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Industrial Noise Source Identification by Using an Acoustic Beamforming System

A noise source identification technique is proposed for industrial applications by using a microphone array and beamforming algorithms. Both of the directions and the distances of long-range noise sources are calculated. The conventional method, the minimum variance (MV) method, and the multiple signal classification (MUSIC) method are the main beamforming algorithms employed in this study. The results of numerical simulations and field tests indicate the effectiveness of the acoustic beamformer in identifying noise sources in industrial environments.

Introduction

Increasing concern with occupational and environmental problems caused by industrial noises has been seen in recent years. The industrial noises have significant effects on labor's health and community living quality. It is highly desired to develop methods that are capable of locating noise sources in an accurate and systematic manner before any noise control measure can be applied. Conventional ways of noise source identifications include, for example, sound pressure measurement (Kinsler et al., 1982), sound intensity measurement (Tichy, 1984; and Fahy, 1989), acoustic holography (Maynard et al., 1985; Bai, 1992; and Bai 1995), genetic algorithm technique (Hamada et al., 1995), and many others. These methods suffer from the drawbacks of being either inaccurate or being restrictive in only small areas or short distances when applied to complex noise fields in industrial environment.

In this study, a noise source identification technique is proposed for industrial applications by using a linear microphone array in conjunction with beamforming algorithms. The acoustic beamforming system provides useful information concerning the noise sources such as the number of sources, directions, distances, and level of strength. In comparison with radar systems, the array systems have the advantages that they generally give higher signal-to-noise ratios (SNR) and more flexible apertures. The proposed method is useful in identifying noise sources in the farfield of the array. The need for identifying long-range sources is frequently encountered in industrial and environmental applications.

Array beamforming techniques have been widely used in the areas such as signal processing of radio waves, seismic waves, and underwater acoustic waves. Excellent references regarding the fundamental aspects of array signal processing can be found in the review papers by Marr (1986), Haykin et al. (1992), and the text by Burdic (1991). Since a vast amount of literature can be found on the applications of array beamformers, it is impossible to review everything except for the previous work relevant to this research. Capon (1969) employed an array to calculate the center of earthquake and propagation of seismic waves. Shan et al. (1985a and 1985b) proposed a spatial smoothing technique to resolve the multipath problem in narrowband beamforming. Chen and Yu (1991) developed another method to deal with the multipath problem for broadband beamforming. Reddi (1979) formulated an eigenvector method to attack the direction of arrival (DOA) problems for plane waves. Schmidt (1986) proposed a multiple signal classification (MUSIC) algorithm that is essentially an eigenvalues-based approach to significantly improve the resolution of multiple sources. Chen et al. (1992) calculated the spherical angles of sources by using a two-dimensional array. Chiang and Lee (1993) utilized beamforming methods to calculate DOAs by using hydrophones in shallow-water environment. In addition to the aforementioned methods for plane waves, Owsley et al. (1982) developed a beamforming algorithm for spherical waves.

This study focuses on developing an acoustic beamforming system for identification of industrial noise sources. The hardware system consists mainly of a linear microphone array and a multi-channel data acquisition system. The beamforming algorithms employed in this research are the conventional method, the minimum variance (MV) method, and the MUSIC method. Both plane-wave sources and spherical-wave sources are investigated. The methods for plane waves give only directions of noise sources, whereas the methods for spherical waves give both directions and distances of noise sources by using a delay-and-sum concept. In addition, this study deals with the multipath problem that frequently arises in practice. Numerical simulations and field tests are carried out to justify the developed techniques.

Acoustic Beamforming Algorithms

First, a brief review of array signal processing is given. Consider two microphones exposed to a plane-wave field from the direction with an angle θ measured from the x-axis, as shown in Fig. 1. If microphone 1 is taken as the reference sensor, the propagation delay sensed by the microphone 2 is

$$\tau = \frac{-d \cos \theta}{c}, \qquad (1)$$

where d is the spacing between the two sensors and c is the speed of sound. Let the signal received by microphones 1 and 2 be $x_1(t)$ and $x_2(t)$ respectively. Then, the latter signal is merely the delayed version of the former, i.e.,

$$x_2(t) = x_1(t - \tau). \qquad (2)$$

For pure tones or narrowband plane waves, Eq. (2) becomes

$$x_2(t) = x_1(t) \exp\left(j\omega \frac{d \cos \theta}{c} \right)$$

$$= x_1(t) \exp\left(j \frac{2\pi d \cos \theta}{\lambda} \right), \qquad (3)$$

Contributed by the Technical Committee on Vibration and Sound for publication in the JOURNAL OF VIBRATION AND ACOUSTICS. Manuscript received April 1995. Associate Technical Editor: C. Fuller.

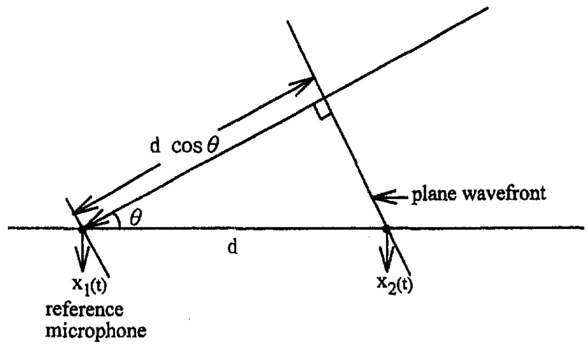

Fig. 1 A two-microphone array receiving plane-wave signals

where ω is the center frequency of the narrowband signal and λ is the corresponding wave length. The expression in Eq. (3) can be easily extended to an array containing M equally spaced microphones

$$x_m(t) = x_1(t)\exp[j(m-1)v(\theta)d],\quad m = 1, 2, \ldots, M \quad (4)$$

where

$$v(\theta) \triangleq \frac{2\pi\cos\theta}{\lambda}.$$

In practice, the signals received by the microphones may be corrupted by noises. To account for this effect, Eq. (4) is modified into

