Active Reduction of Noise by Additional Noise Source and Its Limit

Ken'iti Kido

Hiroshi Kanai

Masato Abe

Research Center for Applied Information Sciences, Tohoku University, Katahira 2-1-1, Sendai 980, Japan

This paper describes further investigations of an active noise control system in which an additional sound source is set close to the primary (noise) source. Successful application of this method to duct noise control has already been reported (Kido, 1987). The synthesized sound radiated by the additional source is identical to that of the primary source, except in polarity. The additional and primary sources form a dipole sound source with reduced effective radiation power. In theory, the distance between these two sound sources should be much less than the shortest wavelength in the required frequency range to realize an ideal dipole source. Then, the total sound pressure would be expected to attenuate in proportion to the square of the distance from the center of the sources, and little sound power would be radiated. However, in practice, the distance cannot be set small enough, so there is only a relatively small area around the dipole where the sound pressure attenuates in proportion to the square of the distance. Further afield, it attenuates in direct proportion to distance. Noise reduction is therefore limited. This paper describes the effects and the limits of performance of such a system as a function of wavelength and the dimensions of sound sources.

1 Introduction

The methods for active noise control fall mainly into two categories: reduction of the noise level in a specified region or in a specified direction (Onoda, 1968); or reduction of total noise power (Mangiante, 1977; Nelson et al., 1987). Most studies involve the former, since it is easier to realize.

Noise control for one-dimensional systems, such as a duct, aims to reduce the total noise power radiated from the duct end. Most methods of this type use a loudspeaker and a microphone of special directivity, or a pair of microphones and loudspeakers to prevent acoustic feedback from the additional loudspeaker to the microphone, from whose output the control signal is synthesized. One such system was reported by Takahashi (1987), who used a pair of microphones and a loudspeaker, well controlled by an adaptive system. This method is effective as long as the assumption of one dimensional space is satisfied within the duct. However, the assumption is not realized in the comparatively higher frequency range, where the width of duct is not so small compared to the wavelength of sound.

A more stable method was proposed by Kido (1987) for the active control of duct noise, though it is effective only at very low frequencies. In that method, a loudspeaker was set close to the duct-end, as an additional noise source, forming a dipole sound source. The report mainly dealt with the method of synthesizing the additional sound signal.

Contributed by the Noise Control and Acoustics Division for publication in the JOURNAL OF VIBRATION, ACOUSTICS, STRESS, AND RELIABILITY IN DESIGN. Manuscript received May 1988.

The present study describes (*1*) the application of an adaptive control system to the formerly presented method, and (*2*) investigations of the limits of noise reduction.

2 Use of Adaptive Filter

The system comprises an additional sound source, set close to the primary source, radiating sound of the same waveform as the primary source, but of inverse polarity. A loudspeaker is used as the additional sound source, with its volume velocity adjusted to the same value as that of the primary source. The effective sound power radiated by such a dipole source is much less than a single source if the volume velocity of one of the dipoles is the same as that of a single source (Kido, 1987).

A dipole sound source for duct noise control can be constructed such that the original noise enters from the upper end of a duct and radiates into space from the lower end. An additional loudspeaker, set close outside the lower end of the duct, is driven by a signal synthesized from the waveform picked up by a microphone set near the upper end of the duct. This is achieved using an impulse response estimated by a special technique (Kido, 1987), and evaluated by an FIR digital filter. Previous experiments have shown that, with the volume velocity of the loudspeaker adjusted until it matches that of the lower end of the duct, except for the polarity, the noise is virtually canceled (Kido, 1987).

A remaining major problem is that the transfer function from the upper end to the lower end of the duct may change with temperature and with the velocity of flow in the duct. It

