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An Analytical Model for Understanding Active Directivity Noise Control with Near-Field Error Sensing

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Abstract: Recent research shows that a directional quiet zone from near to far field can be created with a near-field microphone array. However, there is no systematic theoretical analysis of this error-sensing strategy. In this paper, an analytical model of the angular spectrum of point sources in the free field is proposed, based on which the near-field error-sensing strategy for active directivity noise control can be quantitatively analyzed. It is found that the noise reduction in the far field is determined by the angular spectrum estimation error of both the primary and secondary sources. The impacts of the error microphone array configuration, sound source position, and target direction on the estimation error of angular spectrum are investigated. Finally, experiments were conducted in the anechoic chamber to validate the impact of the location of the secondary source.

Keywords: active noise control; directivity control; angular spectrum analysis



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1. Introduction

Active noise control (ANC) is an effective method to reduce transformer noise, which is dominated by harmonic components at low frequencies. Global control of transformer noise can be achieved by placing a sufficient number of secondary sources and error sensors surrounding it [1]. However, hundreds of control channels are required due to the large size of transformers, which makes the control system complicated and expensive [2]. Therefore, local control systems, which aim to create quiet zones in a target area, are investigated [3], which provides an alternative solution.

The error-sensing strategy is an important issue in the design of compact ANC systems [4]. The noise was reduced by 6 dB or more within an angle of approximately 23° in the far field when an error microphone was placed 30 m away from the transformer [5]. Directivity control from near to far field was achieved by adopting the sum of the squared sound pressures at several points within an angle as the cost function in the active noise control barriers [6]. However, both of the above-mentioned strategies use error points that are far from the primary source. For the stability and compactness of ANC systems, error sensors should be placed in the near field of the primary and secondary sources.

Different near-field error-sensing strategies have been compared to reduce the sound radiation from a monopole and dipole primary source and it was found that the most appropriate strategy is minimizing the sum of the weighted mean active intensity in a direction normal to the surface surrounding all the primary and control sources [7]. A strategy of minimizing the near-field sound intensity perpendicular to the barrier at specific points [8] has been found to be more effective in reducing far-field noise than minimizing the squared sound pressure [9]. However, sound intensity probes are required, and the reduction performance is not always better than that achieved by minimizing the near-field sound pressure [4].

A novel error-sensing strategy was proposed recently, where a microphone array is placed near the sources and the weighted sum of sound pressures at these microphones

is minimized to achieve directivity control [10]. The weighting factor relates to the phase shift between the error microphones and the plane perpendicular to the direction where noise reduction is needed. It was found that the minimum size of the array should be twice the size of the primary source, which is consistent with the requirements in the sound power measurement of a sound source [11]. The maximum microphone spacing is half the wavelength of the frequency of interest, which is similar to the discussions in Beamforming [12]. However, there is no systematic analysis of this error-sensing strategy.

The weighted sum of sound pressure has already been defined as an “angular spectrum” in the study of Fourier acoustics [13]. This technique decomposes the wave field into a series of plane waves that propagate independently, which are superimposed to obtain the field at the desired axial depth. The angular spectrum method can also be employed to characterize pressure fields for various sources [14,15] and to model sound transmission in both layered homogeneous [16] and inhomogeneous media [17]. The computation time to calculate wave propagation in parallel planes can be significantly reduced by employing the angular spectrum method rather than the point-by-point direct integral methods [18]. The surface velocity of sources through acoustic holography can also be effectively predicted by the angular spectrum method [19].

In this paper, an analytical model of the angular spectrum is employed to explain the near-field error-sensing strategy proposed in Ref. [10] for active directivity control. First, the analytical model for angular spectrum estimation of a point source in free field is derived and the directivity control strategy is analyzed. Then, angular spectrum estimation errors and the far-field noise reduction performance with the near-field error strategy are investigated based on the numerical simulations. Finally, the impact of the location of the secondary source is verified by experiments.

2. Theory

Figure 1 shows the theoretical model used in this study. A monopole primary source is located at $\mathbf{r}_p = (x_p, y_p, z_p)$ in the free field with a strength of q_p and a secondary source is located at $\mathbf{r}_s = (x_s, y_s, z_s)$. The error plane with a dimension of $L \times L$ is placed at $z = 0$, and both the primary and secondary sources are at the same side of it. The target direction of noise reduction is defined by the wavevector $\mathbf{k} = (k_x, k_y, k_z) = (k \sin \theta \cos \varphi, k \sin \theta \sin \varphi, k \cos \theta)$ with $0 \leq \theta < \pi/2$, $0 \leq \varphi < 2\pi$, where k is the wavenumber; k_x , k_y , and k_z are the components of the wavevector in the x , y , and z directions, respectively. A far-field point in the target direction, $\mathbf{r}_0 = (x_0, y_0, z_0)$, is used to evaluate the directivity control performance.

