

Development of a loudspeaker system with a unidirectional radiation pattern in a speech frequency range

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1. Introduction

Generally to realize superior directivity of a loudspeaker, the diaphragm size of a direct radiation loudspeaker or the mouth size of a horn loudspeaker must be large enough compared with the wavelength of sound. Several challenges to overcome this restriction had been developed; e.g. the parametric loudspeaker, the ultra directional loudspeaker and the similar developments. However because of larger size or more complicated configuration, they had been limited in restricted application only.

This paper reports on the results of our investigation into a loudspeaker system which provides the superior unidirectional radiation pattern in the speech frequency range, and to verify the system's effectiveness.

2. Unidirectional radiation pattern

If a primary point source and a secondary point source are arranged as shown in Fig. 1, the composite sound from these sources can be expressed by Eq. (1)¹⁾:

$$\dot{P}(r, \theta) = \frac{j\omega\rho_0\dot{Q}_p e^{-jk r_p}}{4\pi r_p} + \frac{j\omega\rho_0\dot{Q}_s e^{-jk r_s}}{4\pi r_s} \quad (1)$$

where, \dot{Q} : represents the strength of the sound source (air volume discharged by sound source per second).

Assuming that the observation point is far enough from each source, as compared with the wavelength and the source-to-source distance d , the following Eq. holds:

$$\begin{aligned} r_p &\approx r + (d/2)\cos\theta \\ r_s &\approx r - (d/2)\cos\theta \end{aligned}$$

If r is infinite, then, $r_p \approx r_s \approx r$.

If the sound pressure at such an observation point is 0

$$\dot{Q}_s = -\dot{Q}_p e^{-jkd \cos\theta_0} \quad (2)$$

Substitute Eq. (2) into Eq. (1)

$$\dot{P}(r, \theta) = \frac{j\omega\rho_0\dot{Q}_p e^{-jk r_p}}{4\pi} \left[\frac{1}{r_p} - \frac{e^{-jk(r_s - r_p + d\cos\theta_0)}}{r_s} \right] \quad (3)$$

When the observation point is far enough,

$$\begin{aligned} r_s - r_p &= -d\cos\theta \\ r &\approx r_p \approx r_s \end{aligned}$$

Eq. (3) can be rewritten as Eq. (4):

$$\dot{P}(r, \theta) = \dot{P}_p(r, \theta) [1 - e^{-jkd(\cos\theta_0 - \cos\theta)}] \quad (4)$$

From

$$|1 - e^{-jx}|^2 = 2(1 - \cos x)$$

Eq. (4) is written as Eq. (5):

$$\frac{|\dot{P}(r, \theta)|^2}{|\dot{P}_p(r, \theta)|^2} = 2(1 - \cos[kd(\cos\theta_0 - \cos\theta)]) \quad (5)$$

Since $\dot{P}_p(r, \theta)$ is a point sound source, $2(1 - \cos[kd(\cos\theta_0 - \cos\theta)])$ represents the radiation pattern. When the secondary source is controlled to effect the sound pressure of 0 at $\theta=0$, then Eq. (5) can be rewritten as Eq. (6):

$$\frac{|\dot{P}(r, \theta)|^2}{|\dot{P}_p(r, \theta)|^2} = 2(1 - \cos 2kd) \quad (6)$$

Accordingly, when $d = \lambda/4$,

$$|\dot{P}(r, \theta)|^2 / |\dot{P}_p(r, \theta)|^2 = 2$$

i.e., the maximum sound pressure of composite sound is double the sound pressure of the primary source.

As understood from above, by setting the source-to-source distance d at $1/4$ the wavelength, the system can realize a unidirectional radiation pattern in which the sound pressure is highest in the direction $\theta = \pi$.

