A novel configuration method of the acoustic random beamforming array for multiple wideband moving sound source localization

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ABSTRACT
A new generating approach of the random beamforming array configuration is proposed for localization of multiple wideband moving sources. The target random beamforming array generation is on the basis of beamforming technique and the performance of random sparse array. Firstly according to the principle of the beamforming technique for moving sound source identification, the array angular resolution of multiple sources, array plane grid spacing and non-equidistance rings radius based on coaxial circular ring array in the polar coordinates are deduced. The candidate random arrays are generated by array plane segmenting schemes and array structural filter conditions. Secondly the target random beamforming array can be obtained from candidate random arrays by the performance evaluation and condition filtering in the polar coordinates. Finally numerical simulation and multiple wideband moving source identification test have been done. The results show that using the method to generate random beamforming array is effective. Compared with the traditional regular array which are suitable for moving sound source location, target random sparse array can save more time for array configuration and has more accurate multiple wideband moving sound source identification performance.

Keywords: Random Beamforming Array, Array Configuration, Moving Sound Sources
I-INCE Classification of Subjects Number(s): 52.3

1. INTRODUCTION
Traffic noise is one of the most important environmental noise source in countries. With the increased traffic intensity and extended road network, traffic noise levels have been enhanced. The characteristics of traffic noise, such as multiple sources, broadband and some sound source exist only if the vehicle moves, which are strongly coupled with the speed of the vehicle. More and more attentions have been focused on the methods for vehicle noise measurement. Beamforming is an important method which has been applied to the measurement of the noise source identification during the pass-by of a car and the characterization of those noise sources (1, 2, 3). Beamforming is an array-based measurement technique. The performance of a beamforming array is to a very large extent determined by the array geometry because this defines the beam former response through the array pattern. Basically, the source location is performed by estimating the amplitudes of plane (or spherical) wave incident towards the array from the directions. The peak in the array pattern which is called the main lobe depicts the actual incident direction of plane (or spherical) waves, i.e. the actual sources. A peak in the array pattern different from main lobe is named side lobe which is focused in a specific direction not equal to the actual incident wave direction, i.e. the ghost sources. The beamforming array geometry has crucial effect on the noise measurement.

At present many researchers have done lots of studies on the design and optimization of beamforming array for measurement the moving noise source (1, 4). Microphone arrays have proven to be useful for identifying noise sources on moving as well as stationary sources (5, 6). Many types of array configurations have been used. Among them are linear arrays, regular two-dimensional arrays, non-redundant arrays and crossed arrays (7, 8, 9). Non-uniform array configurations have been

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used to reduce the amount of redundant information gathered by the array at the expense of increasing sidelobe levels in the array directivity pattern(10, 11). To deal with this problem, Random sparse arrays have been proposed for reducing sidelobe levels based on the segment scheme of the array surface. A sparsely optimized method has been applied to the linear, planar, cylindrical or spherical array geometry by Legendre Fraction Transform. A fast random array generation method which is suitable for identifying the moving sound source has been introduced by partitioning, condition filtering and simulation evaluation. This overview shows that, to identify and characterize the moving sound source, the following conditions should be taken into consideration: 1) To avoid the spatial aliasing, the distance between two adjacent microphones should be at least half-wavelength of the incident wave. 2) Array aperture should be large enough to distinguish multiple wideband moving sound sources, because of the angular resolution is inversely proportional to the array diameter measured in units of wavelength. Especially at low frequencies, this requirement is usually too difficult to be met, so the resolution will be poor. 3) Configuration of array is vital for suppressing the side-lobe effect at high frequency. If an array is to meet conditions, the number of microphones and the aperture will be increased which will increase the measurement cost and add time of signal processing.

This paper is focused on the array layouts that can be used to identify the noise sources when the sources is moving and get more insight to the random sparse array setup of localization multiple wideband moving sound source. In the approach described here, the array configuration is random and two-dimensionally sparse which generate rapidly and efficiently by segment scheme, geometry constraints and array performance optimization. The sound field is generated by a moving car with two wideband sound sources and measured on the basis of the vehicle passby tests. In addition, array microphone outputs are processed in the frequency domain.

2. BEAMFORMING of MOVING SOUND SOURCE

The measurement of moving sound sources by beamforming array describes as Figure 1. Where the fixed coordinate system Oxyz depicts the array plane A, and the reconstruction plane B is moving with the car. The distance from the ground to the bottom of array is b and the focused distance is Zs.

