal as B . Considering that the RY signal progresses at a phase of 90° with respect to the RX signal, the RX signal is shown as the real axis and the RY signal as the imaginary axis in the figure. This method allows the adaptive notch filters A and B that correspond to the amplitude of the perpendicular signals to track the changes in the engine rpm by using the digital signal processing. This method is known as SAN (Single-Frequency Adaptive Notch) filter^{(5), (6)}.

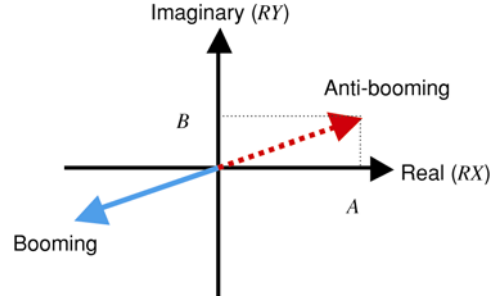


Fig.5 Polar coordinates of booming noise

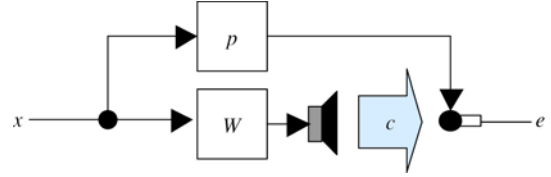


Fig.6 Block diagram of control system

3.2.2. Deployment in the control algorithm

The signal processing method to track changes in the engine rpm is given here. Figure 6 shows the block diagram of the adaptive signal process. Here, the reference signal is taken as x , adaptive filter as w , the acoustic characteristic as c , and its model as $\hat{c}$. Moreover, if p is taken as the unknown part, then the booming noise is assumed to be reference signal which reaches the microphone following the unknown part p . If it is currently presumed that there are no modeling errors, then $c = \hat{c}$ and the microphone signal e is expressed by the equation shown below.

$$e = p \cdot x + w \cdot \hat{c} \cdot x \quad (1)$$

The gradient Δ is found as shown in Fig. 7 to update the adaptive filter w so that the squared value of the microphone signal becomes a minimum. Thus, the gradient Δ is given by the equation below.

$$\Delta = \frac{\partial(e^2)}{\partial w} = 2 \cdot e \cdot \hat{c} \cdot x \quad (2)$$

Therefore, as shown in Fig. 7, the control system to track the changes in the engine rpm is built by updating the adaptive filter w recursively so as to attain the optimum value by the equation below.

$$w_{n+1} = w_n - \mu \cdot \Delta \quad (3)$$

Here, μ expresses the step size parameter.

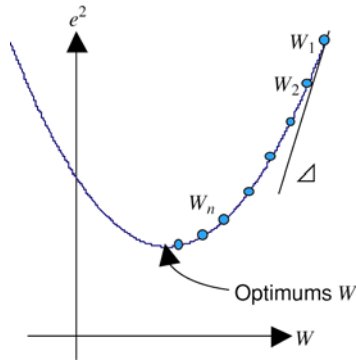


Fig.7 Optimization of evaluation function by LMS method

Next, equation (2) shows the method to reduce the number of convolution processes of the $\hat{c} \cdot x$ calculation as much as possible. For this, the frequencies of the acoustic characteristic $\hat{c}$ are analyzed and saved as a table for the control frequencies as shown in Table 1. The values corresponding to the frequencies to be controlled are read from the table and used in the calculations. The acoustic characteristic $\hat{c}$ in the frequency f is expressed using the complex expression (take j as an imaginary unit) shown below.

$$\hat{C}(f) = C_R(f) + j \cdot C_I(f) \quad (4)$$

Here, the method for realizing $\hat{c} \cdot x$ is shown, based on the acoustic characteristics in equation (4). It shows the calculation method using the RX signal in x for $\hat{C} \cdot x$

$$\begin{aligned} \hat{C}(f) \cdot RX &= C_R(f) \cdot RX + j \cdot C_I(f) \cdot RX \\ &= C_R(f) \cdot RX + C_I(f) \cdot RY \end{aligned} \quad (5)$$