3. System components

Assuming that the sound from a primary or secondary loudspeaker is an ideal point source the unidirectional radiation pattern is expressed by Eq. (6), and the condition for driving the secondary loudspeaker can also be obtained from Eq. (2). However, in many actual applications, sound from a loudspeaker cannot be regarded as a point sound source. Therefore, the loudspeaker system cannot produce the desired radiation pattern if the ideal condition is used. In view of this, we examined the control of secondary loudspeaker by adaptive filtering methods and arrangement. Figure 2 shows the system configuration. This system is composed of the primary loudspeaker, the secondary loudspeaker, the unique sound reflector, one sensing microphone and the signal processor. Each loudspeaker is formed with a horn driving unit and a reentrant horn, for miniaturization. The mouths of both horns are placed closely facing each other, and the sound reflector is installed between them. The sound

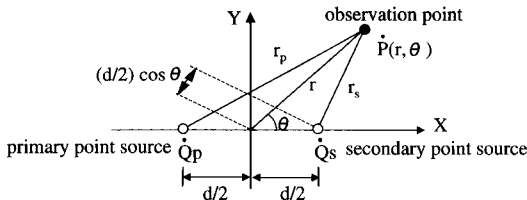


Fig. 1 Primary and secondary point sources separated by a distance.

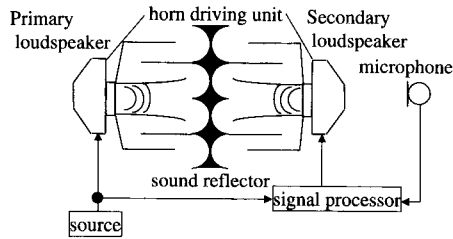


Fig. 2 Schematic diagram of developed loudspeaker system. Primary, secondary loudspeakers and sound reflector are sectional view.

wave from each loudspeaker is designed to radiate in the opposite direction by using this composition. The sound reflector not only changes the direction of each sound propagation axis by 180° , but also fixes the distance between the sound-radiating surfaces of the two loudspeakers. As described in Section 2, with a loudspeaker system providing a unidirectional radiation pattern, the highest sound pressure is obtained in the direction $\theta = \pi$ if the distance between the two sound sources is set at $1/4$ the wavelength. An audio input signal is directly fed to the primary loudspeaker and cancellation signal generated by the signal processor is fed to the secondary loudspeaker. The signal processor creates the cancellation signal by means of the adaptive filtering methods (Filtered-X LMS algorithm).^{2,3)} The cancellation signal providing to secondary loudspeaker is designed to create the 180° -degree out-of-phase sound wave at the sensing microphone comparing to that wave of the primary loudspeaker. It is not necessary for the system to constantly update the factor of the adaptive filter. When output from the microphone becomes sufficiently small, the system may stop the factor-updating operation. If the factor of the adaptive filter for the minimum microphone output is reproduced by an FIR filter, a simple signal processor can produce the same effect as the adaptive filtering methods.

The sound field near each loudspeaker was simulated using the boundary element method (BEM). Figure 3 shows the simulation models. For Type A, two reentrant horns were placed back to back. For Type B, the system configuration shown in Fig. 2. Figure 4

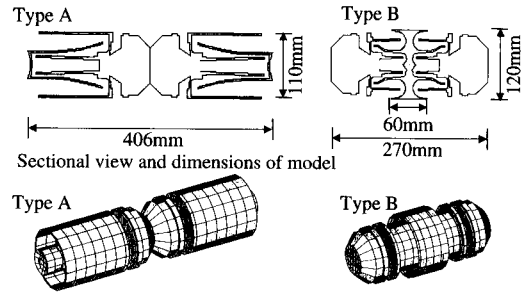


Fig. 3 Boundary element method model.

Type A: Two reentrant horns are placed back to back. Type B: The mouths of the two horns are placed closely facing each other, and a sound reflector is installed between them.