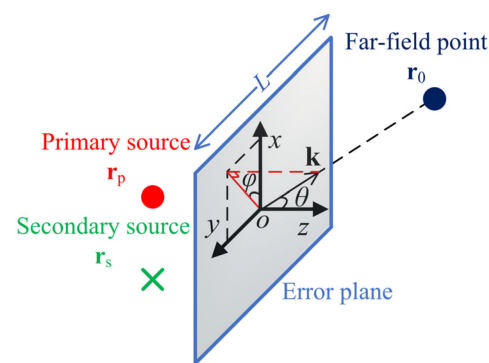


Figure 1. Schematic diagram of the theoretical model.

To reduce the noise in the target direction, the sound pressure at the far-field point $\mathbf{r}_0$ can be directly adopted as the cost function [20]

$$J_1 = \left| \frac{j\omega\rho_0 q_p e^{-jk|\mathbf{r}_0 - \mathbf{r}_p|}}{4\pi|\mathbf{r}_0 - \mathbf{r}_p|} + \frac{j\omega\rho_0 q_s e^{-jk|\mathbf{r}_0 - \mathbf{r}_s|}}{4\pi|\mathbf{r}_0 - \mathbf{r}_s|} \right|^2 \quad (1)$$

where j , ω and ρ_0 are the unit imaginary number, angular frequency and air density, respectively. The optimized strength of the secondary source can be obtained by minimizing Equation (1), which is

$$q_{s,1} = -\frac{|\mathbf{r}_0 - \mathbf{r}_s|}{|\mathbf{r}_0 - \mathbf{r}_p|} q_p e^{-jk(|\mathbf{r}_0 - \mathbf{r}_p| - |\mathbf{r}_0 - \mathbf{r}_s|)} \quad (2)$$

In the study of Fourier acoustics, the angular spectrum method is employed to decompose a complex sound field into a superposition of plane waves from different directions; therefore, the angular spectrum in target direction can also be adopted as the cost function in active directivity noise control. The angular spectrum through the $z = 0$ plane along the direction of $\mathbf{k}$ is [13]

$$\begin{aligned} P_p(\mathbf{k}, 0) &= \iint_S p(x, y, 0) e^{j(k_x x + k_y y)} dS \\ &= \frac{j\omega\rho_0 q_p}{4\pi} e^{j(k_x x_p + k_y y_p)} \iint_S \frac{e^{-jk\sqrt{x^2+y^2+z_p^2}}}{\sqrt{x^2+y^2+z_p^2}} e^{j(k_x x + k_y y)} dS \end{aligned} \quad (3)$$

where S and $p(x, y, 0)$ represent the infinite integration plane at $z = 0$ and sound pressure at a point $(x, y, 0)$ in the plane.

According to Ref. [21], Equation (3) is consistent with the formula for calculating the sound pressure at (x, y, z_p) generated by an infinite plate in the $z = 0$ plane. By extending the relevant conclusion in Ref. [13] to the complex domain, Equation (3) can be written as

$$P_p(\mathbf{k}, 0) = \frac{\rho_0 c_0 q_p}{2 \cos \theta} e^{jk r_p}. \quad (4)$$

The angular spectrum of the secondary source in the direction of $\mathbf{k}$, i.e., $P_s(\mathbf{k}, 0)$, can also be calculated referring to Equation (4).

If the total angular spectrum in the direction $\mathbf{k}$ is defined as the cost function

$$J_2 = |P_p(\mathbf{k}, 0) + P_s(\mathbf{k}, 0)|^2 \quad (5)$$

then the optimized strength of the secondary source is

$$q_{s,2} = -q_p e^{-jk(\mathbf{r}_s - \mathbf{r}_p)} \quad (6)$$

If the receiver is placed sufficiently far away, it can be assumed that $|\mathbf{r}_0 - \mathbf{r}_p| \approx |\mathbf{r}_0 - \mathbf{r}_s|$ and $\mathbf{k}(\mathbf{r}_s - \mathbf{r}_p) \approx k(|\mathbf{r}_0 - \mathbf{r}_p| - |\mathbf{r}_0 - \mathbf{r}_s|)$, then

$$q_{s,2} \approx -q_p e^{-jk(|\mathbf{r}_0 - \mathbf{r}_p| - |\mathbf{r}_0 - \mathbf{r}_s|)} \approx q_{s,1}. \quad (7)$$

The optimized strengths obtained with Equations (2) and (6) are the same, which indicates that the angular spectrum control is equivalent to the performing sound pressure control at the far-field point in the target direction.

