Sound field simulation for circular array based on spatial circular convolution

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Abstract: We propose a sound field simulation method for a circular array. In a linear array, when the distance between the loudspeaker units is equal to the interval between the observation points on the lines paralleled to the array, the sound pressures on the observation line can be calculated by the spatial convolution of the set of transfer functions and loudspeakers’ driving signals. To apply this idea to a circular array, we developed a simulation method with equiangular observation points on the circle. The spatial circular convolution without zero padding, which is necessary in a linear array, can be used with this method. By conducting convolution based on the fast Fourier transform (FFT), the computational complexity is greatly reduced. Moreover, assuming that non-active loudspeakers are included in the loudspeaker array, the proposed method can be applied to an unequal interval array. For example, when the number of observation points is set to 128 and the number of loudspeakers is set to 32, circular convolution with FFT reduces the computational complexity to 75% compared to the conventional method. In addition, we argue that this method can be applied to a room in which the first reflected sounds are reflected from the floor. The proposed method is useful for simulating the sound field for a circular array when the suitable spatial samplings of the circumferential and radial directions are set.

Keywords: Sound field simulation, Loudspeaker array, Circular array, Spatial convolution

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1. INTRODUCTION

The directivity controls of sound sources [1,2] and wave field synthesis [3,4] by using loudspeaker arrays have recently been studied. These methods are expected to be applied to immersive audio communications and synthesis of the directivity of musical instruments. In these studies, it is important to investigate an algorithm to obtain the driving signals of the loudspeakers and the arrangement of loudspeakers. To evaluate the effectiveness of the algorithms and loudspeaker arrangement, we have to observe or calculate the sound field generated by the loudspeaker array.

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The image method [5], ray tracing [6], and finite-difference time-domain (FDTD) [7] methods are commonly used for sound field simulations and are used for rooms that have reflective walls. On the other hand, the basic performance of loudspeaker arrays is generally evaluated in an anechoic room. This means that the transfer function in the free field is briefly used in sound field simulations of loudspeaker arrays [8].

Even though a simulation of a loudspeaker array is simpler than the above methods, many sound pressures generated from every loudspeaker in the array must be calculated. Since many pieces of multi-channel A/D and D/A equipment are affordable, the number of loudspeaker elements used in an array is increasing. For example, we are able to implement a 64- or 128-channel loudspeaker
array [4]. Since the amount of calculation complexity of the sound field simulation greatly increases, an efficient simulation method is needed. The purpose of the fast simulation method of this paper is to increase the efficiency of research on loudspeaker arrangements and circular array processing algorithms. The fast simulation method allows us to quickly simulate many sets of loudspeaker arrangements and algorithms. Linear and circular arrays have been often investigated regarding configuration. One reason is that those configurations are simple and useful for many applications. Another reason is that analytical approaches based on the wave equation can be applied to such simple configurations.

So, we first explain the conventional calculation method for a sound field generated using a linear loudspeaker array [9,10], then discuss this application to compare it to our proposed simulation method for a circle array. Fundamentally, in a linear array, when the interval between the observation points is equal to that between the loudspeaker elements and when the observation line is parallel to the array line, the sound pressure on the observation line can be obtained by the spatial convolution of the array driving signals and transfer functions between the loudspeaker and observation points. Moreover, to reduce computational complexity, spatial convolution is calculated in the spatial frequency domain. This method is commonly used for line arrays.

To apply this idea to a circular array, we propose a simulation method with equiangular observation points on the circle. We argue that this method can be applied to an unequal interval array by setting the gains of the unnecessary loudspeakers to 0.

This paper is organized as follows. The simple conventional calculation method is described in Sect. 2. The principle of the line array simulation method based on spatial convolution is discussed in Sect. 3. Our proposed circle array simulation method is introduced and its performance evaluated in Sect. 4, and the paper is concluded in Sect. 5.