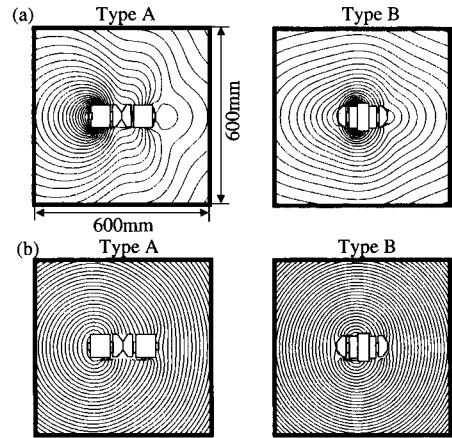


Fig. 4 The distribution of (a) sound pressure and (b) constant phase at 1 kHz when the primary loudspeaker is driven. Sound pressure: 1 dB step, constant phase: 20 deg step.

shows the sound pressure and constant phase distributions near each loudspeaker when the primary loudspeaker is driven. With Type A, sound pressure distribution is complex. With Type B, sound pressure distribution centers around the primary loudspeaker. Since the primary and secondary loudspeakers of Type B can provide a sound radiation pattern resembling that with a point sound source, the unidirectional radiation pattern of Type B meets Eq. (6). Each loudspeaker of Type B can radiate spherical waves in the area close to both loudspeakers. This is why the sound pressure and constant phase distributions are constant regardless of which loudspeaker is driven.

4. Active control of radiation pattern

The sound field analysis described in Section 3 reveals the sound pressure and constant phase distributions in the region near each. If the microphone is

positioned in such a region, the sound from the secondary source can achieve sound cancellation in only a limited area, making it difficult for the system to produce a unidirectional radiation pattern. The subsequent paragraphs present a simulation study conducted to determine the optimal microphone position for unidirectional radiation pattern, and describe the radiation pattern of the developed system.

4.1 Simulation

The unidirectional radiation pattern with Type A and Type B was studied by means of BEM simulation. Figure 5 shows difference in sound pressure and constant phase between primary loudspeaker alone driven and secondary loudspeaker alone driven (each loudspeaker is driven for the same sound pressure and phase). With Type A, the difference is significant. With Type B, although the difference of sound pressure can be observed near each loudspeaker, however it becomes small with distance from the loudspeakers. With Type B, therefore, if the microphone is positioned at a certain distance from the loudspeakers, the spatial radiation pattern can be controlled at one point. Figure 6 shows the relation between microphone position and radiation pattern. If the microphone is positioned close to the secondary loudspeaker, sound radiation in the direction 0° becomes large. Therefore, the microphone must be at least 20 cm from the secondary loudspeaker. With Type A, the radiation pattern is not cardioid under all conditions, due to characteristic sound pressure and constant phase, as shown in Fig. 4.

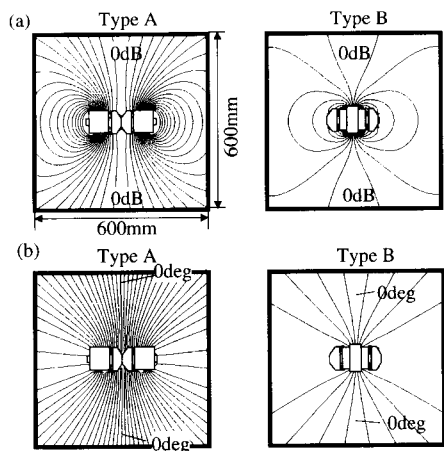


Fig. 5 The distribution of (a) sound pressure difference and (b) constant phase difference at 1 kHz between primary loudspeaker alone driven and secondary loudspeaker alone driven (sound pressure: 1 dB step, constant phase: 20 deg step). Each loudspeaker is driven for the same sound pressure and same phase.