Equation (5) uses the fact that RY is a signal that progresses at a phase of 90° with respect to RX , $j \cdot RX = RY$. In the same way, the calculation method when the RY signal is used in x , is given below.

$$\begin{aligned} \hat{C}(f) \cdot RY &= C_R(f) \cdot RY + j \cdot C_I(f) \cdot RY \\ &= C_R(f) \cdot RY - C_I(f) \cdot RX \end{aligned} \quad (6)$$

Table 1 Control parameter of acoustic characteristic

$f[\text{Hz}]$	C_R	C_I
30	30	-20
40	15	-10
41	13	0
42	40	25
43	62	47
.	.	.
.	.	.
100	-12	-49

Equation (6) uses $j \cdot RY = -RX$. Figure 8 shows the block diagram for engine booming noise control using equations (5) and (6). A feature of this control method is that the Frequency Domain Filtered-Reference (hereafter Freq. F-R SAN) does not make use of any convolution operation, as compared to the use of many convolution operations equivalent to the multiplication of vectors by the F-R LMS method, which is a common control method. That is, the main feature is that all the calculations involve multiplication and addition of singular values. Shown below are the updated equations of the adaptive notch filters A and B

$$\begin{aligned} A_{n+1} &= A_n - \mu \cdot e \cdot \{C_R(f) \cdot RX + C_I(f) \cdot RY\} \\ B_{n+1} &= B_n - \mu \cdot e \cdot \{C_R(f) \cdot RY + C_I(f) \cdot RX\} \end{aligned} \quad (7)$$

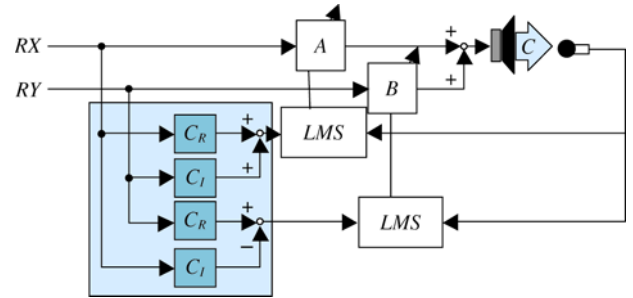


Fig.8 Block diagram of Freq. F-R SAN algorithm

3.3. Effect of Reduction in Arithmetic Processes

The number of arithmetic processes for one sample of Freq. F-R SAN and F-R LMS algorithms were compared. Table 2 shows a comparison of the number of arithmetic processes carried out by the Freq. F-R SAN algorithm and F-R LMS algorithm in one sampling period when the acoustic characteristics were modeled by 128 filter length and adaptive filter modeled by 64 filter length. This table represents the case of one output and one control point. The number of arithmetic processes of filter for the sound model, adaptive filter, the number of updated arithmetic processes of the adaptive filter and their sums were also compared. It is clear from this table that the booming noise is controlled by the Freq. F-R SAN algorithm with approximately $1/25^{\text{th}}$ the number of arithmetic processes of the F-R LMS algorithm. The difference in the number of arithmetic processes becomes larger when the multi-output multi-point control system is constructed in which the number of microphones or output increases.

Table 2 Comparison of arithmetic operations carried out by Freq. F-R SAN algorithm and F-R LMS algorithm in one sampling period

	Freq. F-R SAN	F-R LMS
Acoustic characteristic convolution	6	255
Adaptive filter convolution	3	127
Adaptive signal processing	12	129
SUM	21	511

In addition to the reduction of the number of arithmetic processes, if the acoustic characteristics are maintained as a table at a resolution of 0.5 or 1 Hz, Freq. F-R SAN enables precise modeling restricted to the frequency range, where the control is implemented. Hence, this can resolve the issue of reduction in sampling frequency or increase in the volume of arithmetic processes in F-R LMS.

3.4 Stability Compensation for Control System

Parameters for stable operation of the Freq. F-R SAN system were introduced. The following two cases of unstable operations of the control system may be considered.

- (1) When the noise components other than control frequency components included in the microphone signals are large.
- (2) When frequency difference arises between that of the booming noise included in the microphone signal and the control frequency.