$$x_m(t) = u(t)\exp[j(m-1)v(\theta)d] + n_m(t),\quad m = 1, 2, \ldots, M \quad (5)$$

where $u(t)$ and $n_m(t)$ denote, respectively, the time-modulated part of the signal and a uncorrelated zero-mean Gaussian white noise term. This expression can be extended to the case in which P plane-wave sources are present in the field. In matrix notations,

$$\mathbf{x}(t) = \mathbf{A}(\theta)\mathbf{u}(t) + \mathbf{n}(t), \quad (6)$$

where $\mathbf{x}(t) = [x_1(t)x_1(t)\ldots x_M(t)]^T$ being the data vector containing the signals received by the M microphones, $\mathbf{u}(t) = [u_1(t)u_2(t)\ldots u_P(t)]^T$ being the vector containing the time-modulated parts of the plane-wave signals, $\mathbf{n}(t) = [n_1(t)n_2(t)\ldots n_M(t)]^T$ being the vector containing the Gaussian white noises, and $\mathbf{A}(\theta) = [\mathbf{a}(\theta_1)\mathbf{a}(\theta_2)\ldots\mathbf{a}(\theta_P)]^T$ being the DOA matrix with the vectors $\mathbf{a}(\theta_i)$, $i = 1, 2, \ldots, P$ defined as

$$\mathbf{a}(\theta_i) \triangleq [1\ \exp(jv_i d)\ldots\exp[jv_i(M-1)d]]^T. \quad (7)$$

In the beamforming algorithms, the following covariance matrix is required

$$\mathbf{R}_{xx} = E[\mathbf{x}(t)\mathbf{x}^H(t)], \quad (8)$$

where $E[\cdot]$ denote the expected value and $\mathbf{x}^H$ is the hermitian transpose of $\mathbf{x}$. With some manipulations, it can be shown that

$$\mathbf{R}_{xx} = \mathbf{A}(\theta)\mathbf{R}_{uu}\mathbf{A}^H(\theta) + \mathbf{R}_{nn}. \quad (9)$$

For sampled-data systems, the covariance matrix can be estimated by

$$\hat{\mathbf{R}}_{xx} \triangleq \frac{1}{N}\sum_{l=1}^{N}\mathbf{x}(lT)\mathbf{x}^H(lT), \quad (10)$$

where T is the sampling period.

It should be noted that the above equations are essentially single-frequency formulations. For broadband noises, however, the same concept is still applicable except that the center frequency of the band must be used to compute the DOA matrix in Eq. (6). This method is detailed in the paper by Chen and Yu (1991).

Although the formulations up to this point are valid for only plane waves, with some modifications, a straightforward extension can be made to spherical waves. The major difference lies in how one calculates the time delays. For spherical waves shown in Fig. 2, the time delay associated with the mth microphone can be expressed as

$$\tau_m = -\frac{1}{c}\sqrt{r^2 + [(m-1)d]^2 - 2r(m-1)d\cos\theta},\quad m = 1, 2, \ldots, M \quad (11)$$

where r and θ are, respectively, the distance and the spanning angle of the point source with respect to the first microphone. It should be noted that in the following beamforming algorithms only the time delays of the spherical waves are compensated, but the amplitudes are not. Unlike the plane-wave scheme that produces only the DOA information, the spherical-wave scheme provides an additional distance information.

In array signal processing, data in both the temporal domain and the spatial domain are sampled. The resolutions and apertures in the two domains should be adequate such that aliasing and leakage would not occur (John and Dudgeon, 1993). In general, the spacing between two microphones in an array is selected to be one half of the wave length to ensure a good compromise between the resolution and the aperture for a limited number of sensors.

To keep the presentation in a reasonable length, three beamforming algorithms employed in this study are briefly reviewed. These algorithms are the conventional method, the MV method, and the MUSIC method. Assume that M microphones are employed in an array. The *steering vector* at the look angle θ is defined as (Pillai, 1988)

$$\mathbf{a}(\theta) \triangleq \frac{1}{\sqrt{M}}[\exp(jd_1v)\exp(jd_2v)\ldots\exp(jd_Mv)]^T \quad (12)$$

where $v \triangleq 2\pi\cos\theta/\lambda$ and d_m, $m = 1, 2, \ldots, M$ is the distance from the mth microphone to the first microphone. The major step of the conventional method consists of delaying and summing the outputs from each microphone to yield the total output

$$y(t) = \mathbf{a}^H(\theta)\mathbf{x}(t). \quad (13)$$

It follows that the power output of the array is

$$P_B \triangleq E[|y(t)|^2] = E[\mathbf{a}^H(\theta)\mathbf{x}(t)\mathbf{x}^H(t)\mathbf{a}(\theta)] = \mathbf{a}^H(\theta)E[\mathbf{x}(t)\mathbf{x}^H(t)]\mathbf{a}(\theta) = \mathbf{a}^H(\theta)\mathbf{R}_{xx}\mathbf{a}(\theta). \quad (14)$$

Despite the simplicity, the conventional method suffers from resolution problems when the noise sources are too close to each other. In contrast to time-domain signal processing, the main problem with array signal processing stems from the scarcity of spatial-domain data because of limited number of sen-

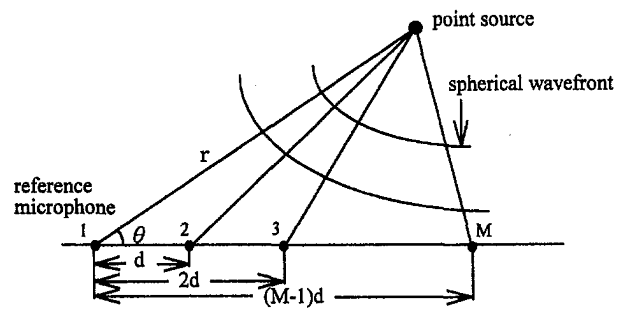

Fig. 2 An *M*-microphone array receiving spherical waves from a point source

sors. This calls for the need of the following two high-resolution algorithms.