2. CONVENTIONAL METHOD

Let us consider the sound field simulation in the frequency domain for the line array shown in Fig. 1. If we can simulate the sound field in the frequency domain, we can easily expand the frequency domain result to the time domain result by using the inverse Fourier transform. Therefore, we discuss only the simulation method in the frequency domain. When the output signal of each loudspeaker at position \( r_s(n) = (x_s(n), y_s(n), z_s(n)) \) on a line is \( X(\omega, r_s(n)) \), the acoustic pressure \( Y(\omega, r_s(m)) \) at observation position \( r_o(m) = (x_o(m), y_o(m), z_o(m)) \) is

\[
Y(\omega, r_o(m)) = \sum_{n=0}^{N-1} G(\omega, r_o(m) - r_s(n)) X(\omega, r_s(n)), \tag{1}
\]

where \( \omega \) is the angular frequency, \( G(\omega, r_o(m) - r_s(n)) \) is the transfer function between \( r_s(n) \) and \( r_o(m) \), and \( n \) and \( m \) respectively indicate the \( n \)-th and \( m \)-th positions of the loudspeaker unit and observation point, and \( N \) is the number of loudspeakers in the array. Usually, \( Y(\omega, r_o(m)) \) is calculated at many observation points \( r_o(m) \) in the evaluation sound field area as shown by the gray hatch region in Fig. 1. Since the basic performance of the loudspeaker array is evaluated in an anechoic room, the transfer function \( G(\omega, r_o(m) - r_s(n)) \) in the free field is used for the simulation [11].

\[
G(\omega, r) = \frac{1}{4\pi|\rho|} \exp(-j|\rho|), \tag{2}
\]

where \( k = \omega/c \) is the wave number, \( c \) is sound velocity, and \( r = r_o(m) - r_s(n) \) is the relative distance between the loudspeaker unit and observation point. We omit the time term \( \exp(j\omega t) \).

We investigated the amount of calculations when obtaining the \( Y(\omega, r_o(m)) \) distribution generated from a 64-element line loudspeaker array with equal intervals of 5 cm. The length of the line array was 3.15 m. The size of the target calculating sound field was set to \( 6.35 \times 6.35 \) m². This size was twice that of the array. Since the interval between the observation points was also 5 cm for the \( x \)- and \( y \)-axes, the total number of observation points was \( 128 \times 128 \). The sound pressure generated by each loudspeaker element has to be calculated at these observation points. Therefore, the total calculation amount of the sound field generated by the 64-element loudspeaker array was \( 128 \times 128 \times 64 \), that is, roughly \( 10^6 \).

This computational complexity is independent of the array shape but depends on only the number of elements in the array and observation points. Therefore, even if
we change the line array to a circular one, the calculation amount is the same for the same \( N \).

3. PRINCIPLE OF LINE ARRAY SIMULATION BASED ON SPATIAL CONVOLUTION

To calculate the sound field generated by a linear array, we first consider the observation line, which is paralleled to the linear array and its distance from the array is \( l \).

If the intervals of each loudspeaker unit and the intervals of the observation point are equal, distance \( d \), \( r_s(n) \) and \( r_o(m) \) are

\[
r_s(n) = (x_s(0) + nd, y_s, z_s), \tag{3}
\]

\[
r_o(m) = (x_o(0) + md, y_s + l, z_s). \tag{4}
\]

Here, \( n = 0, \ldots, N - 1 \), \( m = 0, \ldots, M - 1 \), where \( M \) is the number of observation points on the observation line, and \( r_s(0) \) and \( r_o(0) \) are the absolute positions at the left.

In such a case, Eq. (1) becomes a linear spatial convolution. The convolution in the time domain corresponds to the products in the Fourier transformed frequency domain \([9,12]\). Therefore, the spatial convolution of Eq. (1) can also be calculated in the spatial Fourier transform domain. Although this idea is basically used in the Nearfield Acoustic Holography in a continuous system \([10]\), we explain a sound field simulation method in the spatial frequency domain in which the source positions are discrete and the number of sources is finite.