In practice, the sum of the weighted sound pressures at M microphones spaced d apart in the $z = 0$ error plane is used to approximate the integral result in Equation (3) [10]

$$\hat{P}(\mathbf{k}, 0) = \sum_{m=1}^M p(x_m, y_m, 0) W(x_m, y_m) \quad (8)$$

where $\hat{P}(\mathbf{k}, 0)$ and $p(x_m, y_m, 0)$ are the estimated angular spectrum and the sound pressure at the m -th microphone located at $(x_m, y_m, 0)$. $W(x_m, y_m)$ is the weighting factor, which is defined as

$$W(x_m, y_m) = e^{j(k_x x_m + k_y y_m)}. \quad (9)$$

If the sum of the weighted sound pressures at the M microphones is defined as the cost function, the optimized strength of the secondary source can be obtained as [10]

$$q_{s,3} = -\left(\mathbf{Z}_{se}^H \mathbf{W}^H \mathbf{W} \mathbf{Z}_{se}\right)^{-1} \mathbf{Z}_{se}^H \mathbf{W}^H \mathbf{W} \mathbf{Z}_{pe} q_p \quad (10)$$

where $\mathbf{Z}_{se}$ is the transfer function column vector from the secondary source to the microphones; $\mathbf{W}$ is the row vector composed of weighting factors from each error microphone; and $\mathbf{Z}_{pe}$ is the transfer function column vector from the primary source to the microphones.

Defining the estimated angular spectrum of the primary source and secondary source as $\hat{P}_p(\mathbf{k}, 0)$ and $\hat{P}_s(\mathbf{k}, 0)$, the discrepancy between the estimated and accurate value can be formulated as

$$\begin{aligned} \delta_p &= \frac{\hat{P}_p(\mathbf{k}, 0) - P_p(\mathbf{k}, 0)}{P_p(\mathbf{k}, 0)} \\ \delta_s &= \frac{\hat{P}_s(\mathbf{k}, 0) - P_s(\mathbf{k}, 0)}{P_s(\mathbf{k}, 0)} \end{aligned} \quad (11)$$

In the following investigations, $|\delta_p|$ and $|\delta_s|$ are defined as the angular spectrum estimation errors of the primary and secondary source, respectively. These errors are calculated using the logarithm base 10 and scaled by a factor of 20. Equation (10) can be written as

$$q_{s,3} = -q_p e^{-j\mathbf{k}(\mathbf{r}_s - \mathbf{r}_p)} \cdot \frac{1 + \delta_p}{1 + \delta_s} \quad (12)$$

The noise reduction at the far-field point $\mathbf{r}_0$ is

$$\text{NR} = 20 \lg \left| \frac{1 + \delta_s}{\delta_s - \delta_p} \right| \quad (13)$$

It is obvious that the noise reduction in the far field is affected by the angular spectrum estimation errors of both the primary and secondary sources. When the estimation errors of the primary and secondary sources are both zero, or the two estimation errors are exactly the same, perfect noise reduction can be achieved.

Although Equation (13) is based on the analytical solution of the angular spectrum of a point source in the free field, it can also be used to predict the noise reduction when the primary sound field is complicated, as long as the observation point is in the far field of target direction.

3. Simulations

In the following simulations, 200 Hz is chosen as the frequency of interest considering the line frequency of 50 Hz and the convenience of the noise reduction evaluation in the far field in experiments. The corresponding wavelength λ is 1.7 m.

3.1. Angular Spectrum of a Point Source

Figure 2 shows the estimated angular spectrum by integrating the sound pressure over the error plane with the finite size $L \times L$ and the analytical solution provided by Equation (4) for a primary source located at (0, 0.65, −0.6) m with a volume velocity of $1 \text{ m}^3/\text{s}$. The center of the integration domain is at the origin of the coordinate system. When the side length of the integration plane in Equation (3) is much smaller than the wavelength, the phases of the sound pressures in the plane are similar. Therefore, as the size of the integration plane increases, the magnitude of the integration increases at first, while the phase remains almost unchanged. As the size of the plane further increases, the phase distribution of the sound pressures in the plane becomes complicated, resulting in the oscillatory characteristics of the result. Nevertheless, both the magnitude and phase of the integration converge to the analytical solution, which demonstrates that the derived analytical solution of the angular spectrum in Equation (4) is correct.

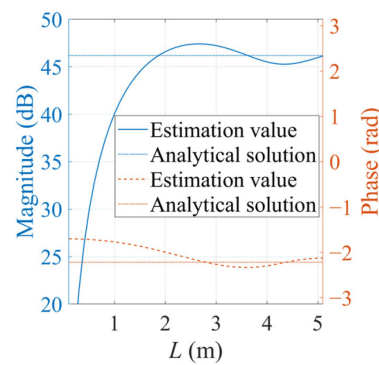


Figure 2. The angular spectrum estimated with Equation (3) and the analytical solution in Equation (4).

Figure 3 illustrates the variation in estimation errors with the error microphone spacing d , the location of source z_p , and the target direction θ , respectively, when Equation (8) is used. In Figure 3a, the estimation error of the angular spectrum is generally smaller with a smaller d , although sometimes the estimation error of $d = \lambda/2$ is smaller than that of $d = \lambda/3$ due to the finite size of the microphone array. In Figure 3b, the estimation error increases with the distance between the source and the microphone array when the microphone spacing is $\lambda/3$. For example, when $L = 1.7$ m, the estimation errors are -13.3 , -0.6 , and -0.2 dB when the distance z_p is -0.6 , -5.6 , and -10.6 m, respectively. As the source moves away from the array, the array appears to shrink because the proportion of the array that lies on the spherical surface centered at the source decreases. In Figure 3c, the overall angular spectrum estimation error increases with the target angle θ . The reason is that as θ increases, the projection of the array in the plane perpendicular to $\mathbf{k}$ decreases.

