Study on the Howling Canceler in Active Noise Control*

(Two Microphone Delay System)

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An active noise control technique for prevention of howling is studied in this paper. A 2-Microphone system is often employed in a long-sized duct, where the propagation wave is the main component. However, in a short-sized duct, the reflected waves must be considered. In this case, the authors introduced the acoustic condition that the component of the reflected wave should be absorbed at the end opposite the opening end. The authors called this system the 2-Microphone delay system. The acoustic property of the duct for the 2-Microphone delay system was examined. The effect of this active noise control system with a howling canceler was compared with a 1-Microphone system. Their characteristics were examined by comparing theoretical results with the experimental results. The features of the individual systems have been determined. It was found that the digital filter in the 2-Microphone delay system implements the control filter characteristics except the frequencies determined by the distance between two microphones. On the other hand, the implementation of the 1-Microphone system was poor.

Key Words: Sound, Noise, Control, Active Control, Howling, 2-Microphone System, Canceler System, Digital Filter

1. Introduction

The active noise control technique has been developed, achieving improvement of the digital signal processing method and an excellent effect on low-frequency-range noise⁽¹⁾. When the signal radiated from a noise source is detected by a microphone, system instability, called howling, will occur according to the gain of the digital filter coefficient for control. The basic concept for preventing this howling phenomenon is to eliminate the signal from the secondary source included in the detection microphone. One of the ways to solve this problem is a 2-Microphone system. The conventional 2-Microphone system utilizes a time delay between two microphones⁽²⁾. As the one-way forward propagation wave is the main component in a long duct, this conventional system eliminates the howling signal effectively. However, in a short duct, there are many waves reflected on both

sides. So, the system must be considered under the condition of many reflected waves. The digital filter implementation is related to the causality of the digital filter. In Refs. (3) and (4), the causality problem of the active control is described. However, few comments relating the digital control filter throughput delay and the duct dimensions are described. In this paper, the acoustic condition and the property were determined for applying the conventional 2-Microphone system. From this result, the control filter characteristic was investigated to confirm the digital filter implementation.

2. Basic Concept of a 2-Microphone System

In a short-sized duct, there are many reflected waves, as shown in Fig. 1. $(P_{PA})_{pn}$ ($n=1, 2, \dots$) and $(P_{PA})_{rn}$ ($n=1, 2, \dots$) show the forward propagated waves generated from the secondary source. $(P_{rA})_{pn}$ and $(P_{rA})_{rn}$ show the backward propagated waves generated from the secondary source. $(P_{rA})_1$ indicates a direct acoustic wave from the secondary source. The summation of these reflected waves is observed as a sound pressure signal at the measuring location. For example, the summation of $(P_{PA})_{pn}$ and $(P_{PA})_{rn}$ is

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observed as P_{PA} , which indicates the forward propagated wave. Figure 2 shows a typical 2-Microphone system. P_{PA} and P_{TA} represent the overall forward- and backward- propagated waves generated from the secondary source, respectively. P_{PS} and P_{TS} show those generated from the primary source. P_M detected by a 2-microphone system consists of P_{MS} , by the noise source contribution signal and P_{MA} , by the secondary source signal as follows:

$$P_M = P_{MS} + P_{MA} \quad (1)$$

P_{MA} is represented in terms of P_{PA} and P_{TA} as follows:

$$P_{MA} = (1 + G_1 \cdot e^{-jk\Delta l}) \cdot e^{jkl} \cdot P_{PA} + (1 + G_1 \cdot e^{jk\Delta l}) \cdot e^{-jkl} \cdot P_{TA} \quad (2)$$

In a howling canceler system, the signal P_{MA} should be zero. In this Eq. (2), as previously discussed, P_{PA} is a component of the total summation signal of the reflected waves from the end opposite to the opening end. When the absorption material is packed in the space of the duct as shown in Fig. 3, the component P_{PA} is zero. In this case, the acoustic signal path is shown in Fig. 3, i.e., $(P_{TA})_1$ and $(P_{TA})'_1$. The authors called this duct an acoustic absorption duct. This acoustic condition is possible because the opening end remains open. So, under the condition that P_{PA} is zero, the condition of G_1 to render Eq. (2) equal to zero is

$$G_1 = -e^{-jk\Delta l} \quad (3)$$

Fig. 1 Reflection wave components

Fig. 2 Diagram for active noise control system using 2-Microphone system

The filter characteristic means a conventional 2-Microphone method. However, the acoustic condition was introduced to apply the conventional 2-Microphone system. The authors called this 2-microphone system a 2-Microphone delay system. The 2-Microphone delay system is shown in Fig. 4. This system is almost the same as the conventional 2-Microphone system except for an absorption material indicated by the shaded area. The output signal of each microphone in the 2-Microphone system is fed into the differential amplifier, which has a phase delay corresponding to the distance between two microphones. The output signal of the differential amplifier is fed into the active digital control filter G . The control signal of G is radiated from the secondary source.