Figure 2 shows the relationships of acoustic paths between the set of transfer functions and the observation points. Figures 2(a) and (b) show acoustic paths of the 1st and 2nd observation points, and Fig. 2(c) shows acoustic paths of the observation point at the center position. The sound pressure at each observation point is the sum of those from the loudspeakers. It requires \( N \) products to calculate \( Y(\omega, r_s(m)) \) at each observation point. Therefore, the entire calculation complexity is \( N \times M \) for an observation point.

From Eq. (2), the transfer function \( G(\omega, r) \) depends on only the relative distance \(|r| = |r_s(m) - r_s(n)|\) in the free field. Since some of the transfer functions are the same as those in Figs. 2(a)–(c), the calculating operations in Fig. 2 can be replaced with the operations in Fig. 3. In Fig. 3, the virtual observation point is set at one position. When a loudspeaker array is positioned at the far right, we can see that the sound pressure at the virtual observation point is equal to \( Y(\omega, r_o(0)) \) in Fig. 2(a). By moving the loudspeaker array position left, the sound pressures are calculated at each loudspeaker array position. This is the convolution of \( M + N - 1 \) transfer functions and \( N \) loudspeaker source signals. The number of transfer function sets is independent of \( r \) between the observation point and loudspeaker position.

Next, we explain the method of this spatial convolution based on the discrete Fourier transform (DFT). The \( M + N - 1 \) transfer functions from right to left are prepared, as shown in Fig. 3.

\[
G(\omega, r_n) = \frac{1}{4\pi|r_n|}\exp(-jk|r_n|), \tag{5}
\]

\[
r_n = r_o(0) - r_s(N - 1 - n) \quad (n = 0, \ldots, M + N - 2), \tag{6}
\]

where \( r_s(N - 1) \) is the actual position of the loudspeaker unit at the right most position. The spatial Fourier transform in the \( x \)-direction of \( G(\omega, r_n) \), \( G(\omega, r_1), \ldots, G(\omega, r_{M+2}) \) is
Here, $k_s$ is an index that corresponds to the wave number. Next, we prepare the vector of the loudspeaker array signals with $M - 1$ zero padding.

$$X(\omega, r_s) = [X(\omega, r_s(0)), X(\omega, r_s(1)), \ldots, X(\omega, r_s(N - 1)), 0, \ldots, 0]^T.$$  

(8)

The DFT of $X(\omega, r_s)$ is

$$\tilde{X}(\omega, k_s) = \text{DFT}(X(\omega, r_s)).$$  

(9)

Finally, we obtain the inverse Fourier transform of the dot product of $\tilde{G}(\omega, k_s)$ and $\tilde{X}(\omega, k_s)$,

$$P(\omega, r_s) = \text{IDFT}(\tilde{G}(\omega, k_s)\tilde{X}(\omega, k_s)).$$  

(10)

The sound pressures $P(\omega, r_s)$ from $n = N$ to $N + M - 1$ correspond to the target sound pressures $Y(\omega, r_s(m))$ ($m = 0, \ldots, M - 1$). The entire sound field can be obtained by calculating $P(\omega, r_s)$ by changing $iy$. Although the computational complexity of the linear convolution is $MN$, the complexity can be reduced by using the fast Fourier transform (FFT) $(M + N - 1)\log_2(M + N - 1)$. Note that the efficiency of FFT requires that $M + N - 1$ is equal to $2^q$ ($q$ is a natural number). To solve this problem, we can modify $M$ and $N$ by using zero padding. For example, virtual loudspeakers with 0 gains are added to the array.

4. SOUND FIELD SIMULATION FOR CIRCULAR ARRAY

In this section, we introduce our sound field simulation method for circular arrays. Usually, the calculation points for a circular array are on an equal interval mesh in the target sound field, like in a linear array. In this situation, the linear convolution method explained in Sect. 3 cannot be applied to a circular array. However, we can use the circular convolution when the observation points are on a circle with equal angle intervals.