For both (1) and (2), the operation of the control system may become unstable. This is because the adaptive filter operates to reduce the frequencies other than those it should control, in regard to control frequencies predicted from the engine rpm, when the signals of control frequency components included in the microphone signal are relatively smaller compared to other frequency components. To inhibit this, mixing of signals of the control frequency component according to the size of adaptive filter in the microphone signal may be considered. Figure 9 shows the block diagram for this process.

A large α is effective in improving the stability. However, increasing α inhibits the control output and reduces the control effect.

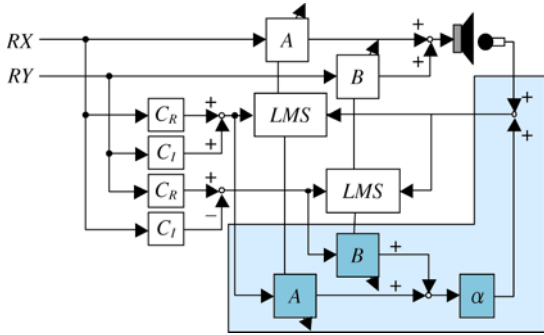
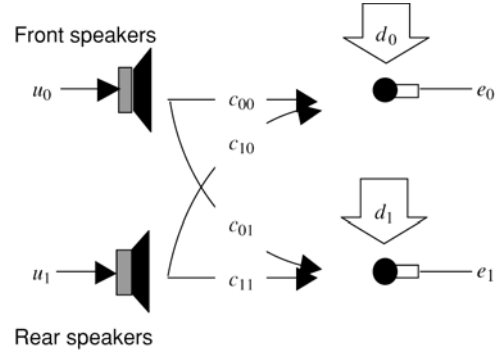


Fig.9 Block diagram of frequency domain filtered-reference SAN with stable parameter

3.5. Simplification of System Configuration

The number of microphones is reduced to simplify the system configuration. Reduction in the number of microphone units has a large effect not only on unit cost reduction, but also on overall cost reduction since items such as harness and garnish become unnecessary. Therefore, we considered a technique to control the front and rear seat booming noise by using a single microphone unit. A single microphone system was developed from the two microphone systems shown in Fig. 10. This single microphone system generates the output signal for the rear seat by compensating the front seat output signal using the rear seat compensation filter F as seen in Fig. 11.



d_0 : Booming noise in front passenger seat

d_1 : Booming noise in rear passenger seat

Fig.10 Two microphone system

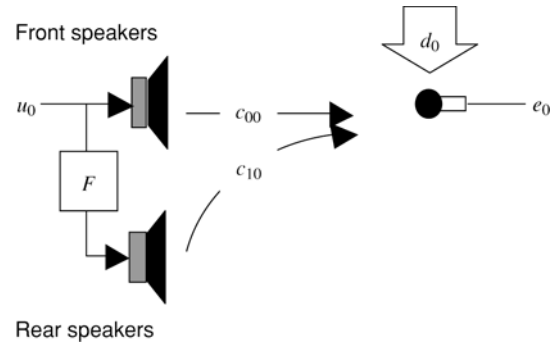


Fig.11 One microphone system

3.5.1. Identification method for rear seat compensation filter F

- (1) Identification method that uses the booming noise distribution at the front and rear seats.

Using this method, the booming noise at the front and rear seats is reduced. The acoustic characteristics up to each of the front and rear microphones from each of the front and rear speakers are distinguished by subscripts from 00 to 11 as seen in Fig 10. The microphone signals e_0, e_1 are expressed in equations (8) and (9). The two equations are assumed to be zero for operations under ideal conditions. Further, when the front and rear seat booming noise distribution is fixed, this condition is expressed by equation (10).

$$e_0 = c_{00} \cdot u_0 + c_{10} \cdot u_1 + d_0 = 0 \quad (8)$$

$$e_1 = c_{01} \cdot u_0 + c_{11} \cdot u_1 + d_1 = 0 \quad (9)$$

$$q = \frac{d_1}{d_0} \quad (10)$$

From equations (8), (9), and (10) the front and rear output ratio F is obtained. $c_{00} - c_{11}$ and q here, are taken as $C_{00} - C_{11}$ and Q in the frequency range. From the above, the equation below is obtained.

$$F(f) = \frac{U_1(f)}{U_0(f)} = \frac{C_{01}(f) - Q(f) \cdot C_{00}(f)}{Q(f) \cdot C_{10}(f) - C_{11}(f)} \quad (11)$$

- (2) Identification method that uses acoustic characteristics of the cabin only.