Next, the acoustic characteristic of the duct shown in Fig. 4 was introduced theoretically. In Fig. 4, in the acoustic absorption duct, the reflection wave is generated at the opening end. α is a reflection coefficient at the opening end. As an active noise control system is employed in the low-frequency region, the reflection coefficient α is almost equal to -1 . Pa' is an incident acoustic signal. Pra is a reflection signal which is equal to $\alpha Pa'$. When the

Fig. 3 Reflection wave components with sound absorption material installed

Fig. 4 Duct system for a 2-Microphone delay system

speaker S radiates the acoustic signal, the acoustic signal characteristic P_x at the microphone location x is represented as follows:

$$P_x = D \cdot e^{-j(kx - \phi)} \quad (4)$$

Fig. 5 Amplitude characteristic of duct with absorption material ($\alpha < 0$)

Fig. 6 Experimental result of the duct with absorption material

where

$$D = \sqrt{1 + \alpha^2 + 2\alpha \cos 2ka} \quad (5)$$

and

$$\tan \phi = \frac{-\alpha \sin 2ka}{1 + \alpha \cos 2ka} \quad (6)$$

The amplitude characteristic of D is shown in Fig. 5. However, in the region where the reflection coefficient α is assumed to be a complex variable, the amplitude characteristic must be compensated according to the open end correction. The frequencies at the maximum and minimum values are determined by the distance ' a ' from the opening end to the location of the secondary source. The acoustic characteristic shows that the sound wave is a propagation wave with a frequency-dependent amplitude. The experimental result is shown in Fig. 6. The frequency at the minimum value indicated by the arrow is almost the same, even though the distance between the speaker and the microphone varies. This experimental result is in good agreement with the theoretical result introduced by Eq.(4). The reason that the frequency of the experimental result is lower than that of the theoretical result is due to the open end correction.

The effect of the absorption material was confirmed by measuring the impulse response function of the duct. Figure 7 shows the impulse response function of the acoustic absorption duct at the microphone M. The left-end area is assumed to be packed with the absorption material. (glass wool plus felt) Therefore no reflection wave exists. The absorption coefficient was not measured in this experiment. However, the authors believe the coefficient of about 0.1 is sufficient to establish the acoustic absorption duct. In Fig. 7, two main reflection waves are observed. These waves correspond to the acoustical path

Fig. 7 Impulse response function from loud speaker (S) to microphone (M) with sound absorption material

shown in Fig. 3.

The effect of eliminating the signal from the secondary source in the 2-Microphone delay system is shown in Fig. 8. The filter coefficients G_1 were represented using Eq.(3), which corresponds to the time delay of the distance between two microphones. Figure 8(a) shows the experimental result of the signal acquired by the single microphone M1 and the 2-microphone system with an absorption material. Figure 8(b) shows the case without an absorption material. When using an absorption material, the signal from the secondary source was cancelled, as in Fig.8(a). On the contrary, no cancel effect was obtained without an absorption material. So, the effect of the absorption material in the 2-microphone delay system was confirmed.

3. Experimental Apparatus for Confirmation of a Digital Filter Characteristic in the Control System

The duct system for an experiment is shown in Fig. 9. The signal processing system utilized a digital signal processor. The control filter function was calcu-

Fig. 8 Output signal from 2-Microphone system

Fig. 9 Duct system

lated by the operation from the measured frequency response function. The impulse response function to be set in the digital filter controller was determined by the inverse FFT of this control filter function. The impulse response coefficients were transferred into the digital signal processing system. Microphone M1 was utilized as a detection microphone in the 1-Microphone system.

In the 2-Microphone system, the delayed signal corresponding to the distance between two microphones was calculated by the digital signal processing system. The sampling frequency was 3.2 kHz, and the absorption material was packed in the area indicated by this figure.

4. Problem with a 1-Microphone Method

The typical 1-microphone system that was granted by Lueg⁽⁵⁾ is shown in Fig. 10. The detection microphone is M. Noise is radiated from the speaker S. G is a filter to control. A is a secondary source. Frequency response function from the noise S to the detection microphone M is indicated as G_{SM} , and so on.

The main problem is system instability called howling, discussed in the previous section. Furthermore, it is difficult to implement the filter coefficient because a non-causality system is constructed. The filter coefficient G to reduce the noise level at the evaluation point O is represented as follows:

$$G = \frac{-1}{\frac{G_{SM}}{G_{SO}} \cdot G_{AO} - G_{AM}} \quad (7)$$

When the duct is the acoustic absorption duct, the propagation wave is the main component as Eq.(4). The respective transfer functions are represented as follows:

$$\begin{aligned} \frac{G_{SM}}{G_{SO}} &= e^{-jk(l_1-l_2)} \\ G_{AO} &= e^{-jk l_3} \\ G_{AM} &= e^{-jk(l_2-l_3-l_1)} \end{aligned} \quad (8)$$

The amplitude coefficient is one unit in magnitude. The final digital filter coefficient must be divided by