4.1. Principle of Proposed Method

Figure 4 shows the relationship between the circular loudspeaker array and observation points of $Y(\omega, R_c(\theta_m))$ at $R_c(\theta_m) = (R_c \cos \theta_m, R_c \sin \theta_m, z_c)$ in the cylindrical coordinate $(\theta, R, z)$, and $R_c$ and $\theta_m$ respectively indicate the radius of the observation circle and the angle of the observation position. It is assumed that $N$ is equal to $M$. The angle intervals between the loudspeaker units are equal (equiangle). When the observation point is changed to the neighbor position by comparing Figs. 4(a) and (b), the set of the transfer functions is simply rotated since $G(\omega, r)$ depends on only $r$ between the loudspeaker unit position and observation position. Therefore, $G(\omega, R_c(\theta_m) - R_c(\phi_n)))$ can be obtained by the following convolution.

$$Y(\omega, R_c(\theta_m)) = \sum_{n=0}^{N-1} G(\omega, R_c(\theta_m) - R_c(\phi_n))X(\omega, R_c(\phi_n)).$$  

(11)

where $R_c(\phi_n) = (R_c \cos \phi_n, R_c \sin \phi_n, z_c)$ denotes the position of the loudspeaker unit, and $R_c$ and $\theta_m$ respectively indicate the radius of the circular array and the angle of the loudspeaker position. Since Eq. (11) is the circular convolution for $\theta$ in the cylindrical coordinate, we can obtain the sound pressures $Y(\omega, R_c(\theta_m))$ by using circular DFT,

$$Y(\omega, R_c(\theta_m)) = \text{IDFT}(\tilde{G}(\omega, n)\tilde{X}(\omega, n)),$$  

(12)

$$\tilde{G}(\omega, n) = \text{DFT}([G(\omega, R_c(\theta_0) - R_c(\phi_n)), \ldots, G(\omega, R_c(\theta_{M-1}) - R_c(\phi_n))]),$$  

(13)

$$\tilde{X}(\omega, n) = \text{DFT}([X(\omega, R_c(\phi_0)), \ldots, X(\omega, R_c(\phi_{M-1}))]).$$  

(14)

If $N = 2^q$, DFT can be calculated by FFT. Although the computational complexity of a simple convolution (Eq. (11)) is $M^2$, the computational complexity of the FFT version is $M \log_2 M$. Since this spatial convolution is exactly circular, zero padding techniques are not required.

4.2. Different $N$ and $M$

We can choose different $N$ and $M$ by assuming that some loudspeakers’ gains are set to 0, as shown in Fig. 5. In Fig. 5(a), the loudspeakers are alternately active. The position of the working loudspeakers (loudspeakers with no 0 gains) can also be placed at arbitrary positions, as shown in Fig. 5(b). If the number of working loudspeakers is $L$, the computational complexity of the simple convolution is $LM$. On the other hand, that of the circular convolution with FFT is still $M \log_2 M$.  

Fig. 4 Relationship of acoustic paths between circular array and circular observation points.
Figure 6 shows the computational complexity when using \(L\) loudspeakers and \(M\) observation points. The vertical axis indicates the logarithm of computational complexity and the horizontal axis indicates the logarithm number of working loudspeakers in the array. The symbols \(\bigcirc\), \(\checkmark\), and, \(\ast\) respectively correspond to \(M = 32\), \(64\), and \(128\). The dashed lines correspond to the computational complexities of the simple convolution \(LM\), solid lines: circular convolution by FFT \(M \log_2 M\).

If \(L\) is over 8, the computational complexities of the proposed method are less than the simple calculations. For example, when \(M\) is set to 128 and \(L\) is 32, the circular convolution with FFT reduces the computational complexity by 75\% (\(\ast\) on the solid and dashed lines for \(L\) of 32 in Fig. 6).