![Figure 1](image)

**Figure 1 – Measurement of moving sound sources with beamforming array**

On the basis of the beamforming processing of pressure signals from the microphones on the array plane A, let \( F_s(\varepsilon, \eta) \) be the sound field features function at the random point \( s(\varepsilon, \eta) \) of the reconstruction plane B during the period of time and given as

\[
F_s(\varepsilon, \eta) = \int_{t_1}^{t_2} b^s(t, \varepsilon, \eta) dt
\]  

(1)

All points on the plane B can be calculated by formula (1). The three-dimensional sound pressure distribution graph will be obtained. Based on the relative intensity of the numerical value of feature function on the point of graph and the main source position can be identified. Here \( b(t, \varepsilon, \eta) \) is the delay-and-Sum beamformer output in the time domain. The FFT of it can be express as:

\[
B(\omega, \kappa) = \frac{1}{N} \sum_{m=1}^{w} w_m p_m(\omega) e^{i\omega \kappa} \quad m = 1, 2, \ldots N
\]  

(2)

Where \( p_m \) is the measured pressure signals of the \( m \)th microphone, \( w_m \) is a set of weighting factor that are...
applied to each individual microphone, $\kappa$ is a unit vector and $\vec{k} = (2\pi f / c)\kappa$ is the wave number vector of a plane wave incident from the direction $\kappa$ which the array is focused, where $f$ and $c$ represent the incident frequency and the propagation speed of the acoustic wave in the medium respectively. Now assume a plane wave incident with a wave number vector $\vec{k}_0$ different from the preferred direction. The pressure measured by the array can be written as

$$p_m(\omega) = p_0 e^{-j\omega t}$$

(3)

According to Eq. (2), array output can be represented as

$$\bar{B}(\omega, \kappa) = p_0 \sum_{m=1}^{M} \bar{w}_m e^{-j \vec{r}_m} = p_0 \bar{w} (\vec{k} - \vec{k}_0)$$

(4)

Here $\bar{w}$ function is the so-called array pattern

$$\bar{w} (\vec{k}) = \sum_{m=1}^{M} \bar{w}_m e^{-j \vec{r}_m}$$

(5)

Where $\vec{r}_m$ is the array microphones position vector. The array pattern determines the performance of a beamforming array and that is largely influenced by array geometry.

3. RANDOM ARRAYS FORMATION

3.1 Segment Schemes

The principle of segment schemes of basic array plane is that can be ensure the microphone arrangements as uniform as possible. For this aim, firstly, the array plane is divided into several equally areas, i.e., uniform subsections, by grid point spacing. Secondly, the alternative grid points can be generated randomly in subsections. Every one of alternative grid points has a possible to be selected to place microphone.

3.1.1 Grid Point Spacing

The grid spacing of basic array is the vital factor for localization the moving sound sources. On the assumption of plane phase plane, to avoid spatial aliasing, the minimum grid distance can be calculated as:

$$d_{\text{min}} = c / 2 f_{\text{max}} \sin \theta$$

(6)

Where $f_{\text{max}}$ is the highest frequency that contributes significantly to the sound field incident on the array, and $\theta$ is the incident angle. However in reality, the noise source lie within the near field of the microphone array, i.e., spherical spreading effects are significant. An analysis performed assuming spherical phase surfaces would result in a decrease in the allowable microphone spacing. So, in this paper, the grid point spacing in arc length is equal or greater than which is given by the below formula:

$$d = c / 2 f_{\text{max}}$$

(7)

Considering the frequency range of the sound source and the array geometry, the polar radius of the first ring is selected to be:

$$\rho_1 = c / 2 f_{\text{max}}$$

(8)

The radius difference of two adjacent rings is assumed as $\rho_\Delta = \rho_m - \rho_{m-1}$. In order to reduce mutual coupling and side lobe effect, the space of ring radius should be:

$$0.5 \lambda \leq \rho_\Delta < \lambda$$

(9)

Where $\lambda$ is the wavelength.

3.1.2 Alternative Grid Point

The microphone position will be selected from the alternative grid points in each subsection. For the uniformity of microphone layout, in the approach presented here, the same number alternative grid points and microphones are placed in each subsection, and the phase reference microphone need to be set on the center of the array. Based on the principle of aperture, the number of alternative grid point sets ten and the number of microphone is two. The ten alternative grid points’ polar angle can be generated randomly in each subsection. They are set on the rings from inner to outer according to the order of polar angle from large to small. The number of alternative grid point on the rings is
proportional to the rings numerical order, i.e., the number of alternative grid point on the rings from first to fourth is 1:2:3:4 in each subsection. The eight partition of a four rings coaxial circular array with eighty alternative grid points are showed in Figure 2.