By applying this method, the amplified sound generated at the rear seat is controlled when reducing the booming noise at the front seat. The output signal for control of the front seat arrives at the rear seat according to the transmission characteristic c_{01} . The rear seat output signal is used to cancel the amplified sound generated at the rear seat. The rear seat compensation filter F used here is given in the equation below.

$$F(f) = -\frac{C_{01}(f)}{C_{11}(f)} \quad (12)$$

3.5.2. Use of rear seat compensation filter F

Since the identified rear seat compensation filter F in the frequency range is a complex number, it is expressed by the equation below together with equations (11) and (12).

$$F(f) = F_R(f) + j \cdot F_I(f) \quad (13)$$

The rear seat compensation filter F is retained as a table for frequency f similar to the table for acoustic characteristics $\hat{C}$ shown in Table 1, and it is used in calculations after reading the values corresponding to the control frequencies. Figure 12 shows the corresponding control block diagram. This control system requires an extremely low number of arithmetic processes for the rear seat output similar to acoustic characteristics $\hat{C}$. It enables a single microphone system with a cost effective microprocessor to be realized, and is therefore, an effective method. While using this method, it is necessary to use the equation below in acoustic characteristics $\hat{C}$.

$$\hat{C}(f) = \hat{C}_{00}(f) + F(f) \cdot \hat{C}_{10}(f) \quad (14)$$

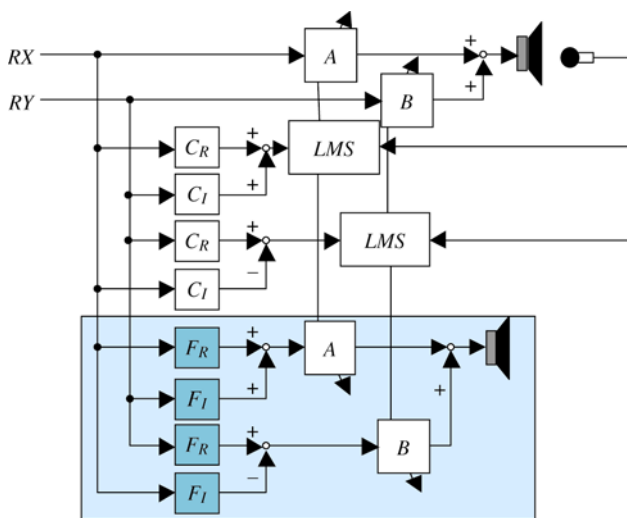


Fig.12 Block diagram of frequency domain filtered-reference SAN with rear area compensation

The front and rear seat booming noise reduction system shown in (1) requires prior studies because of the premise that the booming noise ratio q of the front and rear seat, including the phase in equation (10), is constant during acceleration / deceleration or when the difference of body weight in mass production vehicles is confirmed. The system for inhibiting increase in noise at the rear seat of (2) uses a parameter that can be identified only from the acoustic characteristics of the cabin. Therefore, when compared to (1), its operation is stable, but it cannot effectively reduce the booming noise at the rear seat. Consequently, it is important to select and use the appropriate method according to the vehicle characteristics.

4. Integration with the Audio System

To integrate this system with the audio system, a method was applied in which the control output signal was incorporated in the audio unit, as shown in Fig. 13, the power amplifier in the audio head unit was shared, and the speakers were driven. Consequently, the following two specifications were added to the audio head unit:

- (1) A mixer circuit was fitted just in front of the power amplifier in the audio head unit
- (2) The power amplifier was set to operate even when the user has turned off the main switch

In addition to the above two additional specifications, a function for communicating between the ANC-ECU and audio head unit was added as a checking function for the assembly process at the plant or support services for malfunction occurrences.