4.3. Spatial Sampling and Number of Observation Points

Although the proposed method can reduce a large amount of computational complexity, the observation points on the circles are different from the square mesh pattern, as shown in Figs. 7(a) and (b). Figure 7(a) shows a \(50 \times 50\) square mesh of observation points. The observation area is restricted to the circle, which is the same area with the circular convolution method. In this case, the total number of observation points is 1,963.

When preparing the same number of observation points so that the distance between the neighboring observation points might become as uniform as possible, the circular convolution method can, for example, use the observation points that are divided into 64 circumferential directions and 30 radial directions. In this case, the total number of observation points is 1,920. If \(L\) in the array is 32, the computational complexity of Fig. 7(a) is 1,963 + 32 = 62,816. The computational complexity of Fig. 7(b) is \(30 \times 64 \times \log_2 64 = 11,520\) (about 18\% of the mesh method) thanks to the FFT. Although the square mesh method can be used to uniformly calculate the sound field, the circular convolution method calculates the sound field non-uniformly. The circular convolution method calculates the sound pressures that are dense near the center of the circle but calculates those that are sparse away from the center.

To compare the differences in calculated sound pressure distributions, we simulated a sound field under the conditions mentioned in the next section.

4.4. Simulation Results

We simulated the sound field inside a circle with a radius of 3 m generated by a circular array with a radius 1.5 m. The number of loudspeakers was set to 64, and the frequencies were 1 and 2 kHz. The sound velocity was set to 340 m/s. The driving signals of the loudspeaker array were obtained using the following equation to synthesize the monochromatic plane waves in the circular array [13],

\[
X(\omega, R_s(\phi_s)) = \frac{1}{2\pi R_s} \sum_{v=1}^{V} \hat{P}_s(\omega) G_s(\omega) e^{iv\phi_s}. \tag{15}
\]
Here, $\tilde{G}_s(\omega) = \frac{j}{4} H^{(2)}_0(kR_0)$, $\tilde{P}_s(\omega) = j^{-\nu} e^{-j\theta_{pw}}$. $V$ is the order of the wave number, $H^{(2)}_0$ is the Hankel function of second kind, and $\theta_{pw} = 9/8\pi$ is the propagation direction of the plane wave.

The simulations were conducted under the four conditions listed in Table 1. The number of observation points of Condition I was calculated using $60 \times 60 \times \pi$. This corresponds to the number of observation points inside the same circle with radius of 3 m. Since the interval of the mesh was 5 cm in Condition I, the longest interval corresponding to the diagonal line was $5\sqrt{2}$ cm. Then, the aliasing frequency was calculated as the frequency of a wavelength twice the length of a diagonal line. This spatial aliasing frequency was constant for the entire area. For Conditions II to IV, the spatial aliasing frequencies were calculated on the circle with radius of 1.5 m. The interval of the observation points on the circle was $d_\theta = 2 \times 1.5 \times \pi/N_\theta$ ($N_\theta$ indicates the number of $\theta$ division). The interval of the axial direction was $d_R = 3/N_R$ ($N_R$ indicates the number of $R$ division). Therefore, the longest interval near the circle was approximately calculated as the diagonal line of the rectangle with $d_\theta$ and $d_R$. Then, each aliasing frequency was calculated as the frequency that has the wavelength twice as long as the diagonal line. Note that, the spatial aliasing frequencies varied with the radius of the observation positions in Conditions II to IV.

Figures 8 and 9 show the simulation results at 1 and 2 kHz, respectively. Figures 8 and 9(a) to (d) correspond to simulation Conditions I through IV in Table 1, respectively. We used the Delaunay triangulation and linear interpolation for the triangle plain in order to smooth the images [14].