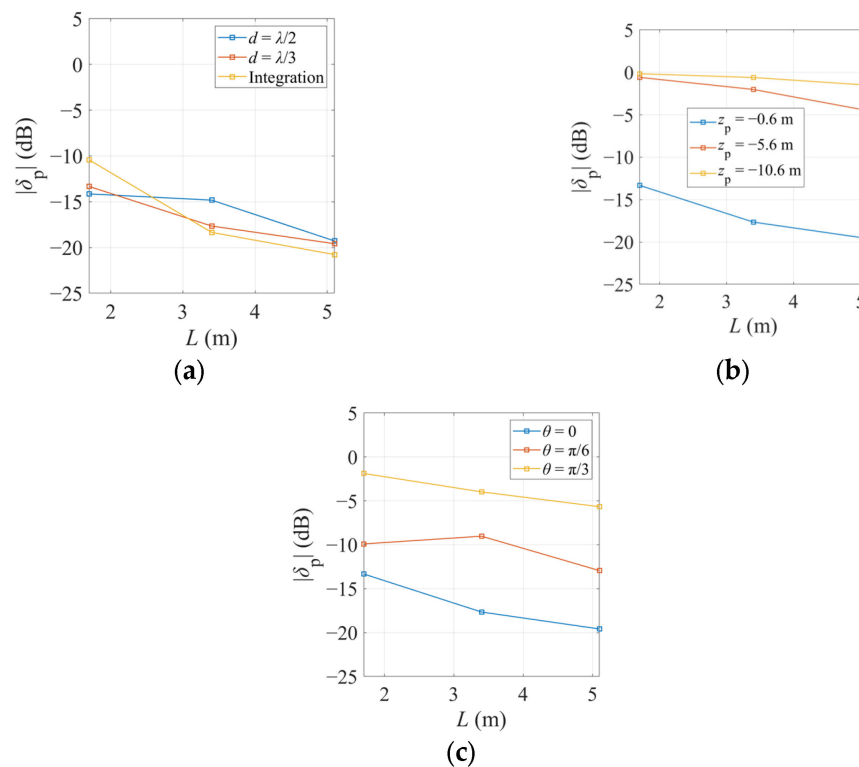


Figure 3. Variation in the angular spectrum estimation error (a) with the microphone spacing when $\mathbf{r}_p = (0, 0.65, -0.6)$ m and $\theta = 0$, (b) with the distance between the array and source when $x_p = 0$, $y_p = 0.65$ m, $d = \lambda/3$ and $\theta = 0$, and (c) with the target direction θ when $\mathbf{r}_p = (0, 0.65, -0.6)$ m, $d = \lambda/3$ and $\varphi = \pi/2$.

Table 1 lists the angular spectrum estimation error of the primary source located at $(0, 0.65, -0.6)$ m with different error microphone spacings when the size of error plane $L = 1.7$ m. When $d = \lambda/2$, smaller estimation errors can be obtained for both θ compared with the other two configurations; therefore, the microphone spacing is $\lambda/2$ and the size of the microphone array is chosen as $L = 1.7$ m in the following simulations.

Table 1. Angular spectrum estimation error of the primary source located at $(0, 0.65, -0.6)$ m when $L = 1.7$ m.

$ \delta_p $ (dB)	$\theta = 0$	$\theta = \pi/6$
Microphone array, $d = \lambda/2$	−14.2	−20.9
Microphone array, $d = \lambda/3$	−13.3	−11.6
Integration	−10.4	−13.7

3.2. Effect of the Secondary Source Location

It is seen from Equation (13) that to achieve effective noise reduction in the target direction, the angular spectrum estimation errors of the sources need to be as small as possible. This section aims to investigate the effect of the position of the secondary source on the angular spectrum estimation error, as well as the performance of the active directivity control system.

Figure 4 shows the schematic diagram of the simulation system. The secondary source is assumed to be placed at $z = -0.6$ m plane where the primary source is located. A $1.7 \text{ m} \times 1.7 \text{ m}$ error array with nine evenly spaced microphones distributed within the $z = 0$ plane is used to estimate the angular spectrum. The sound pressure level (SPL) reduction at a receiver which is 3 m away from the origin in the target direction is used to evaluate the noise reduction performance of the system. The evaluation plane is at the $x = 0$ plane.

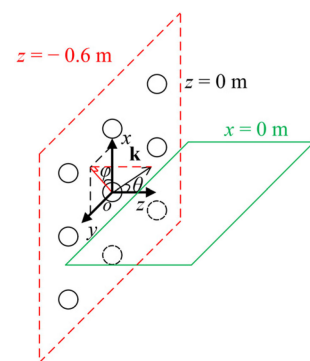


Figure 4. Schematic diagram of the simulation system. The red rectangle represents the plane where the sound sources are located, the black circles represent the error microphones, and the green rectangle represents the evaluation plane.