The MV method was originally proposed by Capon (1969) who conducted a frequency-wavenumber analysis on earthquake data. The underlying principle of the method amounts to finding an optimal steering vector $\mathbf{w}_{opt}$ such that the array output power is minimized while maintaining the gain along the look direction to be constant, say unity. That is,

$$\min_{\mathbf{w}} \mathbf{w}^H \mathbf{R}_{xx} \mathbf{w}$$

$$s.t.\ |\mathbf{w}^H \mathbf{a}(\theta)| = 1. \tag{15}$$

By the method of Lagrange's multiplier, it can be shown that

$$\mathbf{w}_{opt} = \frac{\mathbf{R}_{xx}^{-1}\mathbf{a}(\theta)}{\mathbf{a}^H(\theta)\mathbf{R}_{xx}^{-1}\mathbf{a}(\theta)} \tag{16}$$

and the corresponding array output power

$$P_{MV} = \frac{1}{\mathbf{a}^H(\theta)\mathbf{R}_{xx}^{-1}\mathbf{a}(\theta)}. \tag{17}$$

Care has to be taken that the covariance matrix $\mathbf{R}_{xx}$ may become singular when the number of microphones is less than the number of sources. Otherwise, the MV method should work because the covariance matrix is always positive definite and invertible.

The third beamforming algorithm employed in this study is the MUSIC method. Consider a linear array consisting of M identical sensors and receiving signals from K narrowband plane-wave sources incoherent to each other. It is assumed that signals and noises are stationary, zero mean, and uncorrelated random processes. Suppose that the noises are uncorrelated but have the same variance σ^2. Hence, Eq. (9) becomes

$$\mathbf{R}_{xx} = \mathbf{A}(\theta)\mathbf{R}_{uu}\mathbf{A}^H(\theta) + \sigma^2\mathbf{I}. \tag{18}$$

It can be shown that the non-negative definite matrix $\mathbf{R}_{uu}$ is of rank K. Let $\mu_1 \geq \mu_2 \geq \ldots \geq \mu_K$ denote its nonzero eigenvalues. Then the eigenvalues of the array data covariance matrix $\mathbf{R}_{xx}$ are

$$\lambda_m = \begin{cases} \mu_m + \sigma^2 & m = 1, 2, \ldots, K \\ \sigma^2 & m = K+1, K+2, \ldots, M \end{cases}. \tag{19}$$

Let $\mathbf{b}_m$, $m = 1, 2, \ldots, M$ be the associated eigenvectors. Then,

$$\mathbf{R}_{xx} = \sum_{m=1}^{M} \lambda_m \mathbf{b}_m \mathbf{b}_m^H. \tag{20}$$

With some manipulations, it can be proved that (Pillai, 1989)

$$\mathbf{b}_m^H \mathbf{a}(\theta_k) = 0, \quad K+1 \leq m \leq M, 1 \leq k \leq K, \tag{21}$$

where θ_k is the arrival angle of the kth source. In addition, the *signal* subspace S and the *noise* subspace N are mutually orthogonal complements:

$$S = N^{\perp}, \tag{22}$$

where

$$S = span\{\mathbf{a}(\theta_1), \mathbf{a}(\theta_2), \ldots, \mathbf{a}(\theta_K)\}$$

and

$$N = span\{\mathbf{b}_{K+1}, \mathbf{b}_{K+2}, \ldots, \mathbf{b}_M\}.$$

As a consequence, the peaks of the function

$$P_{MUSIC} = \frac{1}{\sum_{m=K+1}^{M} |\mathbf{b}_m^H \mathbf{a}(\theta)|^2} \tag{23}$$

yields the DOA of the signals. For Eq. (23) to be meaningful, the condition that $M \geq K + 1$ has to be satisfied. This restricts the minimum number of required sensors to be greater than the total number of sources. The initial guess on the source number K then becomes important. Note also that, unlike the methods mentioned earlier, P_{MUSIC} does not physically correspond to the signal power.

A comparison of the aforementioned beamforming algorithms is in order. Although the conventional method is computationally the simplest, it generally produces the poorest resolution of multiple sources among the three methods. The MV method gives improved resolution than the conventional method. The MUSIC method gives the best resolution among the three methods at the expense of increased computational complexity. When a multipath problem is present or the number of microphones is not greater than the number of sources, the last two methods will break down because of the singular covariance matrix. The multipath problem generally arises due to reflections from the walls or obstacles in a closed space. In this case, array signals will become coherent to render a singular covariance matrix. To alleviate the problem, a spatial smoothing procedure can be used in conjunction with the MUSIC method to improve resolution of multiple sources for plane-wave cases (Shan, 1985b). This procedure involves dividing the original array into subarrays to scramble the coherence among the multipath signals. In this method, the number of subarrays must be greater than the number of sources.

Numerical Simulations and Field Tests

Numerical simulations and experimental tests were designed to evaluate the performance of the acoustic beamforming sys-

Table 1 Summary of numerical simulation cases

case number	noise source	array number	microphone spacing
1	plane waves: 800Hz at 45°+ 750Hz at 50°	6	0.2 m
2	plane waves: 800Hz at 45°+ 750Hz at 50°	12	0.2 m
3	plane waves: 800Hz at 45° +750Hz at 50°	6	0.288 m
4	plane waves: 800Hz at 45°+ 750Hz at 50°	6	0.425 m
5	plane waves: 800Hz at 30°+ 800Hz at 110°	6	0.2 m
6*	plane waves: 800Hz at 30°+ 800Hz at 110°	6	0.2 m
7	plane waves: 463~1100Hz+ 1273~1974Hz	6	0.2 m
8	spherical waves: 800 Hz at (10m, 10m)	6	0.2 m
9	spherical waves: 800 Hz at (10m, 10m)	6	0.425 m

* : Spatial smoothing was applied.