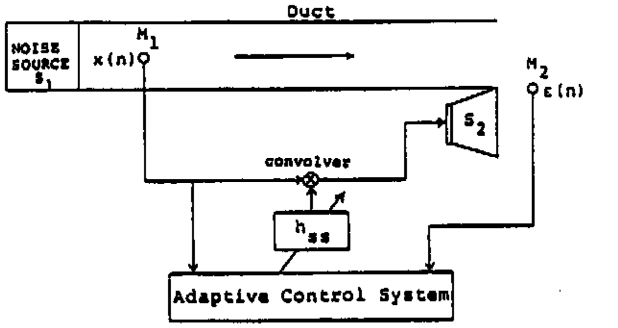

Fig. 1 Block diagram of the automatic noise control system for a duct

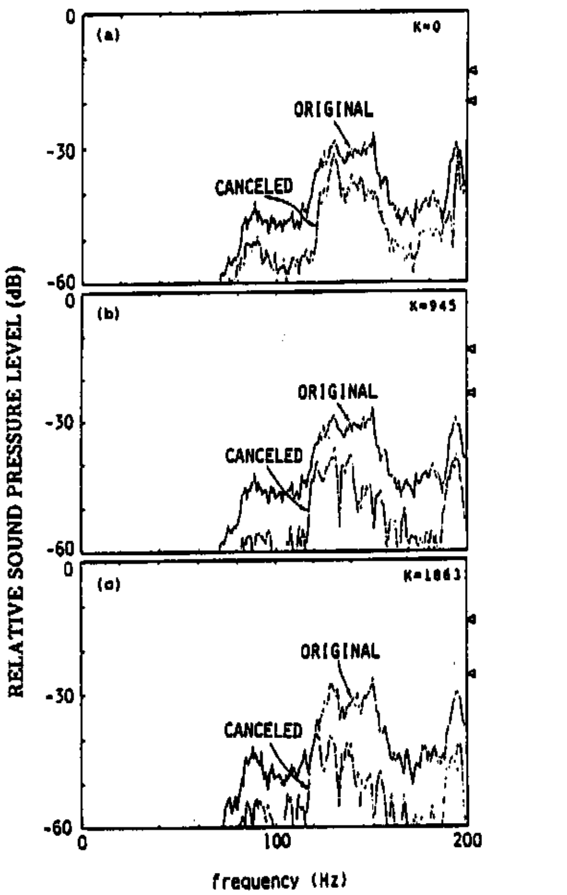

Fig. 2 The changes in the sound pressure levels of duct noise using an automatic noise control system with adaptive filter, where k is the number of repetition for adaptation, and the triangle marks at the right hand side indicate the total power levels

is therefore necessary to ensure that the impulse response of the FIR filter changes accordingly, in order for the system to remain effective. This problem is solved by using an adaptive filter to minimize the sound pressure at the position of a monitor microphone. Figure 1 shows a block diagram of the system. The filter coefficients are adaptively changed to minimize the output of the monitor microphone M_2. The FLMS (Fast Least Mean Square) algorithm (Ferrara, 1980; Mansour and Gray, 1982) was used in the system to shorten the time for adaptation.

Spectra for experimentally obtained noise reduction are given in Fig. 2. Figure 2(*a*) shows the initial state ($k = 0$) at which the filter coefficient was determined by the previously reported method (Kido, 1987). After 945 [Fig. 2(*b*)] and 1863 [Fig. 2(*c*)] repetitions of automatic adjustment, noise suppression was more than 10 dB and 12 dB, respectively.

The effectiveness of the adaptive system with temperature fluctuation is shown in Fig. 3: continuous recordings of attenuation and temperature during a 24 hour period. This experiment was carried out in a room without a facility of temperature control. About 12 dB of noise reduction was achieved regardless of a 3°C fluctuation in temperature.

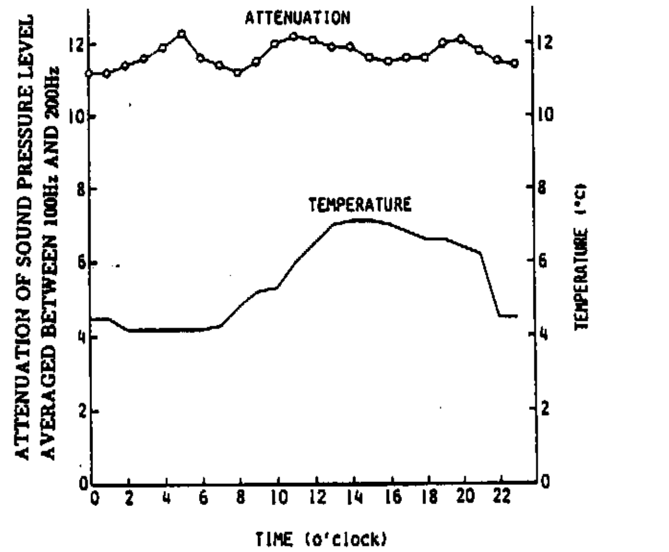

Fig. 3 The changes in temperature and in attenuation of sound pressure level of noise by the adaptive noise control system during a 24 hour period