Figure 5a shows the angular spectrum estimation error when the sources are located at different positions in this plane and the target direction is $\theta = 0$. In this case, it can be seen from Equation (3) that the integrand is symmetric with respect to x and y . Moreover, it can be observed from Equation (4) that within the x - y plane, the angular spectra of two point sources with the same strength are equal. Therefore, the distribution of estimation errors is bivariate symmetric within the x - y plane. Additionally, since the integrand can be decomposed into the form of sine and cosine integrals, the estimation errors will exhibit oscillatory characteristics in the x - y plane. The reason that grid-like patterns appear in areas with smaller estimation errors is that the microphone array is distributed in a square. The primary source is located at $\mathbf{r}_p (0, 0.65, -0.6)$ m, and $|\delta_p|$ is -14.2 dB. When the secondary source is located at $\mathbf{r}_{s1} = (-0.42, 0.42, -0.6)$ m and $\mathbf{r}_{s2} = (0, -0.11, -0.6)$ m, $|\delta_s|$ is -10.8

and 1.6 dB, and the corresponding noise reductions are 22.1 dB and 4.5 dB, respectively. The smaller the estimation error of the secondary source is, the larger the noise reduction will be, as shown in Figure 5b.

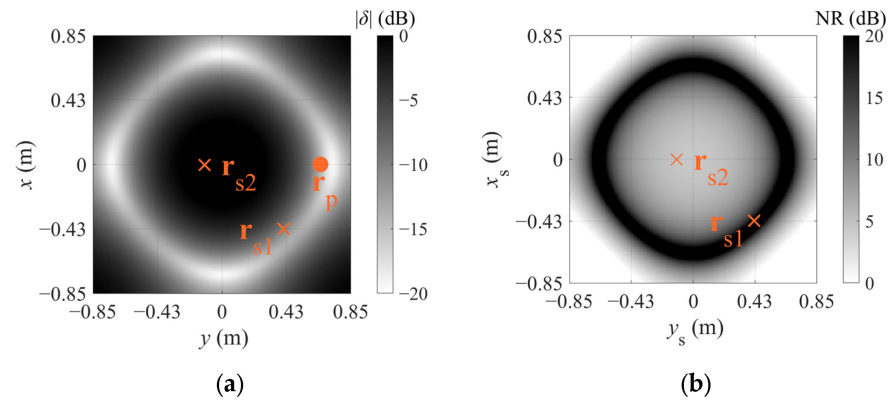


Figure 5. (a) The angular spectrum estimation error when the source is at different locations. (b) The NR at the receiver when the secondary source is located at different positions in the $z = -0.6$ m plane, and the target direction is $\theta = 0$.

Figure 6 shows the distribution of SPL and NR in the evaluation plane when the target direction is $\theta = 0$. When the secondary source is placed at r_{s1} , a more significant directional quiet zone from the near field to the far field in the target direction is achieved, as shown in Figure 6b,e. For instance, the average SPL at the 11 orange points spaced 0.3 m apart in Figure 6c is reduced by 19.8 dB and 1.2 dB, respectively, when the secondary source is located at positions r_{s1} and r_{s2} .