4.2 Measurement

We developed the system of Type B based on simulation results. Figure 7 shows the frequency response in the front ($\theta = 180^\circ$) and in back ($\theta = 0^\circ$) when primary loudspeaker alone driven. Figure 8 shows the radiation pattern with the developed system. The system achieves a unidirectional radiation pattern in the frequency range between 500 Hz and 4 kHz. In this range, sound pressure difference between the front and the back is approximately 20 dB. At frequencies higher than 4 kHz, there is sound pressure difference between front and back even if the primary loudspeaker

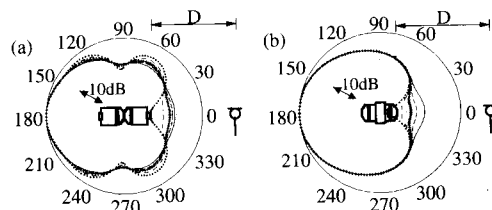


Fig. 6 Polar diagrams of (a) Type A's and (b) Type B's simulated unidirectional radiation pattern at 1 kHz. — D : 0 cm, ---- D : 5 cm, — · — D : 10 cm, ···· D : 20 cm, - - - - D : 40 cm. (D is distance between microphone and secondary loudspeaker).

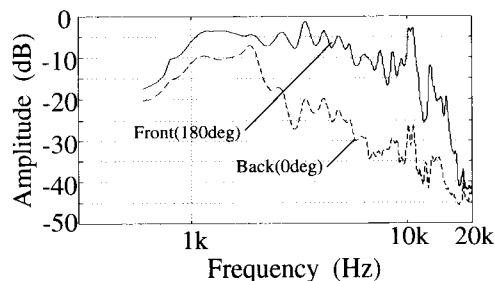


Fig. 7 Amplitude-frequency characteristic when primary loudspeaker alone driven.

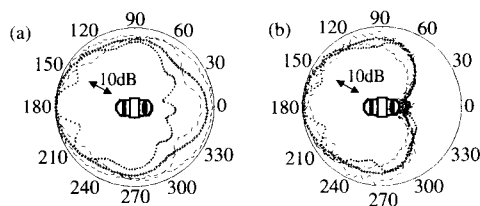


Fig. 8 Polar diagrams of (a) Type B's primary loudspeaker alone driven, (b) Type B's primary and secondary loudspeakers driven measured unidirectional radiation pattern. ---- at 500 Hz octave band, — at 1 kHz octave band, ··· at 2 kHz octave band, - - - - at 4 kHz octave band.

alone is driven. Therefore, the secondary loudspeaker should be driven in the frequency range below 4 kHz, to achieve the unidirectional radiation pattern.

5. Applications (Reduction of siren sound in an ambulance)

Some experiments were conducted to study the practicability of the developed systems.

The siren sound of an ambulance disturbs various emergency medical care procedures carried out in the ambulance, such as conversation of patient and ambulance attendant, communication by wireless, and examination with a stethoscope (see Fig. 9).⁴⁾ In Japan, the siren sound volume for the ambulance is legally regulated, as given in Table 1. Accordingly, the siren sound power level measured inside an ambulance is around 80 dB.

Generally, the siren loudspeakers for an ambulance are installed in the front part of the vehicle, and the medical care space is located in the rear part. Therefore, to ensure sufficiently high siren sound volume ahead of the ambulance, while minimizing the sound power level inside the medical care space, the siren sound source should provide higher sound radiation to the front than to the back. In addition, since the siren sound source for an ambulance is installed atop the vehicle, it must be small. With the conventional horn design or array loudspeaker design, therefore, it is

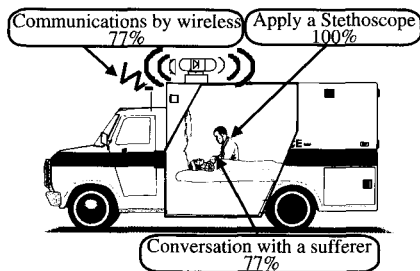


Fig. 9 Obstruction of first aid due to siren sound. Result of questionnaire to 38 Tokyo Fire Department ambulance attendants. Number is the percentage of the ambulance attendants, who have an obstruction feeling.

Table 1 A provision of ambulance siren sound in Japan.

