7 - 12 SEPTEMBER 2014, KRAKÓW
FORUM ACUSTICUM

Design of inverse filters for multi-channel directivity reproduction of a sound source -Application to telecommunication system using sound casks-

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Summary
We developed a 96-ch immersive sound reproduction system based on the boundary surface control and are working on implementing it into an acoustical-space-sharing telecommunication system. In this paper, we propose a sound directivity reproduction method for a telecommunication system based on the boundary surface control. We estimate the radiation from a sound source by solving an inverse problem between the microphone array and the secondary sources. Then, the estimated radiation is reproduced by the secondary sources at the location of a virtual speaker or musical player in the shared acoustical space. Because the sound field of the virtual listener's position can be reproduced on the basis of boundary surface control, a distant listener can listen the speaker's sound involving changes which occurs as the result of the speaker's body movements as if they were in the same room or hall. We verify the effectiveness of this method by evaluating the directivity of the secondary sound sources in a free-field simulation and an experiment in an anechoic chamber.

PACS no. 43.60.Fg, 43.60.Dh, 43.55.Lb

1. Introduction

The development of telecommunication technologies has made it possible for us to communicate anywhere at any time. However, the importance of actual meetings has not diminished.

Much research has been conducted on the subject of 'reality' in telecommunication since 1980, when Minsky proposed the concept of telepresence[1]. Whether acoustical reality can be achieved depends on the performance quality of the sound reproduction system.

We have developed a three-dimensional sound field reproduction system based on the boundary surface control (BoSC) that delivers high performance in sound localisation[2, 3]. In recent years, we developed a BoSC sound reproduction system, 'Sound Cask', which has enough interior space to play a small musical instrument. Using more than one BoSC system, we have been developing a telecommunication system that enables some distant parties to play music as if they were in the same room or concert hall. We call this telecommunication system a 'sound field sharing system'[4].

One of the features of the BoSC sound reproduction system is that it allow a listener to move his/her head. In addition, the players and speakers can move their bodies during the communication process as well. It is possible that the minor changes in the sound caused by the speaker's body movements stimulate the sense of the presence of the other party[5]. A change in the sound directivity is specified as one of the physical changes caused by the players' or speakers' movements.

The sound field sharing system requires the transfer functions of the spaces shared by the players. In most cases, we acquire these transfer functions by taking measurements. The measured transfer functions contain information on the loudspeakers' sound directivity, which, in the sound field sharing system, replaces the players' or speakers' sound directivity. We used this feature in developing the previous telecommunication system, which can change the speaker's sound directivity in accordance with his/her facing angle[6]. However, this method using a loudspeaker's directivity cannot reproduce the original sound directivity of the sound source.

In this paper, we propose a sound directivity reproduction method that estimates the radiation from a sound source by solving an inverse problem between the secondary sound sources enclosing the sound source and a microphone array outside the secondary sound sources. We demonstrate the effectiveness of this method by simulations and measurements made in an anechoic chamber.

2. Method of sound directivity reproduction with inverse filters

2.1. Concept

Figure 1 shows the concept of the proposed method of sound directivity reproduction.

First, we consider the radiation from sound sources inside a three-dimensional volume V_1 that is bounded by a surface S_1. On the basis of the outer Helmholtz integral equation (HIE)[7], the sound pressures of a volume V_O outside of V_1 are

$$p\left(r'_1\right) = \iint_{S_1} \left(G\frac{\partial p\left(r_1\right)}{\partial n_1} - p\left(r_1\right)\frac{\partial G}{\partial n_1} \right) dS_1 \quad (1)$$
$$\left(r_1 \in S_1, r'_1 \in V_O\right),$$

where G is a Green's function, and n_1 is a normal vector to the surface S_1. This equation implies that the radiation of the sound sources is expressed by the sound pressures and particle velocities on the closed surface S_1.