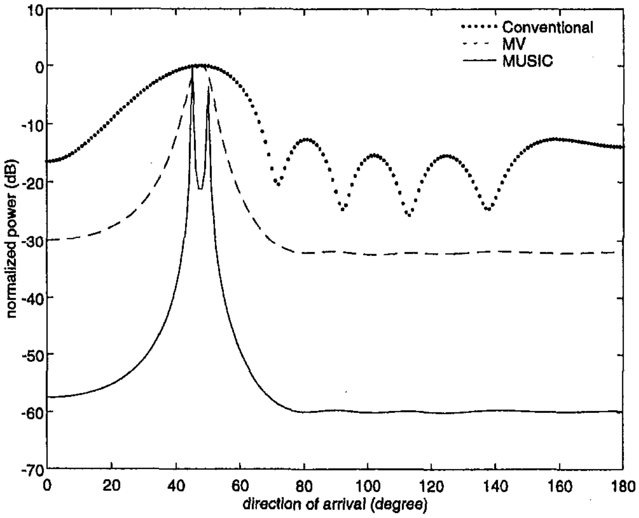

Fig. 3 Simulation result of Case 1 in Table 1 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.2 m.

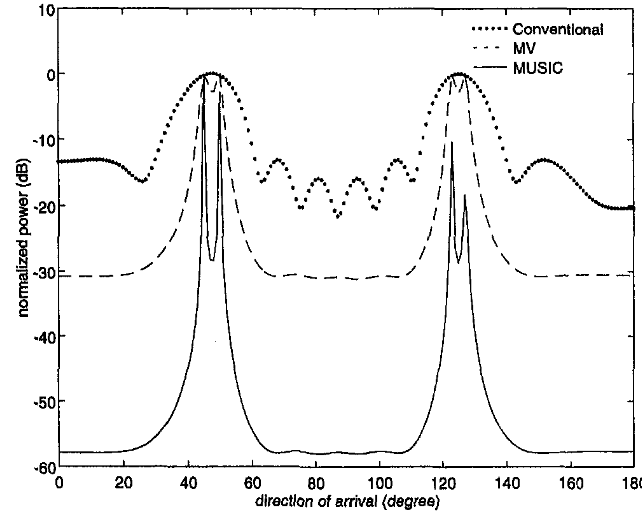

Fig. 5 Simulation result of Case 3 in Table 1 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.288 m. This case is intended to see the effect of microphone spacing.

tem. The conventional method, the MV method, and the MUSIC method were employed for identifying plane-wave sources and spherical-wave sources.

The simulation cases for plane waves and spherical waves were summarized in Table 1. The associated parameters of sources and arrays are also included. The first six cases are concerned with plane-wave beamforming schemes. In these cases, appropriate sampling rates were selected in accordance with the sampling theorem, where the sampling rates are at least twice as large as the bandwidths of the signals in order for the original signal to be faithfully reconstructed (Johnson and Dudgeon, 1993). On the other hand, because the MUSIC method does not produce real signal power as the other two methods, the following beamforming results are normalized with respect to their maximum value to facilitate the comparison.

In Case 1, an array consisting of six microphones was used to determine the directions of two plane-wave sources at θ = 45 deg and 50 deg, respectively. The microphones were equally spaced and the spacing was 0.2 m that corresponded to approximately one half of the wave length for the 800 Hz source. The result of Fig. 3 indicated that the MUSIC method produced significantly better resolution than the other two methods. In addition, sidelobes were clearly seen in the result of the conventional method.

In Case 2, the number of microphone was doubled, while the spacing remained unchanged. It can be seen in Fig. 4 that the MV method gave an improved result than the first case because of the increased size of array aperture. On the other hand, the result of the conventional method contained more but smaller sidelobes because of the larger aperture.

In Case 3, the same conditions as in Case 1 were retained except that the spacing was increased to 0.288 m. Then, aliased peaks appeared in the result of Fig. 5. However, the resolution had been improved because the effective size of the array aperture was doubled. If the number of array microphones was fixed, we could determine which peaks were aliased components by further increasing the spacing. This was illustrated in Case 4, where the spacing had been increased to 0.425 m. As can be seen in the result of Fig. 6, the peaks at 45 deg and 50 deg

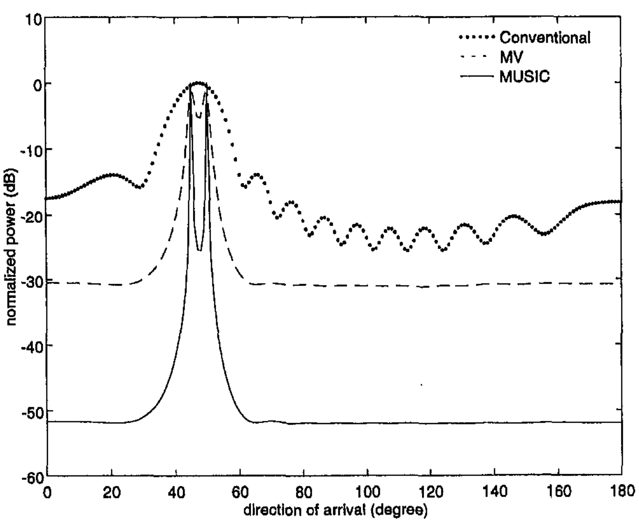

Fig. 4 Simulation result of Case 2 in Table 1 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 12 microphones with a spacing 0.2 m. This case is intended to see the effect of number of microphones.