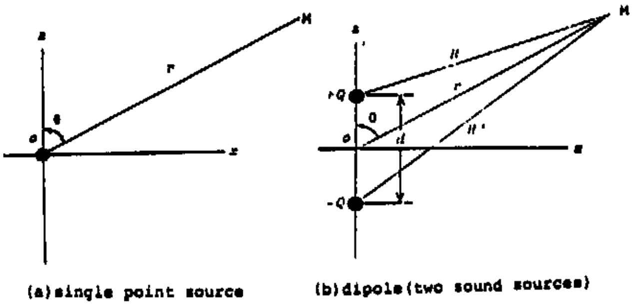

Fig. 4 Arrangement of point sources and monitor microphone (*a*) single point source (*b*) dipole (two point sources)

3 Attentuation of Radiated Power

Relatively low sound pressure levels at the monitor microphone M_2 (Figs. 2 and 3) show the effectiveness of the active noise cancellation system, but they are not sufficient to show the reduction of total radiated noise power. The latter will decrease if the sound sources form an ideal dipole source. However, since the primary and additional sound sources have some dimensions, the effect of the dimensions of the sound sources and the distance between the primary and additional sources requires investigation.

When a point sound source of volume velocity Q is at a point O in a free space, as shown in Fig. 4(*a*), the velocity potential ϕ_0 at point M is expressed as follows:

$$\phi_0 = \frac{Q}{4\pi r} \exp\{j(\omega t - kr)\} \tag{1}$$

where k is the wave number and r is the distance between the sound source and the point M. From equation (1) the power, W_0, radiated into the upper hemisphere shown in Fig. 5(*a*) is easily obtained as follows (Kido, 1987):

$$W_0 = \frac{Q^2}{16\pi} \rho c k^2 \tag{2}$$

The radiated power can be decreased by setting another

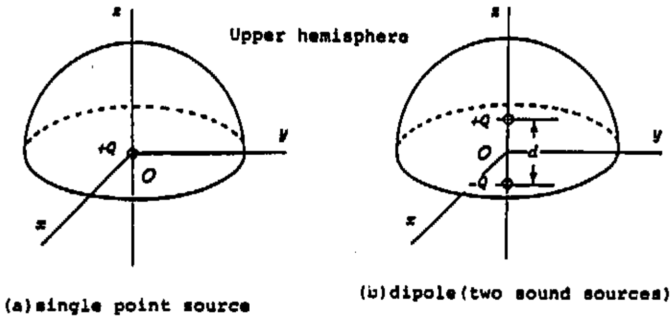

Fig. 5 Illustration of the point sources and radiation hemispheres for which the sound power was calculated

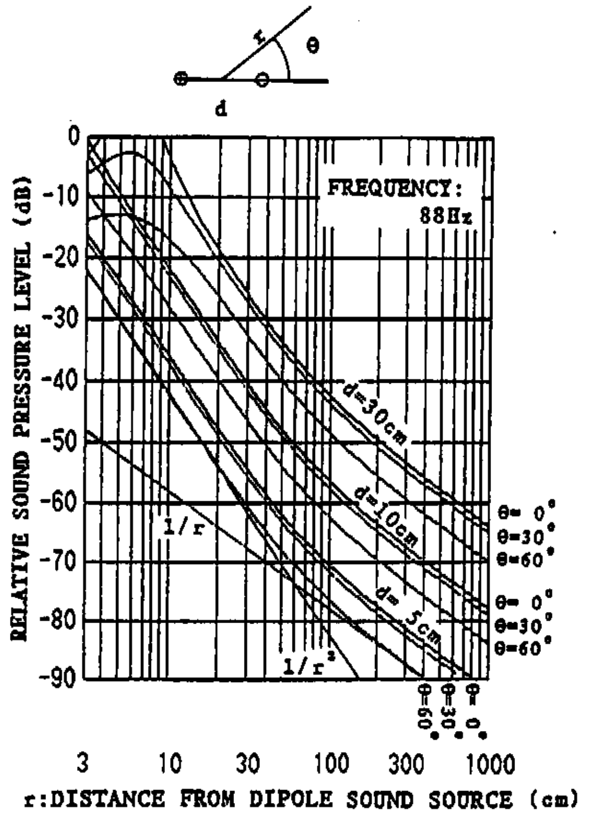

Fig. 6 Computed near field sound pressure distribution of the dipole sound source for different values of point separation *d*