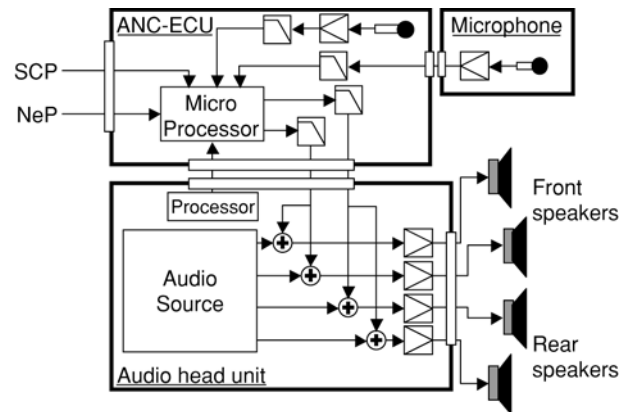


Fig.13 System configuration

5. Application Results

This system was used in vehicles fitted with V6 cylinder-on-demand-engines. During three-cylinder operation, besides the engine vibration, the intake sound and exhaust sound also deteriorate, and the booming sound in the cabin deteriorates due to torque converter lock-up. This system was set so that it could demonstrate maximum noise reduction by incorporating the SCP signal that shows three-cylinder operation for good efficiency in the reduction of booming noise. Figures 14 and 15 show the reduction effect attained by this system. It is confirmed from the figure

that the reduction effect is approximately 10 dB for 1.5th order component of the engine booming noise that deteriorates due to torque converter lock up in the vicinity of 1 800 rpm. Further, this system uses a Fujitsu semiconductor device, which is a 16-bit micro controller F2MC-16LX/MB90562/A.

This system enables the user to drive without being aware of three-cylinder operation.

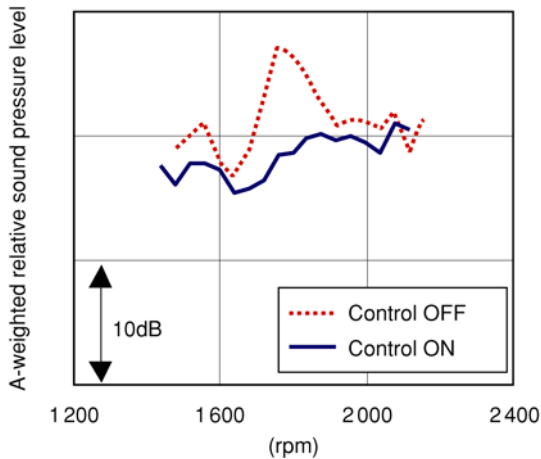


Fig.14 1.5th order sound pressure level at front seat

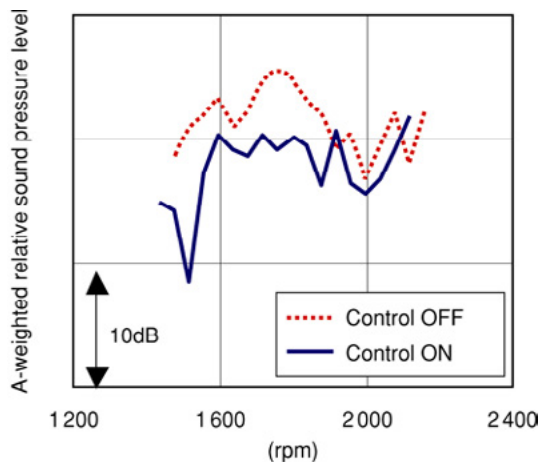


Fig.15 1.5th order sound pressure level at rear seat

6. Conclusion

The Active Noise Control System was developed aiming at reducing the engine booming noise, integrated with the audio system, and was mass produced. This system enabled a balance to be obtained between fuel economy and NV performance by using measures against the booming noise during three-cylinder operation for a V6 cylinder-on-demand system. Henceforth, this active control technology for noise and vibration is anticipated to be given a prominent position because of the good balance between weight reduction/ fuel economy and noise vibration performance.

Acknowledgments

We are deeply grateful to Matsushita Electronic Components Co., Ltd. for their wholehearted cooperation in this development.

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