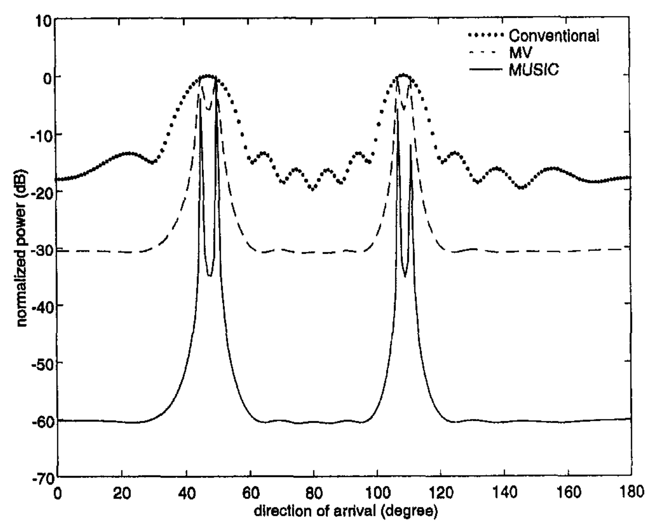

Fig. 6 Simulation result of Case 4 in Table 1 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.425 m. This case is intended to see the effect of aliasing.

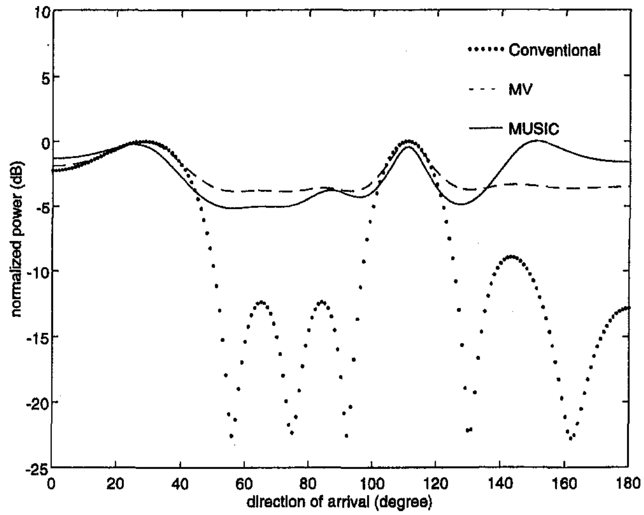

Fig. 7 Simulation result of Case 5 in Table 1 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.2 m. This case is intended to see the effect of multipath problem.

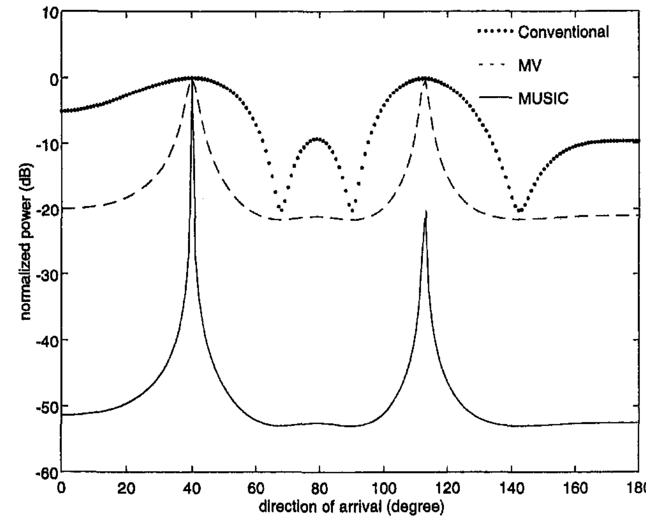

Fig. 9 Simulation result of Case 7 in Table 1 obtained by the broadband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.2 m.

remained unchanged, whereas the peaks at 108 deg and 113 deg were changed. Hence, we were able to tell immediately the latter two peaks were aliased components. In other words, the rule of half wave length can sometimes be relaxed to get a sufficiently large aperture if the aliasing problem is moderate.

In Case 5, the effect of coherent signals due to multipaths was investigated. Two coherent plane waves of the same frequency were assume to arrive at the array from 30 deg and 110 deg. The performance of the MV method and the MUSIC method was significantly degraded (Fig. 7). The MUSIC method gave even incorrect number of peaks and associated DOAs. As mentioned earlier in the last section, this was due to the fact that the covariance matrix required in the computation of the MV method and the MUSIC method had become singular for the coherent signals.

To alleviate the multipath problem, the spatial smoothing procedure was incorporated into the MUSIC technique (Case 6). The result was shown in Fig. 8. It was found that the spatial smoothing procedure indeed eliminated the multipath problem. Large number of subarrays will generally yield more satisfactory results than small number of subarrays under the condition that the size of subarray aperture is sufficiently large and the number of subarrays must be greater than the number of sources.

In comparison with the narrowband cases, broadband signals were more difficult to deal with in beamforming calculations. In Case 7, two broadband plane-waves from $\theta = 40$ deg and 110 deg (bandwidths = 637 and 701 Hz respectively) were investigated. In Fig. 9, small errors (approximately 3 deg) occurred in finding the DOA of the source at $\theta = 110°$.