<table>
<thead>
<tr>
<th>Condition</th>
<th>Number of $X-(\theta)$ divisions</th>
<th>Number of $Y-(R)$ divisions</th>
<th>Number of observations</th>
<th>Aliasing Frequency at 1.5 m</th>
<th>Computational complexity</th>
</tr>
</thead>
<tbody>
<tr>
<td>I (square)</td>
<td>120</td>
<td>120</td>
<td>11,309</td>
<td>2,404 Hz</td>
<td>723,776</td>
</tr>
<tr>
<td>II (circular)</td>
<td>128</td>
<td>60</td>
<td>7,680</td>
<td>1,910 Hz</td>
<td>53,760</td>
</tr>
<tr>
<td>III (circular)</td>
<td>128</td>
<td>90</td>
<td>11,520</td>
<td>2,103 Hz</td>
<td>80,640</td>
</tr>
<tr>
<td>IV (circular)</td>
<td>64</td>
<td>180</td>
<td>11,520</td>
<td>1,147 Hz</td>
<td>69,120</td>
</tr>
</tbody>
</table>

(a) Cond. I: square mesh pattern (120 × 120).
(b) Cond. II: circular mesh pattern (128 circumferential directions × 60 radial directions).
(c) Cond. III: circular mesh pattern (128 circumferential directions × 90 radial directions).
(d) Cond. IV: circular mesh pattern (64 circumferential directions × 180 radial directions).

Fig. 8 Sound field simulation results of circular array at 1 kHz.

(a) Cond. I: square mesh pattern (120 × 120).
(b) Cond. II: circular mesh pattern (128 circumferential directions × 60 radial directions).
(c) Cond. III: circular mesh pattern (128 circumferential directions × 90 radial directions).
(d) Cond. IV: circular mesh pattern (64 circumferential directions × 180 radial directions).

Fig. 9 Sound field simulation results of circular array at 2 kHz.
Condition I was a conventional one in which $Y(\omega, r_0)$ at 120 $\times$ 120 mesh points in a 3 $\times$ 3 m$^2$ area were obtained, as shown in Figs. 8(a) and 9(a). Since the distance between the observation points was 5 cm, it is suitable to express 2 kHz sound waves considering the spatial aliasing frequency given in Table 1.

Condition II was not suitable for expressing 2 kHz sound waves for the radial direction. Therefore, the differences between the sound pressure distributions in Figs. 9(a) and (b) can be seen, especially outside the array. Although Condition III was suitable for expressing 1 kHz sound waves for the entire area, it is only suitable for expressing 2 kHz sound waves inside the circular array. In Condition IV, we can see spatial aliasing due to the low resolution simulator outside the circular array even for 1 kHz sound waves, as shown in Fig. 8(d). The sound pressures at the observation points are exact for both methods.

Next we evaluated the quantitative qualities for each condition. The evaluations were conducted on the $x$- and $y$-axes and two circles, as shown in Fig. 10. We used the two circles with different radii of 1.025 and 2.025 m for the circumferential direction’s evaluations. These radii were not the same as the observation points. The number of evaluation points was set to 240 for each case. We evaluated the mean square errors (MSEs) between the true values and the calculated pressures including the interpolation results of the 2.5 cm intervals on the axial direction. The MSE was calculated as

$$MSE = 10 \log_{10} \frac{\sum_{n=1}^{N} S^2(\omega, r_{E}(n))}{\sum_{n=1}^{N} (S(\omega, r_{E}(n)) - \hat{Y}(\omega, r_{E}(n)))^2}. \quad (16)$$

The $S(\omega, r_{E}(n))$ is the true values, $\hat{Y}(\omega, r_{E}(n))$ is the calculated pressures including the interpolation results, $r_{E}(n)$ indicates the evaluation position, and $N (=240)$ is the number of evaluation points. The interpolations were performed by the above-mentioned method [14].

The MSEs of each evaluation are shown in Fig. 11 ((a) 1 kHz and (b) 2 kHz). The results of Conditions I and II, those having the same observation points, were exactly the same on the $x$- and $y$-axes. This is because the difference between these two conditions was only the use of the FFT for the spatial convolutions. Therefore, the calculated results were the same including the interpolation results. The accuracy of Conditions III and IV were higher than the Conditions I and II on $x$- and $y$-axes because the numbers of the observation points of Conditions III and IV were more than those of Conditions I and II.