![Figure 2 – The partition of basic coaxial circular ring array](image)

### 3.2 Candidate Random Arrays

The partition layout can be resolving the uniform distribution of microphone on the array plane as a whole. However, the number of the generating random arrays is very larger. In order to improve the seeking of the reliable random array layout, some structural constraint conditions have been applied for producing candidate random array.

The random array can be described with matrix of microphone position’s radius and angle in the polar coordinates. The row of matrixes depicts the distribution of the radius and angle and the column represents the subsections. Assuming \( C \) and \( J \) depicts the number of the subsections and alternative grid point in each subsection. Here the polar radius matrix \( R \) and the polar angle matrix \( \Phi \) of the basic coaxial circular array are described as:

\[
R = \begin{bmatrix}
\rho_1 & \rho_1 & \cdots & \rho_1 \\
\rho_2 & \rho_2 & \cdots & \rho_2 \\
\vdots & \vdots & \ddots & \vdots \\
\rho_j & \rho_j & \cdots & \rho_j \\
\end{bmatrix}_{J \times C} \\
\Phi = \begin{bmatrix}
\phi_1 & \phi_1 & \cdots & \phi_1 \\
\phi_2 & \phi_2 & \cdots & \phi_2 \\
\vdots & \vdots & \ddots & \vdots \\
\phi_j & \phi_j & \cdots & \phi_j \\
\end{bmatrix}_{J \times C} \tag{10}
\]

Assuming the candidate random array is composed of \( N \) microphones. The \( k^{\text{th}} \) ring has a number of microphones \( N_k \), polar angle \( \varphi_k \) and alternative grid point \( w_k \). The number of alternative grid point and microphones position of each subsection describes as \( \Sa \) and \( \Sm \). The structural constraint conditions are described as:

**Condition1**: The random array can be generated by choosing two microphone positions, and ten alternative grid point set in each subsection:

\[
\Sa = 2 , \quad \Sm = 10 \tag{11}
\]

**Condition2**: Ensure the distribution of microphones on each ring of the candidate random array meets the uniformity and avoiding the periodicity. In practice:

\[
\Sa / C \leq N_k \leq 2\pi
\]

\[
N_k \geq \left( W_j / w_j \right) \times N_1 , \quad q_k \geq \left( W_j / w_j \right) \times q_1 , \quad (k = 2,3,\ldots, M) \tag{12}
\]

**Condition3**: In order to resolve the clumping problem and considering the predefined position couldn’t be placed by other microphones, the constraint can be expressed as:

\[
\varphi_{pq} \geq \begin{cases}
2\pi / C & (p \neq q) \\
2\pi / (\Sa \times C) & (p = q) 
\end{cases} , \quad (p = 1,2,\ldots,C ; q = 1,2,\ldots,C) \tag{13}
\]

Here, \( \varphi_{pq} \) represents the element of row \( p \) and column \( q \) of the angle difference matrix \( \Psi \). This
matrix shows the angle difference of microphone position in the same or different subsection from the polar axis along counterclockwise direction. When a randomly generated array satisfied the above conditions, the random array is called a candidate random array which has been depicted by generated polar radius matrix $R'$ and the polar angle matrix $\Phi'$ which are described as:

$$
R' = \begin{bmatrix}
0 & \rho_1 & \ldots & \rho_j \\
\rho_2 & \rho_2 & \ldots & 0 \\
0 & 0 & \ldots & 0 \\
\vdots & \vdots & \ddots & \vdots \\
0 & 0 & \ldots & \rho_{j-1} \\
\rho_j & 0 & \ldots & 0
\end{bmatrix}_{j \times c}
$$

$$
\Phi' = \begin{bmatrix}
0 & \phi_1 & \ldots & \phi_j \\
\phi_2 & \phi_2 & \ldots & 0 \\
0 & 0 & \ldots & 0 \\
\vdots & \vdots & \ddots & \vdots \\
0 & 0 & \ldots & \phi_{j-1} \\
\phi_j & 0 & \ldots & 0
\end{bmatrix}_{j \times c}
$$

if not, it has to produce the microphone position matrix including the polar radius and polar angle matrix again. Then, it will re-search until creating candidate random arrays. Figure 3 shows one of the candidate random arrays generated by structural constraint conditions.