Next two cases were concerned with the spherical-wave beamforming. The MUSIC technique was employed in these two cases because it generally produced the best resolution among three methods. Suppose that a point source of 800 Hz was located at the coordinate (10 m, 10 m). In Case 8, we attempted to calculate the location of the source by using the MUSIC method based on a 0.2 m spacing. For convenience, the result was shown in a contour plot (Fig. 10). It appeared that the calculated DOA was correct, while the distance of the source was hard to distinguish. Although this problem can be solved by increasing the number of microphones, it is more

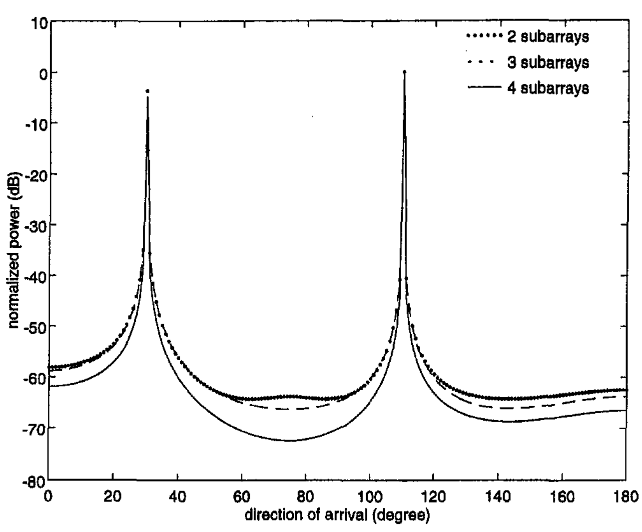

Fig. 8 Simulation result of Case 6 in Table 1 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.2 m. This case is intended to see the effect of spatial smoothing.

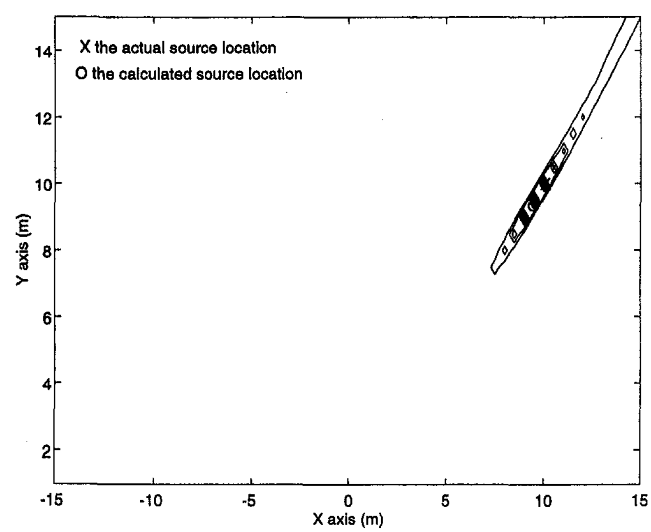

Fig. 10 Simulation result of Case 8 in Table 1 obtained by the broadband spherical-wave MUSIC algorithm. The array consists of 6 microphones with a spacing 0.2 m.

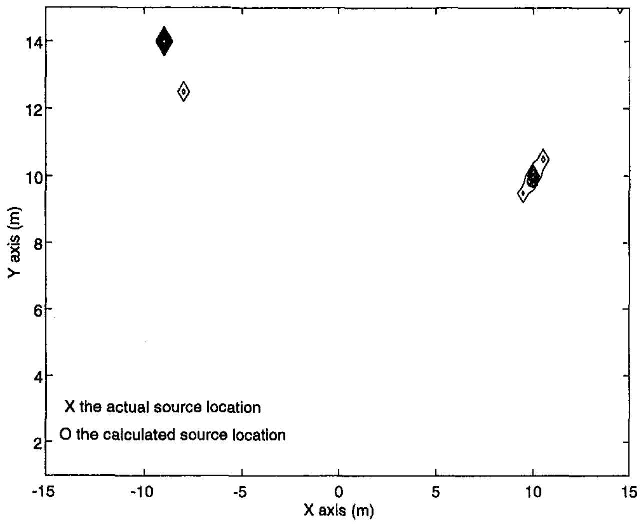

Fig. 11 Simulation result of Case 9 in Table 1 obtained by the broadband spherical-wave MUSIC algorithm. The array consists of 6 microphones with a spacing 0.425 m.

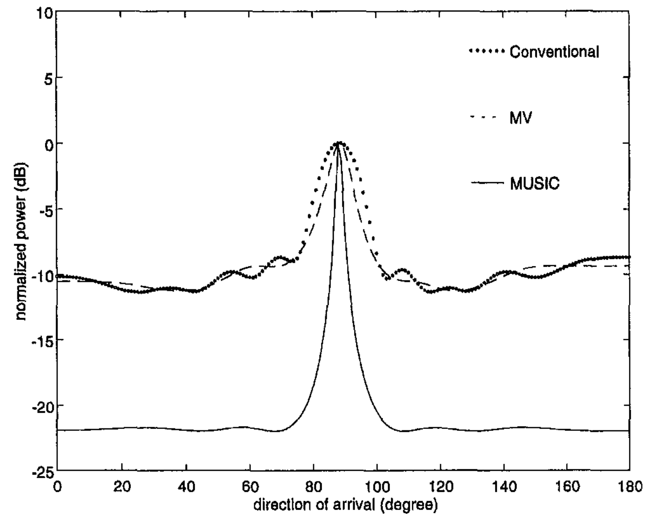

Fig. 13 Experimental result of Case 1 in Table 2 obtained by the narrowband plane-wave beamforming algorithms. The array consists of 6 microphones with a spacing 0.05 m.

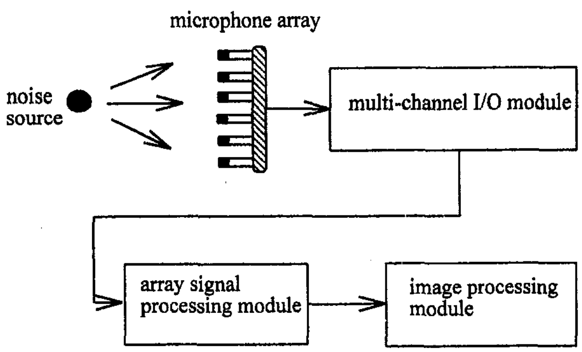

Fig. 12 Block diagram of the acoustic beamforming system

practical to increase the spacing between microphones if the number of microphones is fixed and the source of interest is located very far away. Hence, the spacing was changed to 0.425 m in Case 9. The result shown in Fig. 11 indicated that two sources were located at (10 m, 10 m) and (−9 m, 14 m), where the latter was apparently due to aliasing. Thus, increasing spacing between microphones indeed significantly improved the accuracy in distance determination.