Fig. 10 One-Microphone system

the digital filter throughput delay. This delay is composed of the antialiasing filter delay, calculation delay, smoothing filter delay for the acoustic output and the secondary source delay. By using θ , The throughput delay G_{th} is represented as follows :

$$G_{th} = e^{-j\theta\omega} \quad (9)$$

In Eqs. (7) - (9), the filter coefficient G is represented as follows :

$$G = \frac{e^{j(\frac{\pi}{2} + \theta\omega)}}{2 \sin k(l_2 - l_3 - l_1)} \quad (10)$$

where k is the wave number. Therefore, the model filter characteristics are as shown in Fig. 11. The frequency points f_{un} at maximum value are determined as follows :

$$f_{un} = \frac{c}{2 \cdot (l_2 - l_3 - l_1)} \cdot n \quad (11)$$

where c is sound velocity. As the figure indicates, the filter coefficient is basically uncausal, because the

Fig. 11 Digital filter characteristic of 1-Microphone system

Fig. 12 Experimental result of the digital filter implementation in 1-Microphone system

phase has a lead. The implementation result of the digital filter of the duct is shown in Fig. 12. The bold line shows the filter to be implemented. The narrow line shows the digital filter characteristic. The frequency points at maximum value were in good agreement with Eq. (11). The implementation characteristics are poor when the phase has a lead.

The noise reduction effect using the filter coefficient indicated by Fig. 12 is shown in Fig. 13. The duct system has an absorption material, as introduced in the previous section. The noise reduction effect is observed only at maximum amplitude points of the control filter gain where the digital filter implements the frequency response well, as the phase has a lag. Howling was observed at a frequency of 270 Hz.

5. The Effect of a 2-Microphone System

The 2-Microphone delay system is shown in Fig. 14. D is the time delay corresponding to the distance between two microphones. The frequency response function from noise (speaker) S to the detection microphone $M1$ is indicated as G_{SM1} , and so on. O is an evaluation point (microphone).

The cancellation effect of the acoustic signal from the secondary source was already shown in Fig. 8. The acoustic signal is indicated as a propagation wave indicated in Eq. (4). So, each acoustic signal

Fig. 13 Noise reduction effect in 1-Microphone system

Fig. 14 Two-Microphone delay system

property at the microphone location is shown in the following form :

$$\begin{aligned} G_{SM1} &= e^{-jkl_1} \\ G_{SM2} &= e^{-jk(l_1 + \Delta l)} \\ G_{SO} &= e^{-jkl} \\ G_{AO} &= e^{-jka} \end{aligned} \quad (12)$$

The amplitude coefficient is one unit in magnitude. The output signal F of the 2-Microphone system against the noise signal S is indicated as follows :

$$F = S \cdot e^{-jkl_1} \cdot (1 - e^{-2jkl\Delta l}) \quad (13)$$

where Δl is the distance between two microphones. The model characteristic of Eq. (13) is shown in Fig. 15. There are insensitive frequencies f_{ins} determined by the distance between two microphones as follows :

$$f_{ins} = \frac{c}{2 \cdot \Delta l} \cdot n \quad (n=0, 1, 2, \dots) \quad (14)$$

This characteristic was confirmed by the experiment.

Since the transfer function causing the howling is rejected, the control filter G is represented as follows :

Fig. 15 Output characteristic of 2-Microphone system against the noise source

Fig. 16 Digital filter characteristic of 2-Microphone delay system

$$G = -\frac{G_{FO}}{G_{AO}} \quad (15)$$

Considering the system throughput delay as Eq. (9) and the transfer function as Eq. (12), this Eq. (15) is represented as follows :

$$G = \frac{-e^{-j[k\{l-(l_1+\Delta l+a)\}-\theta\omega]}}{2j \sin kx} \quad (16)$$

The model filter characteristic is shown in Fig. 16. The frequency at the maximum point where the digital filter is difficult to implement is calculated as follows :

$$\omega_n = \frac{c \cdot \pi}{\Delta l} \cdot n \quad (n=0, 1, 2, \dots) \quad (17)$$

At these points, the target filter coefficient has a phase lead. The implementation characteristic of the digital filter for a 2-Microphone delay system is shown in Fig. 17. The bold line shows the target control filter characteristic, and the narrow line shows the digital filter characteristic. The implementation is in good condition except at the frequencies determined by the distance between two microphones.

The noise reduction effect using this filter coefficient is shown in Fig. 18. A good noise reduction effect was obtained except at the frequencies of about 480 and 900 Hz. Neither the digital filter implements the control filter nor the 2-Microphone system can

Fig. 17 Experimental result of the digital filter implementation in 2-Microphone delay system

Fig. 18 Noise reduction effect in 2-Microphone delay system

sense the signal at these frequencies, as described in the previous section. For this reason, a peak was observed at the insensitive frequency of 480 Hz. The peak is due to the noncausality of the digital filter system. No howling phenomenon was observed other than the peak. If the distance between two microphones is shortened, the control region in the frequency domain is extended. This was also confirmed by the experiment. As the calculation error of 2-Microphone differential output increases when the distance between two microphones is shortened, the distance between two microphones must be determined according to the upper limit frequency in the control region.

6. Conclusions

The acoustic condition for a conventional 2-Microphone system has been considered theoretically. The howling reduction effect was confirmed by the experiment. The characteristic of the control filter

was examined theoretically and experimentally. The result and the noise reduction effect were compared with a 1-Microphone system.

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