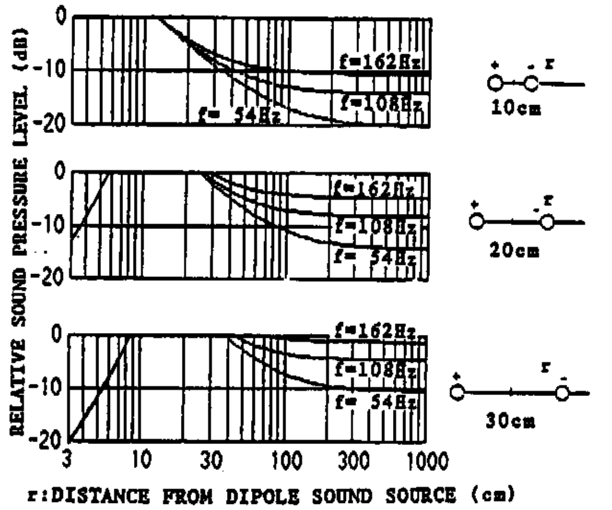

Fig. 7 Computed attenuation in sound pressure level by a dipole source compared to that by a single source

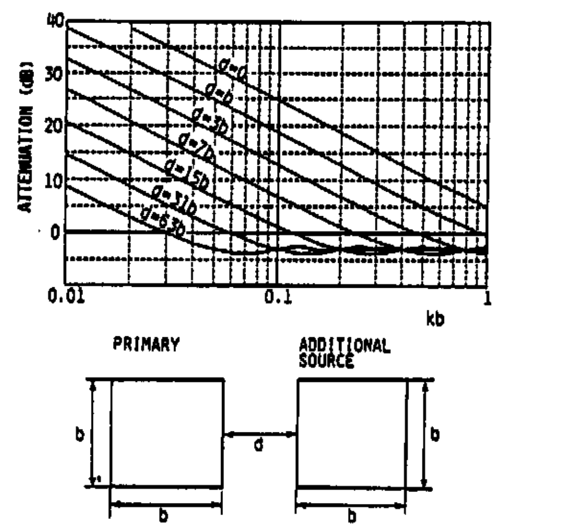

Fig. 8 Attenuation of the total radiated power by the addition of a sound source of similar dimensions but of inverse phase. (*k*, wave number; *b*, length of one side of rectangular sound-emitting area; *d*, distance between the edges of the sources).

sound source of the same volume velocity and inverse polarity close to the primary source. When the volume velocity of the additional source is $-Q$ and the distance between the two sound sources is d, as shown in Fig. 4(*b*), the velocity potential ϕ_d is expressed as follows:

$$\phi_d = \frac{Q}{4\pi R_1}\exp[j(\omega t - kR_1)] - \frac{Q}{4\pi R_2}\exp[j(\omega t - kR_2)] \quad (3)$$

Under the condition that $r \gg d$ and that d is sufficiently small, the sound power radiated by a dipole source into the upper hemisphere shown in Fig. 5(*b*) can be expressed as follows (Kido, 1987):

$$W_d \doteq \frac{Q^2}{8\pi}\rho c k^2\left(1 - \frac{1}{kd}\sin kd\right) \quad (4)$$

Figure 6 shows an example of the relation between the distance from the center of the two point sources and the sound pressure level. The sound pressure decreases in proportion to the square of the distance r from the center of the sources, within the very near field (closer than 30 cm), but decreases in proportion to the distance r in the far field.

The attenuation, A_p, of sound by the additional sound source is computed as follows:

$$A_p = 10\log\frac{W_d}{W_0} = 10\log\left\{2\left(1 - \frac{1}{kd}\sin kd\right)\right\} \quad [dB] \quad (5)$$

Figure 7 shows three examples of the attenuation obtained from a dipole source when the monitor microphone M_2 is in the 0 deg direction. The attenuation is larger in the lower frequency range and d is small.

The distance between two sources has a lower limit, since they are not point sources. Therefore, the effect of the dimensions of the sound sources was investigated by numerical computation. Both the primary and additional sound sources are considered as rectangular plates, with heights and widths b, as shown in Fig. 8. The primary plate vibrates uniformly in an infinite solid wall (infinite baffle board), and has a volume velocity G per unit area. Then, the velocity potential ϕ_0 at point M is expressed as follows:

$$\phi_0 = \int \frac{G}{4\pi r'}\exp[j(\omega t - kr')]dS \quad (6)$$

where r' is the distance from the point M to the center of a small area dS. The additional sound source is assumed to have