The remaining part of the section is focused on the application of the acoustic beamforming system to the identification of industrial noise sources. To evaluate the practicality of the beamforming techniques, five experimental cases were selected for the field tests. The experimental setup was schematically shown in the block diagram of Fig. 12. The beamforming system mainly included six microphones, a multi-channel I/O module (with a maximum sampling rate up to 33 kHz per channel), an array signal processing computer, and an image processing module.

Table 2 summarized the experimental cases with the associated conditions of the sources and the arrays. These field tests were performed in a petro-chemical plant in northern Taiwan. In Cases 1–3, the beamforming system was placed on an open ground outside a cyclohexanone-oxime factory. A preliminary analysis of the noise data received by the array revealed that the noise was a narrowband signal centered at approximately

Table 2 Summary of experimental field tests

case number	noise source	array number	microphone spacing
1	narrowband, out-door, at (r, θ) = (27m, 85°)	6	0.05 m
2	narrowband, out-door, at (x,y) = (2.4m, 26.9m)	6	0.05 m
3	narrowband, out-door, at (x,y) = (2.4m, 26.9m)	6	0.25 m
4	broadband, out-door, at (r, θ) = (29.4m, 88°)	6	0.05 m
5	broadband, out-door, at (x,y) = (1m, 29.4m)	6	0.2 m

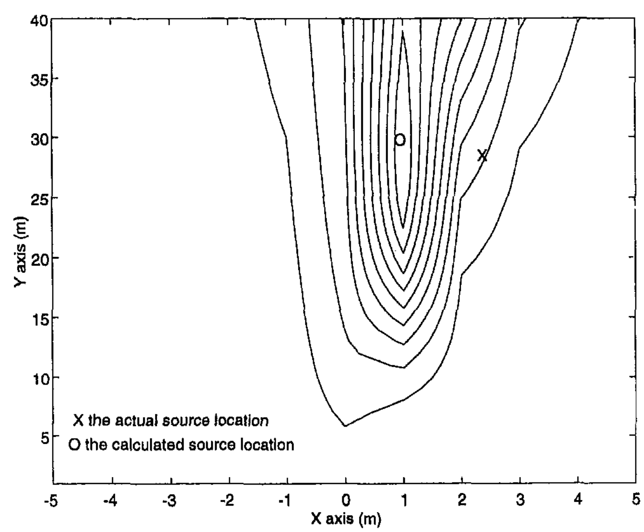

Fig. 14 Experimental result of Case 2 in Table 2 obtained by the narrowband spherical-wave MUSIC algorithm. The array consists of 6 microphones with a spacing 0.05 m.

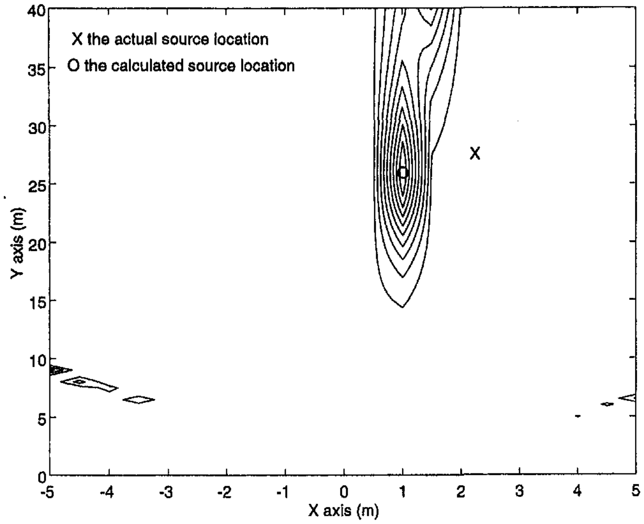

Fig. 15 Experimental result of Case 3 in Table 2 obtained by the narrowband spherical-wave MUSIC algorithm. The array consists of 6 microphones with a spacing 0.25 m.

2.5 kHz corresponding to the wave length 0.138 m. In Case 1, the narrowband plane-wave beamforming algorithms, based on 0.05 m spacing, were employed to determine the DOA of the source. The error between the calculated result and the true direction of the source was found to be within 2.8 deg (Fig. 13). Again, the MUSIC method produced the most superior resolution among three algorithms.

Although the plane-wave beamforming methods yielded accurate DOA information of the noise source, the spherical-wave methods had to be employed for calculating the distance of the source. In Case 2, the spherical-wave MUSIC algorithm based on 0.05 m spacing was used to process the noise data under the same conditions as in Case 1. The result was shown in Fig. 14. The contour plot roughly indicated the location of the noise source with large spreading. To reduce the spreading, we further increase the effective array aperture by increasing the microphone spacing to 0.25 m. This was done in Case 3 and the corresponding result was shown in Fig. 15. In the result, the spreading of contours was considerably reduced at the expense of slight aliasing that could be distinguished from the observation in Case 2. The calculated result indicated that the source was located at (1 m, 27 m). The noise source was in fact a hydrogen compressor whose center line was at (2.4 m, 26.9 m) which was not far from the calculated result.