On the other hand, the evaluation results of circumferential directions on the circles $r = 1.025$ m and $r = 2.025$ m showed that the MSEs of Condition III were almost the same as those of Condition I because the number of evaluation points was the same as those of Condition I. The MSEs of Condition IV were wrong because of the small number of observation points on the
circles. In particular, the MSE of \( r = 2.025 \) m of Condition IV was the worst in Fig. 11(b) due to spatial aliasing. The sound waves outside the circular loudspeaker array in Fig. 9(d) greatly differed from those in Fig. 9(a). From these results, the MSEs of Condition III were almost the same as Condition I on average.

If suitable spatial samplings of the circumferential and radial directions are set, we can conduct a wave field simulation of the circular array with low computational complexity by using the circular convolution with FFT. We can also change the spatial angle sampling at near and far observation points from the center of the circle.

Note that when evaluating the error of the target and synthesized sound fields, we have to regularize the energy of the error by \( \frac{1}{2R^2} \) for each observation circle. This is because the observation point distribution does not uniformly depend on the radius of the circle, as shown in Fig. 7. Moreover, the proposed method cannot be applied to obtain the original target sound field, for example, the monopole sound source located far from the array. In such a case, the original sound field has to be calculated using the conventional method. However, once the original sound field is calculated, the circular array simulation can be efficiently conducted using the circular convolution.

4.5. Simulation for First Reflected Sound from Floor

Let us consider the simulations of a circular array in a room in which the first reflected sound is reflected from the floor. Although the side walls and ceiling can be made of absorption materials, it is difficult for the floor to absorb sound. Moreover, when a circular array is set on a table, the first reflection from the table surface may affect the basic performance of the array. Therefore, investigation of the first reflection from the floor is very important. Fortunately, the proposed circular convolution method is independent of height (z-direction) if the circular array and observation circles have the same center position. Therefore, when adding the sound field generated by the original array and the reflected sound field generated by the image sources of the array, we can obtain the entire sound field. We simulated the sound field of the array at the z = 1 m position in a room whose floor had perfect reflection (reflection coefficient = 1). Figures 12(a) and (b) respectively show the simulation results of the conventional mesh method and the proposed method (Condition III) at 1 kHz. Figures 12(c) and (d) show those at 2 kHz. Both sets of simulation results were mostly equal.

5. CONCLUSION

We described our efficient sound field simulation method for a circular array by using spatial convolution with DFT. When the distance between the loudspeaker units is equal to the interval between the observation points on the lines paralleled to the array in the linear array, the sound pressures on the observation line can be represented by the spatial convolution of the set of transfer functions and loudspeakers’ driving signals. To apply this idea to a circular array, we investigated a simulation with cylindrical coordinates. When the observation points are set equiangular on the circle instead of in a linear mesh, the sound pressures can be obtained using spatial circular convolution. Moreover, assuming that non-active loudspeakers are included in the loudspeaker array, we showed that the proposed method can be applied to an unequal interval array.

We conducted simulations to confirm the effectiveness of the proposed method. The proposed method was more efficient than the conventional method when the number of observation points on the observation circle was over 32 and the number of working loudspeakers was 8 for a circular array with a 1.5-m radius at 1 and 2 kHz. For example, when the number of loudspeakers was 32 and the number of observation points in the target sound field area was about 1,920, the computational complexity of the proposed method was 18% that of the conventional method. Other simulations showed that the images of the sound field were mostly the same when selecting suitable spatial sampling for circumferential and radial directions.
by taking spatial aliasing into consideration. In addition, we showed that the proposed method can be applied to a room with the first reflected sounds from the floor. The proposed method can enable efficient sound field simulation for a circular array and is also very useful for obtaining impulsive waves in the time domain since we have to calculate the sound field for many frequencies.

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