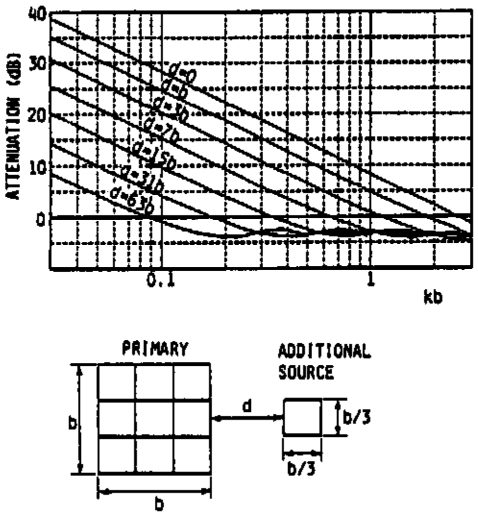

Fig. 9 Attenuation of total radiated power by the addition of a sound source with a surface area one-ninth that of the primary source

the volume velocity $-G'$ per unit area. Then, the following equation is obtained:

$$\phi_d = \int \frac{G}{4\pi R'_1} \exp\{j(\omega t - kR'_1)\} dS_1$$

$$- \int \frac{G'}{4\pi R'_2} \exp\{j(\omega t - kR'_2)\} dS_2 \qquad (7)$$

where R'_1 is the distance from point M to a center of a small area dS_1, and R'_2 is that from M to dS_2.

From equations (6) and (7), the sound pressure levels and sound powers radiated from single and dipole sources were calculated by numerical integration, and the attenuation was computed. Figures 8 and 9 show examples of the results of computation under the conditions that $R'_1 \gg d$ and $R'_2 \gg d$, respectively. The integration was performed as follows. The rectangular plates of the primary and additional sound sources are divided into many smaller ones, with heights and widths b'. The integrated volume velocity radiated from a smaller rectangular plate is approximated by that radiated from an imaginary point source located at the center of the plate, as follows:

$$\int G dS = G b'^2 = \Delta Q \qquad (8)$$

where ΔQ is the volume velocity of the point source—that is, the integration of the volume velocities of the small rectangular plates can be approximated by the summation of those of point sources. If the length b' is sufficiently smaller than the wavelength of the sound, the volume velocity obtained under the above mentioned approximation is almost the same as that given by equation (7). The results in Figs. 8–16 are for $b' = b/24$.

In Fig. 8, the area of the additional source is the same as that of the primary source, and the volume velocity G' is equal to G. In Fig. 9, the width of the primary source is b and that of the additional source is $b/3$, but their total volume velocities are the same. That is, the area of the additional source has only 1/9 of the area of the primary source, and the volume velocity $G' = 9G$.

In the symmetrical system (Fig. 8), the maximum attenuation is achieved when the two sources are closely arranged, i.e., for $d/b \approx 0$ where d is the distance between the edges of sources. The attenuation decreases in proportion to the value of $1/kb$ and is larger in the lower frequency (smaller kb) range than in the higher frequency (larger kb) range. The attenuation decreases when the additional source is far away from the

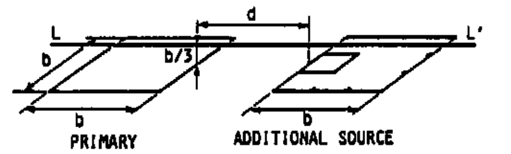

Fig. 10 The geometry of the primary and additional sound sources on an infinite baffle board. Sound pressure distribution was computed along the line L-L'.

primary source. The maximum attenuation for $d/b \approx 0$ is 25 dB at $kb = 0.1$.

For the asymmetrical system (Fig. 9), similar results are obtained (cf. Fig. 8). The maximum attenuation at $kb = 0.1$ is 28 dB for $d \approx 0$, and there is no attenuation in the range $kb > 1.0$, where $d/b = 1/3$.

For both situations (i.e., an additional source area equal to or one-ninth the area of the primary source) the radiated sound power is doubled when the distance between the two sources is much more than the sound wavelength, and the two sources radiate sounds independently. Then, attenuation by the additional sound source is -3 dB.

When $b = 30$ cm, $kb \approx 0.1$ corresponds to a frequency of 18 Hz and $kb = 3$ corresponds to 540 Hz. These figures indicate that this type of active noise suppression system is effective only at very low frequencies. However, since the usual sound absorptive materials are not very effective at low frequencies, and considerable space is required for the reactive silencer, the system offers effective active noise control at low frequencies.

4 Near Field Sound Pressure Distribution

The adaptive control system described briefly in Section 2 is considered to be effective when temperature and other boundary conditions cannot be controlled. The output of the monitor microphone M_2 is the reference signal for adaptation: the system minimizes the sound pressure at the point of the monitor microphone. The total radiated power will decrease if the additional source forms an ideal dipole source with the primary source and the monitor microphone is at the direction of minimum sound pressure for the dipole source, that is, on the plane $x-o-y$ shown in Fig. 5. If the monitor microphone is not set on the $x-o-y$ plane, the total radiated sound power may not necessarily decrease even if the output of the monitor microphone is set at zero.