Case 4 was concerned with a field test performed outside a nitrogen factory. Spectral analysis of the data received by the array indicated that the measured noise had a broadband spectrum with bandwidth approximately 700 Hz and center frequency 1.6 kHz. Hence, the broadband plane-wave beamforming algorithms (based on 0.05 m spacing) were used to determine the DOAs of the noise sources. The result was shown in Fig. 16. From the result, the DOA of the noise source appeared to be at 90 deg with respect to the first microphone. This was a satisfactory result since the noise source of interest was actually a vent from the blower in the factory at $\theta = 88$ deg direction. To determine the distance of the noise source, the broadband spherical-wave algorithm was employed to process the array data Case 5). Since the originally calculated result was somewhat noisy (see Fig. 17(a)), a spatial filtering procedure was used to smooth the image. The low-pass filter for truncating the high-frequency components in the wavenumber domain was a fifth-order two-dimensional Butterworth filter (Gonzalez and

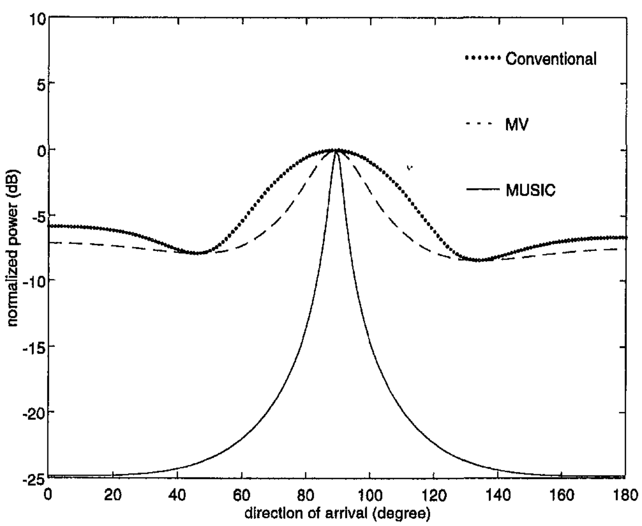

Fig. 16 Experimental result of Case 4 in Table 2 obtained by the broadband beamforming algorithms. The array consists of 6 microphones with a spacing 0.05 m.

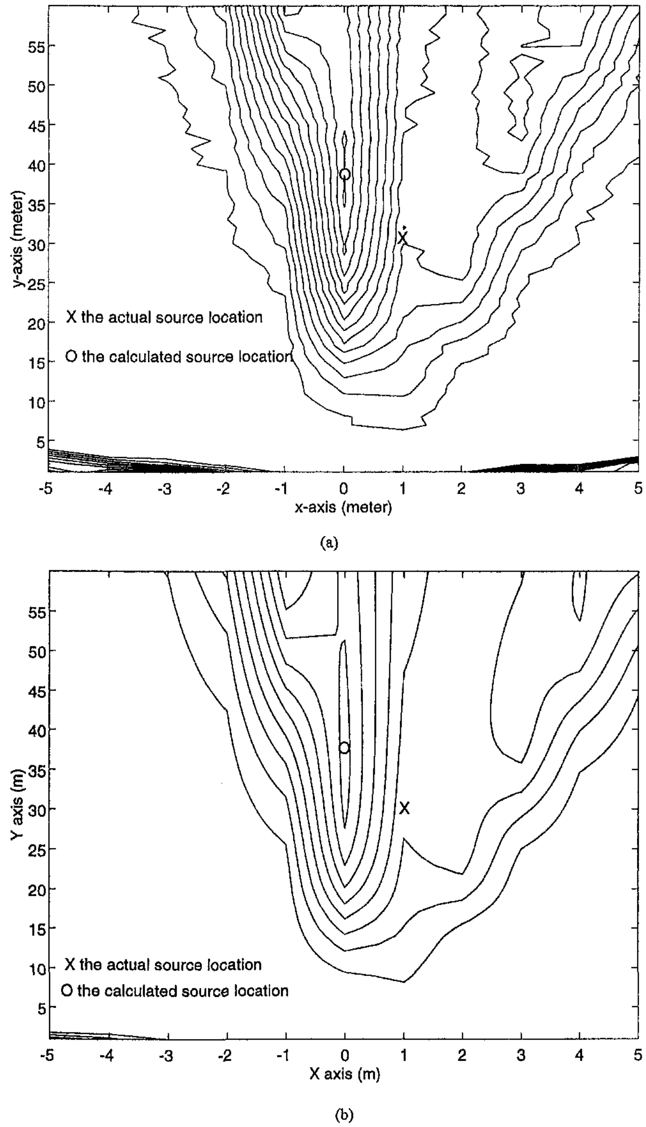

Fig. 17 Experimental result of Case 5 in Table 2 obtained by the broadband spherical-wave MUSIC algorithm. The array consists of 6 microphones with a spacing 0.2 m. (a) Without image smoothing; (b) with image smoothing.

Wintz, 1977). All filtering procedures including two-dimensional Fourier transformations between the spatial domain and the wavenumber domain were done by MATLAB subroutines (Thompson and Shure, 1993). The result was shown in Fig. 17(*b*). From the result, the noise source appeared to be located at (0 m, 38 m), while the source was actually located at (1 m, 29.4 m). Notable error in distance arose because of the broadband nature of the noise.

Conclusions

In this study, an acoustic beamforming system for identifying narrowband and broadband noise sources has been developed. From numerical simulations and field tests, the methods exhibit remarkable effectiveness in finding the directions as well as the distances of farfield sources in industrial environments. These techniques serve as useful alternatives to conventional noise source identification methods, especially for farfield applications.

The beamforming algorithms employed in this study differ in complexity and performance. The conventional method is computationally simple and less susceptible to multipath problems, while on the other hand it generally gives poor resolutions. The MV method generally produces improved resolution than the conventional method, provided the covariance matrix is invertible. The MUSIC method gives the best resolution among three methods at the expense of increased computational complexity. If there is a multipath problem, a spatial smoothing procedure can be used in conjunction with the MUSIC method to yield results with improved resolutions for plane-wave cases. In practical applications, one can always first use the conventional method or the MV method to determine the rough geometric extent of noise sources and then uses the spherical-wave MUSIC method to calculate the exact locations of noise sources. If the number of array microphones is a constraint, as usually is in practice, one may have to use a small spacing (say one half of the wave length) to determine the directions of the sources and then a large spacing to determine the distance within acceptable aliasing.

Acknowledgment

The work was supported by the National Science Council in Taiwan, Republic of China, under the project number NSC 83-0401-E-009-024.

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