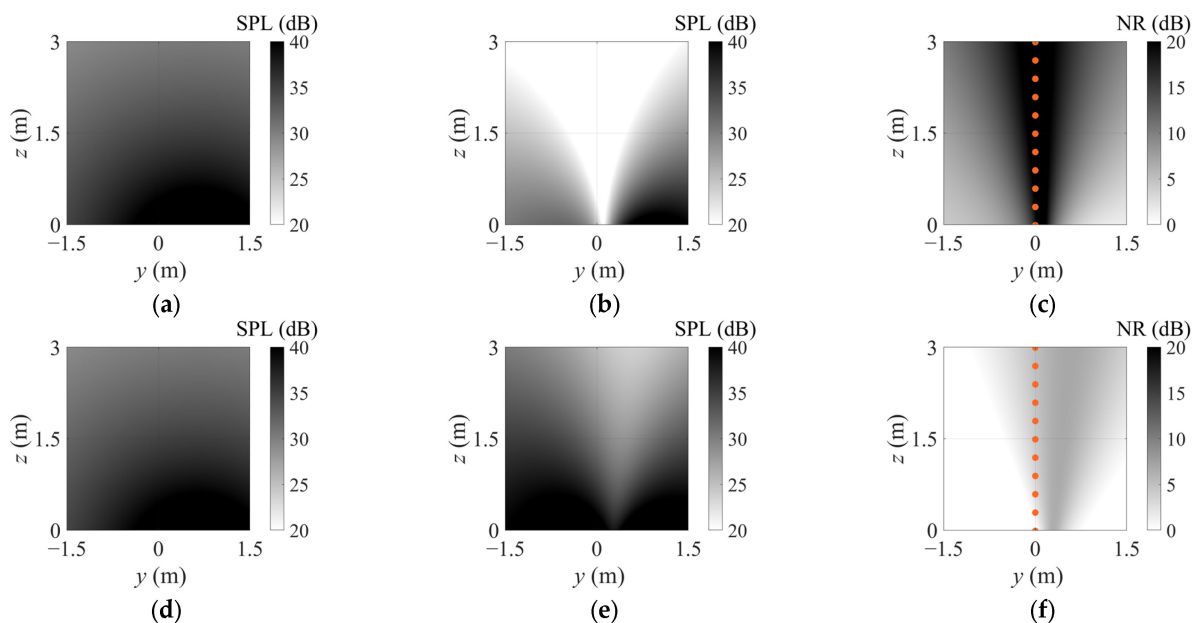


Figure 6. The distribution of SPL and NR when the target direction is $\theta = 0$ (a) SPL without control, (b) SPL with control and (c) NR when the secondary source located at r_{s1} , (d) SPL without control, (e) SPL with control and (f) NR when the secondary source located at r_{s2} .

Figure 7a shows the angular spectrum estimation errors when the target direction is chosen as $\theta = \pi/6$ and $\varphi = \pi/2$. The estimation errors of the secondary source are -3.9 dB and -19.7 dB when the secondary source is placed at r_{s1} and r_{s2} , respectively. The noise reduction is 5.5 dB with the secondary source placed at r_{s1} , while it is enhanced to 21.2 dB with the secondary source at r_{s2} , as shown in Figure 7b.

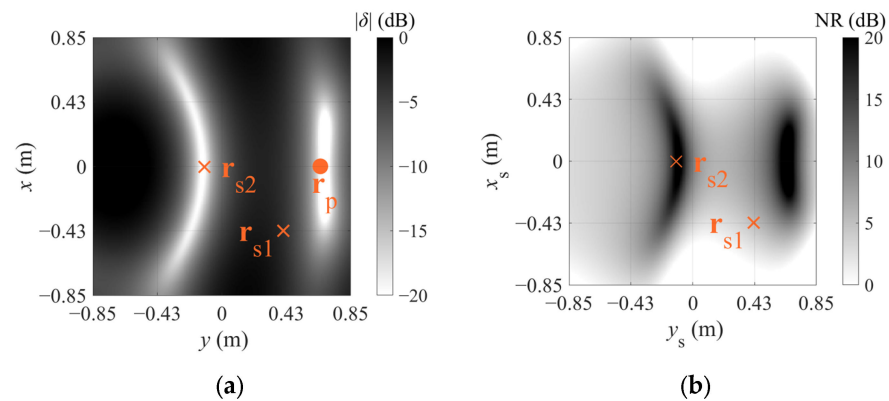


Figure 7. (a) The angular spectrum estimation error when the source is at different locations. (b) The NR at the receiver when the secondary source is located at different positions in the $z = -0.6$ m plane, and the target direction is $\theta = \pi/6$ and $\varphi = \pi/2$.

Figure 8 shows the distribution of SPL and NR when $\theta = \pi/6$ and $\varphi = \pi/2$, where better noise reduction in the target direction can be achieved by placing the secondary source at r_{s2} rather than r_{s1} . For instance, the average SPL at the 7 orange points spaced 0.3 m apart in Figure 8c,f is reduced by 5.4 dB and 10.6 dB, respectively, when the secondary source is located at positions r_{s1} and r_{s2} .