When the primary source and the additional loudspeaker are different in shape and size, the sound pressure distribution in the near field is very different from the distribution of a dipole source, as shown by the following numerical computation.

Consider (Fig. 10) a primary source as a rectangular plate, of width b, which vibrates uniformly in an infinite baffle board. An additional source, of width b or $b/3$, is set beside the primary source to vibrate with the same waveform, but with opposite phase. The near field sound pressure distribution is computed using equation (7), without any approximation such as $R \gg d$. The line L-L' lies $b/3$ from the baffle board, above the centers of two sources.

Figure 11 shows the computed sound pressure distribution on the line L-L', where the sources are symmetrical (i.e., rectangles of width b). Three different computations are shown for different values of the distance between the two sources. In each case, the zero point of the sound pressure distribution is at the center of the interval between the two sources. This is therefore the optimum point at which to place the monitor microphone for the adaptive control system.

For the asymmetrical system (Fig. 12), the zero point of the sound pressure distribution shifts toward the primary source. When the two sources are adjacent, the zero point is on the surface of the primary source [Fig. 12(a)].

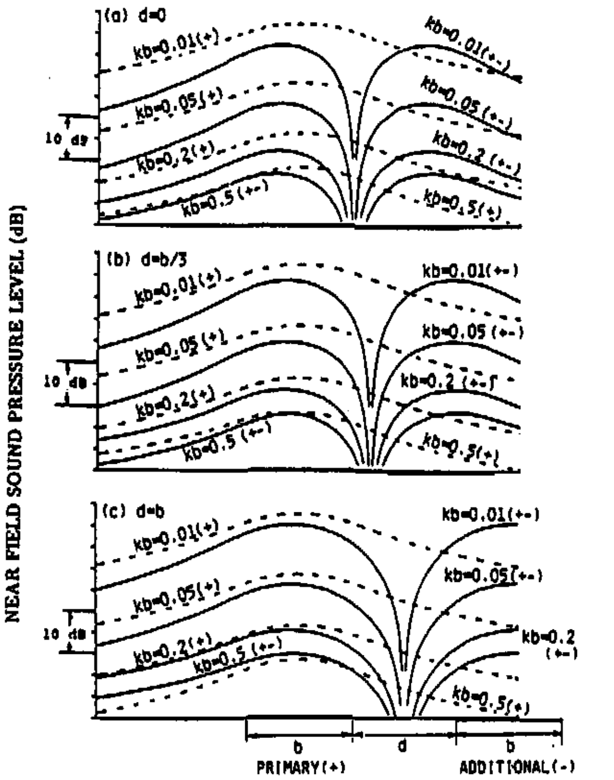

Fig. 11 The sound pressure distribution along the line L-L' (see Fig. 10) for three distances *b*. The broken lines indicate the sound pressure distributions for the cases of no additional source, and the solid lines show the sound pressure distribution when both sources are in operation.

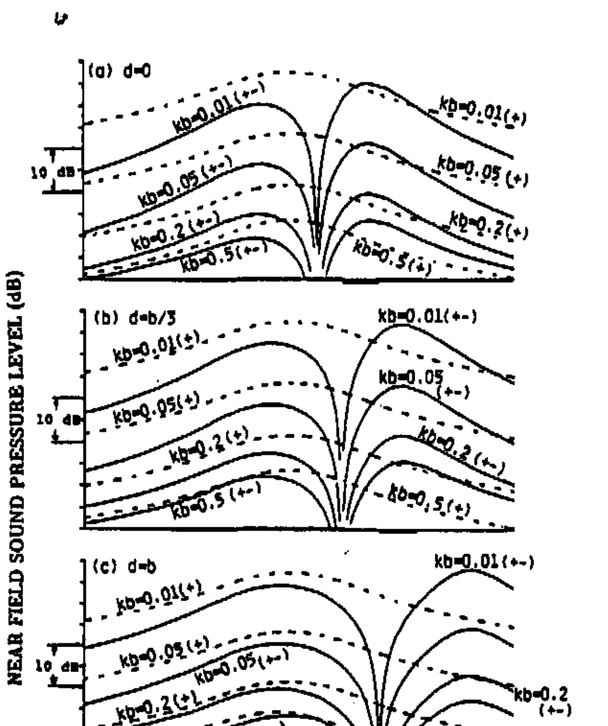

Fig. 12 The sound pressure distribution along the line L-L' (see Fig. 10) for three distances *d*. The area of the additional sound source is one-ninth that of the primary source, and the amplitude source is 9 times that of the primary source (i.e., the total volume velocity of the additional source is the same as that of the primary source).