Level	90 dB or more less than 130 dB at 20 m in front of the ambulance
Characteristic	1) Fundamental frequencies low frequency : 770 Hz high frequency : 960 Hz 2) Length of sound 770 Hz and 960 Hz are respectively running for 0.65 second mutually

difficult to achieve satisfactory radiation pattern control in the fundamental frequencies of siren sound. The siren sound volume ahead of the ambulance and that in the medical care space, were studied by producing a unidirectional radiation pattern with Type B. Figure 10 and 11 show the effectiveness of Type B in reducing the siren sound power level in the ambulance, around the stretcher. That is, Fig. 11 shows the siren sound reduction distributions measurements taken in a plane set at the head height of a patient lying on a stretcher (low measuring plane), and in a plane set at the head height of the ambulance attendant in a seated position (high measuring plane). Figure 11 shows the mean of 80 measurements in each measuring plane. For these experiments, the siren sound volume was set so that the sound pressure measured at a point 20 m ahead of the test vehicle was 90 dB. The experiments revealed that the siren sound power level with Type B is about 12 dB lower in all measuring planes than with the conventional siren loudspeaker. This indicates that Type B

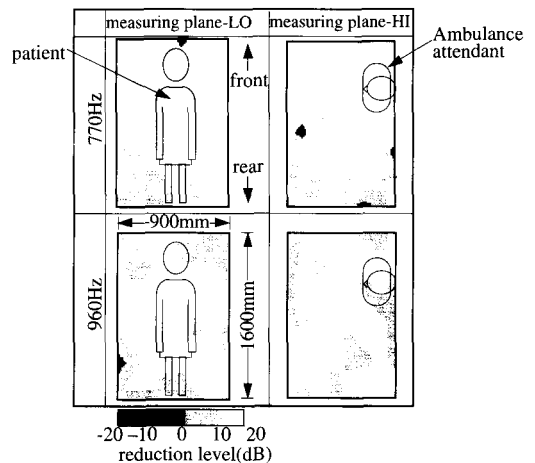


Fig. 10 The distribution of siren sound reduction level within the ambulance. Type B (unidirectional radiation pattern) vs. conventional siren loudspeaker. Measuring plane-LO : at the head height of a patient lying on a stretcher. Measuring plane-HI : at the head height of the ambulance attendant.

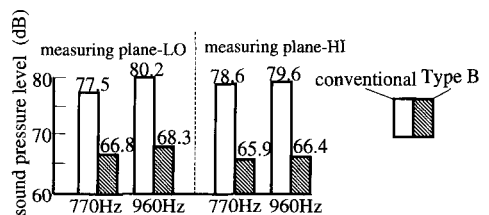


Fig. 11 The mean of siren sound pressure level in each measuring plane.

can reduce the siren sound power level inside the ambulance, while maintaining the necessary sound volume ahead of the ambulance. The above-mentioned experiments were conducted jointly with the Tokyo Fire Department's Fire Science Laboratory. Studies are continuing at the Laboratory, to further develop the system for practical application.

6. Conclusion

A unidirectional loudspeaker system comprising two loudspeakers and one microphone has been developed. Simulation and experimental studies have revealed that, given the optimal microphone position and the optimal distance between the sound-radiating surfaces of the primary and secondary loudspeakers, the developed system can realize a unidirectional radiation pattern in

the frequency range from 500 Hz to 4 kHz. This is achieved by simply controlling the sound of the secondary loudspeaker, at the position of the microphone.

References

- 1) P. A. Nelson and S. J. Elliott, *Active Control of Sound* (Academic Press, London, 1992).
- 2) B. Widrow and S. Stearns, *Adaptive Signal Processing* (Prentice-Hall, Englewood Cliffs, N. J., 1986).
- 3) H. Hamada, "Signal Processing for Active Control," Proc. Int. Symp. Active Control of Sound and Vibration, 33-44 (1991).
- 4) K. Waki, M. Nakanishi, K. Sasaki, and Y. Oohara, "Study on reducing of siren sound in ambulance (Series I)," Rep. Fire Sci. Lab. **33**, 119-132 (1996) (in Japanese).