![Figure 3 – One of the candidate random arrays](image)

The target random array will be selected from the above candidate random array by array performance parameter sidelobe suppression ratio $r_{sp}$ and mainlobe width $r_{MW}$. In array pattern, the level of the side lobe relative to the main lobe defines the ability of the beamformer to suppress ghost images. Based on the three-dimensional random array response pattern $r_{sp}$ is defined as:

$$
r_{sp} = 20 \times \log \left( \frac{h_p}{h_v} \right)
$$

Where $h_p$ and $h_v$ represent the peak of mainlobe and the highest sidelobe peak respectively, as shown in Figure 4.

![Figure 4 – Three-dimensional candidate random array response pattern](image)
A good array design can be characterized with having a side lobe suppression ratio which meets the criterion. The parameter’s quantitative criterion has been acquired by statistics analysis of the numerical results. Defining as

\[
 r_{sp} \geq \begin{cases} 
 15 & \text{if } 16 \leq N \leq 24 \\
 20 & \text{if } N > 24 
\end{cases} \quad (16)
\]

In candidate random arrays, the one meeting the above criterion becomes a target random array, i.e., the random beamforming array. The flow diagram of the generating process is shown as Figure 5.

![Flow diagram of the generating process](image)

Figure 5 – Flow diagram of the generating process

Excepting sidelobe suppression ratio \( r_{sp} \) as a key performance factor, mainlobe width is also considered. The trade-off between them has been discussed through numerical simulation.

4. VERIFICATION

4.1 Numerical Simulation

In order to prove the validity of the generating method for testing multi moving sound source, two types of sources are discussed. The main parameters depict in Table 1.

<table>
<thead>
<tr>
<th>Factors</th>
<th>Azimuth angle, rad</th>
<th>Elevation Angle, rad</th>
<th>Frequency, Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Source1</td>
<td>( \pi / 4 )</td>
<td>( \pi / 4 )</td>
<td>1250-2000</td>
</tr>
<tr>
<td>Source2</td>
<td>(-\pi / 4)</td>
<td>(-\pi / 4)</td>
<td>800-1000</td>
</tr>
</tbody>
</table>

Candidate arrays are the four rings concentric circular ring array with 25 microphones. One microphone is place on the center of array and another 24 microphones are positioned randomly on the rings. The array configuration is the same as that in Figure 3. To makes sure that the target random array performance is optimal. In addition to sidelobe suppression ratio \( r_{sp} \), the trade-off of 3dB down mainlobe width and the sidelobe suppression ratio have been given through statistical analysis of two-thousand array layouts. The trade-off curves are plotted which is shown in Figure 6.
Figure 6 – Trade-off curves of two sources

The target random array is selected with $\theta = 16.2\degree$ and $r_s = 16\,\text{dB}$, as shown in Figure 7. The sidelobe suppression ratio curve of target random array with the frequency range of 500-3000Hz is presented in Figure 8.

Figure 7 – Target random array

Figure 8 – The curve of sidelobe suppression ratio

In case the two sources can be effectively identified. The three-dimensional array response pattern of above target random array shows in Figure 9.
4.2 Experimental Measurement

The testing array layout is as the same as that in Figure 7. Two sources are placed on the moving car with the speed 40-80km/h. The array plane aperture is 1m. The focal distance is 2m. The height from floor to the bottom of the array is 1.1m. The acoustic testing and imaging system is the Acoustic Camera of Institute of Acoustics of Chinese Academy of Sciences. The test-site picture is as Figure 10.

![Figure 10 - The test-site](image)

![Figure 11 - Testing results picture](image)

The test results from Figure 11 are as follow: when the car is in the testing domain, the recognition area of sound source identification is 0.04m² and 0.03 m², the sound pressure dynamic characteristic is 66-70dB. The number of ghost source is zero. From results, it was obtained that the target random array can effectively locate the two moving sound source with the difference wideband.

5. CONCLUSIONS

In the present work, a novel method was developed to generate a random array in which a number of microphones are randomly positioned on rings of coaxial circular ring array. The generated process was transformed to matrix operation in polar coordinates. Based on the moving sound source properties and array geometry, the candidate arrays matrixes could be created effectively. Through selecting the positions of matrixes element meet the condition of the required array, the target array could be generated rapidly and reduce the screening blindness.

From the simulated random array configuration generated based on different sound sources, it was observed that the sidelobe suppression ratio was strongly related to the frequency of sources below the 1000Hz. The influence from array layouts with the same aperture on the main lobe width was quite small.

To improve the accuracy of the simulation, the testing was operated with a target array. The results of the moving sound source test verify the feasibility and availability of the generating procedure.
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