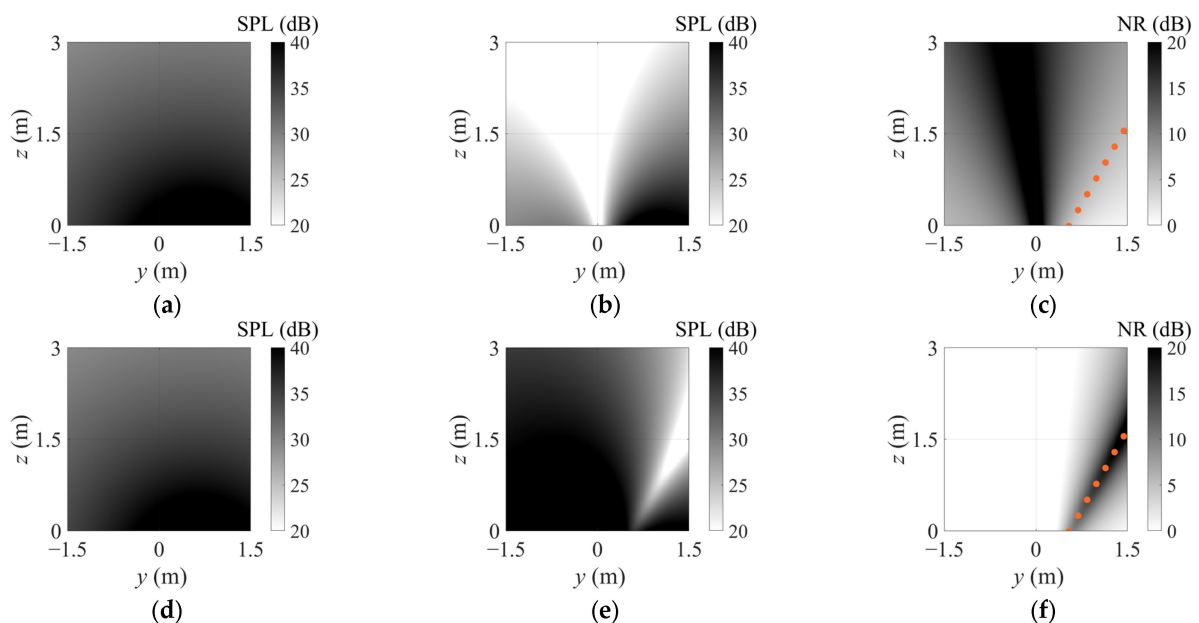


Figure 8. The distribution of SPL and NR when the target direction is $\theta = \pi/6$ and $\varphi = \pi/2$. (a) SPL without control, (b) SPL with control and (c) NR with the secondary source located at r_{s1} , (d) SPL without control, and (e) SPL with control and (f) NR with the secondary source located at r_{s2} .

4. Experiment

Figure 9 shows the experimental setup in the anechoic chamber in Nanjing University, and the system configuration is the same as that adopted in Figure 4. Nine microphones are evenly placed in the error plane, and 11 microphones are arranged in a line to measure the SPL distribution in the evaluation plane $x = 0$, with an interval of 0.3 m, which is approximately $1/6$ wavelength of 200 Hz. The type of all the microphones is Anty M1212 [22], and all the microphones were calibrated using a calibrator (Brüel & Kjær, type 4231).

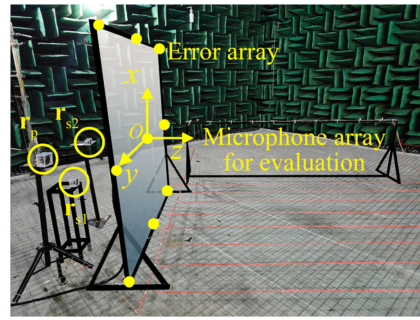


Figure 9. The experimental setup in the anechoic chamber. Yellow solid circles represent the 9 error microphones.

In the experiment, the sound pressures at the error microphones generated by the primary and secondary sources were measured using a multi-channel analyzer PULSE 3560D, and the transfer functions from the loudspeakers to all the microphones are obtained by recording the sound pressures at the microphones excited by the primary and secondary loudspeakers. The optimal input fed to the secondary loudspeaker is then calculated with Equation (10) and the weighting vector $\mathbf{W}$ is calculated with Equation (9). Therefore, the secondary sound field at evaluation microphones can be obtained. The total sound pressure level after control is then obtained by superimposing the primary and secondary sound fields, and the noise reduction is the difference between the primary sound pressure level and that after control. Finally, the voltage was fed to the secondary loudspeaker for active directivity noise control, and the noise reduction was obtained by measuring the sound pressure level in the evaluation plane before and after control.

Figure 10 shows the distribution of sound pressure levels and the corresponding NR in the evaluation plane when the target direction is chosen as $\theta = 0$. The average SPL at 11 orange points spaced 0.3 m apart in Figure 10c,f is reduced by 15.7 dB and 2.0 dB, respectively, when the secondary loudspeaker is located at $\mathbf{r}_{s1}$ and $\mathbf{r}_{s2}$.