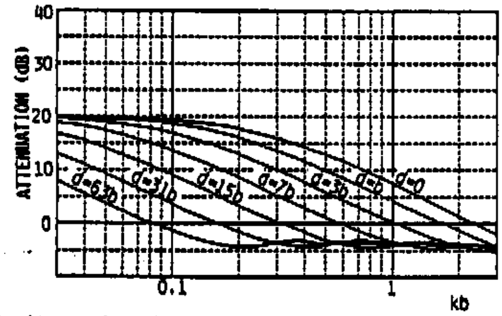

Fig. 13 Attenuation of total radiated power by a dipole source. The additional source has one-ninth the area and 110 percent of the volume velocity of the primary source.

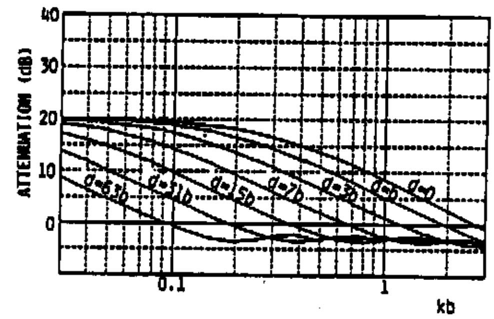

Fig. 14 Attenuation of total radiated power by a dipole. The additional source has one-ninth the area and 90 percent of the volume velocity of the primary source.

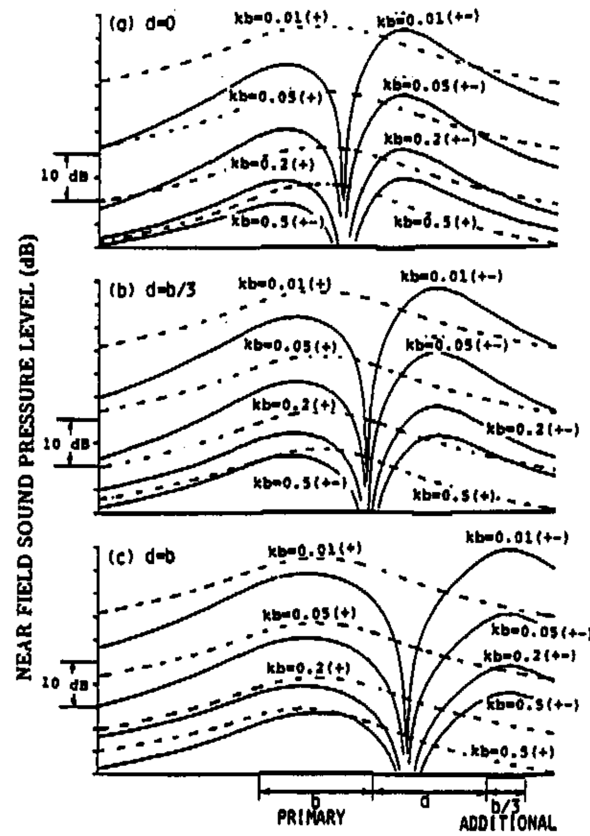

Fig. 15 The sound pressure distribution along the line L-L' (see Fig. 10) for the same asymmetrical dipole source as defined in Fig. 13

The results of the computations of near field sound pressure distribution (Figs. 11 and 12) therefore provide the information required for accurate placement of the monitor microphone,

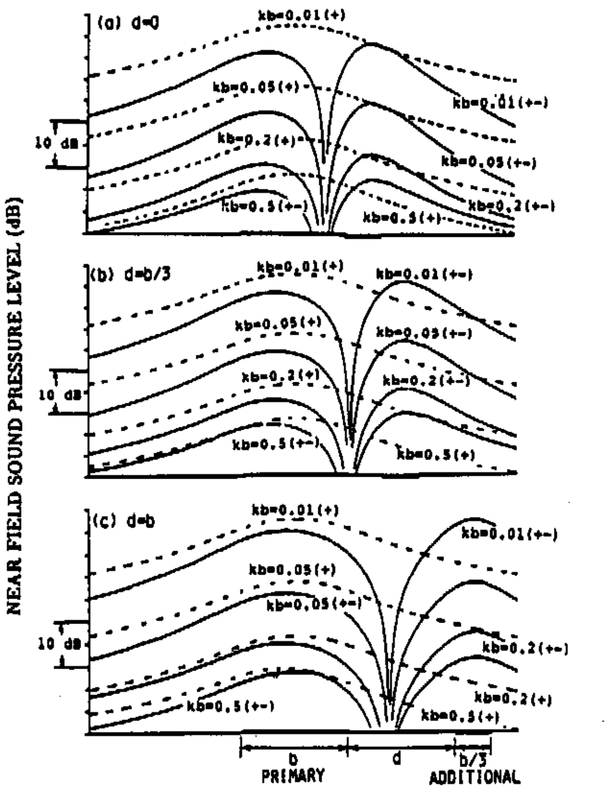

Fig. 16 The sound pressure distribution along the line L-L' (see Fig. 10) for the same asymmetrical dipole source as defined in Fig. 14