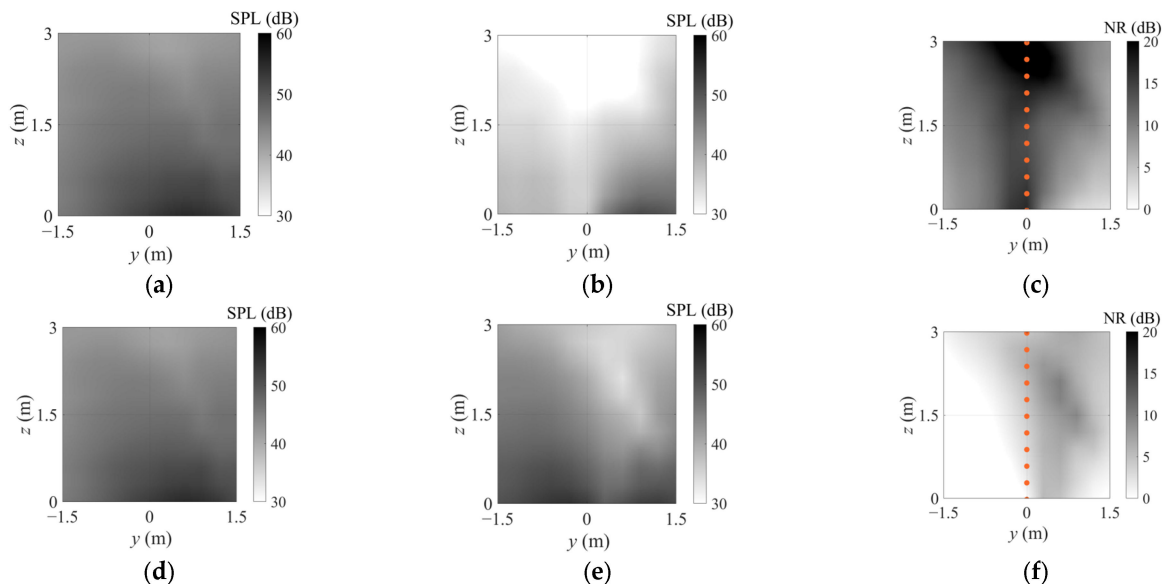


Figure 10. The distribution of measured SPL and NR when the target direction is $\theta = 0$. (a) Primary SPL, (b) the SPL with control, (c) NR with the secondary loudspeaker at $\mathbf{r}_{s1}$, (d) primary SPL, (e) the SPL with control, and (f) NR with the secondary loudspeaker at $\mathbf{r}_{s2}$.

Table 2 shows the NR of the average SPL at the 11 orange points when the target direction is $\theta = 0$. The difference between the noise reduction in the point source model simulation and in the offline simulation is within 2.3 dB. The noise reduction with the

secondary source at $\mathbf{r}_{s1}$ is significantly larger than that with the secondary source at $\mathbf{r}_{s2}$, because the estimation error of the angular spectrum is smaller when the secondary source is placed at $\mathbf{r}_{s1}$.

Table 2. NR in simulations and experiment when the target direction is $\theta = 0$.

Position of the Secondary Source	Noise Reduction (dB)		
	Point Source Model Simulation	Experiment	
		Offline Simulation	Measurement
$\mathbf{r}_{s1}$	19.8	17.5	15.7
$\mathbf{r}_{s2}$	1.2	2.1	2.0

Figure 11 compares the measured noise reduction when the target direction is chosen as $\theta = \pi/6$ and $\varphi = \pi/2$. As predicted in Figure 7b, more effective noise reduction in the target direction is achieved by placing the secondary loudspeaker at $\mathbf{r}_{s2}$ because of the smaller angular spectrum estimation error of the secondary source.

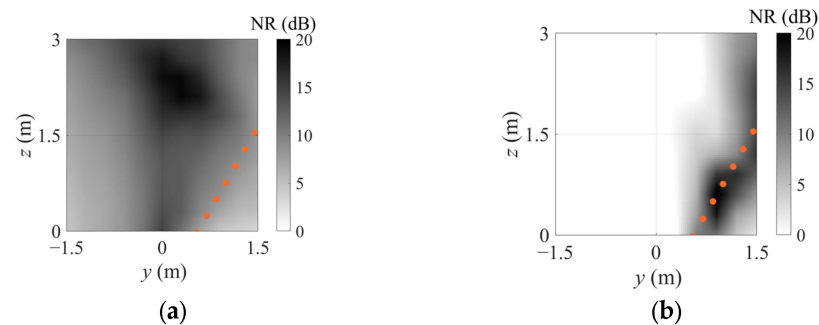


Figure 11. The distribution of measured NR in the evaluation plane when the target direction is $\theta = \pi/6$ and $\varphi = \pi/2$ with the secondary loudspeaker at (a) $\mathbf{r}_{s1}$, (b) $\mathbf{r}_{s2}$.

Table 3 shows the NR of the average SPL at the seven orange points in Figure 11, when the target direction is $\theta = \pi/6$ and $\varphi = \pi/2$. The difference between the point source model simulation and the offline loudspeaker simulation results is within 1.9 dB.

Table 3. NR in simulations and experiment when the target direction is $\theta = \pi/6$ and $\varphi = \pi/2$.

Position of the Secondary Source	Noise Reduction (dB)		
	Point Source Model Simulation	Experiment	
		Offline Simulation	Measurement
$\mathbf{r}_{s1}$	5.4	7.3	7.0
$\mathbf{r}_{s2}$	10.6	12.3	10.5

The reason for the discrepancy between the point source simulation and the offline loudspeaker simulation is that the loudspeaker adopted in the experiment cannot be modelled as an ideal point sound source. In a point source model, the sound field of a point source is omnidirectional as shown in Figure 6a. However, the measured sound field of a loudspeaker is not omnidirectional, as shown in Figure 10a. In addition, when using the evaluation microphone array for sound field scanning, the positions of the microphones may also deviate from those in the point source simulation. The difference between the offline simulation and experimental results is within 1.8 dB. This difference could be caused by measurement error or the deviation of the evaluation point positions in the process of scanning the sound fields.

5. Conclusions

The principle of a recently proposed error-sensing strategy for active directivity noise control is discussed in this paper. It is revealed that this error-sensing strategy equals a reduction in the angular spectrum in the target direction. Therefore, the noise reduction performance is closely related to the accuracy of the angular spectrum estimation of both the primary source and the secondary sources. Experiment results in the anechoic chamber investigated the impact of the location of the secondary source. In the future, the effect of a reflective surface on the performance of the active directivity control system adopting the near-field error-sensing strategy will be investigated.

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