which should be set as close as possible to the zero point in order for the system to operate with maximum efficiency.

5 Effect of Errors on Attenuation

If the monitor microphone is not placed in the ideal position (i.e., at the point of zero sound pressure), the system will not operate efficiently, and noise attenuation will be lessened accordingly. The attenuation will also decrease when the total volume velocities of the two sources are unequal.

The effect of errors in the volume velocity on the total sound radiation has been computed numerically when the width of the additional source is 1/3 that of the primary source. Figures 13 and 14 show examples of attenuation in the total radiated power, where the volume velocity of the additional source differs by ± 10 percent from that of the primary source. Maximum attenuation is much lower in both cases, showing that the accuracy in matching the volume velocities is very important.

Changes in the volume velocity causes the zero point of the sound pressure distribution to shift. Figure 15 shows the sound pressure distribution on the line L-L' for a 10 percent increase in the volume velocity of the additional source: the zero point shifts in the direction of the primary sound source. In Fig. 16 there is a 10 percent decrease: the zero point shifts in the direction of the additional sound source.

The error due to the difference in the volume velocity can be reduced by using the adaptive filter explained in Section 2. However, the adaptive filter cannot control the positioning error of the monitor microphone.

6 Conclusion

From the theoretical and experimental investigations described in this paper, it is clear that (*1*) the total radiated sound power can be reduced by creating a dipole source, as long as the wavelength of sound is sufficiently longer than the distance between the two sound sources comprising a dipole; (*2*) the total radiated power also decreases even if the two sound sources are not point sources but rectangular plates; (*3*) the attenuation to total radiated sound power is affected by the size of the sound source, by the difference in the volume velocities of the two sound sources, and by the positioning error of the monitor microphone; and (*4*) the error due to the difference in the volume velocities of the two sound sources and the error due to the slow change of the impulse response of duct can be reduced by using the adaptive filter, but it cannot control the positioning error of the monitor microphone. In practice, the attenuation of sound pressure level will probably be limited to less than 20 dB. Better results should be obtained if a small-sized loudspeaker (with high performance at low frequencies) is used as the additional sound source.

Acknowledgments

The authors would like to thank the Sendai Institute of Applied Information Sciences for its support, and Messrs. S. Morikawa and A. Shima for their assistance in the experiments.

References

Onoda, S., and Kido, K., 1968, "Automatic Control of Stationary Noise by Means of Directivity Synthesis," *Proc. 6th ICA Congress*, Tokyo, Vol. IV, pp. F-185–F-188.

Mangiante, G. A., 1977, "Active Sound Absorption," *J. Acoust. Soc. Am.*, Vol. 61, No. 6, pp. 1516–1523.

Ferrara, E. R., 1980, "Fast Implementation of LMS Adaptive Filters," *IEEE Trans.*, Vol. ASSP-28, No. 4, pp. 474–495.

Mansour, D., and Gray, Jr., A. W., 1982, "Unconstrained Frequency-Domain Adaptive Filter," *IEEE Trans.*, Vol. ASSP-30, No. 5, pp. 726–734.

Kido, K., Morikawa, S., and Abe, M., 1987, "Stable Method for Active Cancellation of Duct Noise by Synthesized Sound," ASME JOURNAL OF VIBRATION, ACOUSTICS, STRESS AND RELIABILITY IN DESIGN, Vol. 109, pp. 37–42.

Nelson, P. A., Curtis, A. R. D., Elliot, S. J., and Bullmore, A. J., 1987, "The Minimum Power Output of Free Field Point Sources and the Active Control of Sound," *Journal of Sound and Vibration*, Vol. 116, No. 3, pp. 397–414.

Takahashi, M., Kuribayashi, T., Asami, K., Enokida, T., Hamada, H., and Miura, T., 1987, "Broadband Active Sound Control System for Air-Conditioning Duct Noise," *J. Acoust. Soc. Jpn.*, (E), Vol. 8, No. 6, pp. 263–269.