Now we consider an observation surface S_{E} that is outside of the surface S_1. Because $S_{\mathrm{E}} \subset V_O$, the sound pressures on S_{E} are also given by equation 1. We discretise the surfaces S_1 and S_{E} into N_1 and M_1 small elements of areas $\Delta S_{1,k} (k = 1, \ldots, N_1)$ and $\Delta S_{\mathrm{E},j} (j = 1, \ldots, M_1)$, respectively. From equation 1, the sound pressure in the area $\Delta S_{\mathrm{E},j}$ is

$$p_{\mathrm{E}}\left(j\right) = \sum_{k=1}^{N_1} \left(G_{j,k}\frac{\partial p_1\left(k\right)}{\partial n_1} - p_1\left(k\right)\frac{\partial G_{j,k}}{\partial n_1} \right) \Delta S_{1,k}, \quad (2)$$

where $p_1(k)$ is the sound pressure in $\Delta S_{1,k}$, and $G_{j,k}$ is the Green's function between the areas $\Delta S_{\mathrm{E},j}$ and $\Delta S_{1,k}$.

Let $\Delta S_{\mathrm{IN},1,k}$ and $\Delta S_{\mathrm{OUT},1,k}$ be small elements of area that are inside and outside of $\Delta S_{1,k}$, respectively, in the direction normal to S_1 and at a distance h from its surface. When the distance h is small enough, the sound pressures and particle velocities in the small area $\Delta S_{1,k}$ are

$$p_1\left(k\right) \cong \frac{p_{\mathrm{IN},1}\left(k\right) + p_{\mathrm{OUT},1}\left(k\right)}{2}, \quad (3)$$

$$\frac{\partial p_1\left(k\right)}{\partial n_1} \cong \frac{p_{\mathrm{IN},1}\left(k\right) - p_{\mathrm{OUT},1}\left(k\right)}{2h}, \quad (4)$$

where $p_{\mathrm{IN},1}(k)$ and $p_{\mathrm{OUT},1}(k)$ are the sound pressures in $\Delta S_{\mathrm{IN},1,k}$ and $\Delta S_{\mathrm{OUT},1,k}$ respectively.

Inserting equations 3 and 4 into equation 2 yields

$$p_{\mathrm{E}}\left(j\right) = \frac{1}{2}\sum_{k=1}^{N_1} \left(\left(\frac{G_{j,k}}{h} - \frac{\partial G_{j,k}}{\partial n_1} \right) p_{\mathrm{IN},1}\left(k\right) - \left(\frac{G_{j,k}}{h} + \frac{\partial G_{j,k}}{\partial n_1} \right) p_{\mathrm{OUT},1}\left(k\right) \right) \Delta S_{1,k}. \quad (5)$$

Therefore, we obtain a matrix form of equation 5:

$$\mathbf{p}_{\mathrm{E}} = \mathbf{H}_{\mathrm{E}}\mathbf{p}_1, \quad (6)$$

where

$$\mathbf{p}_1 = \left[p_{\mathrm{IN},1}\left(1\right), \ldots, p_{\mathrm{IN},1}\left(N_1\right), p_{\mathrm{OUT},1}\left(1\right), \ldots, p_{\mathrm{OUT},1}\left(N_1\right)\right]^T,$$
$$\mathbf{H}_{\mathrm{E}} = \frac{1}{2}\mathbf{GS},$$
$$\mathbf{G} = \left[\mathbf{G}_1 \quad \mathbf{G}_2\right], \quad \mathbf{S} = \begin{bmatrix} \mathbf{S}_d & 0 \\ 0 & \mathbf{S}_d \end{bmatrix},$$
$$\mathbf{G}_1\left(j,k\right) = \frac{G_{j,k}}{h} - \frac{\partial G_{j,k}}{\partial n_1}, \mathbf{G}_2\left(j,k\right) = \frac{G_{j,k}}{h} + \frac{\partial G_{j,k}}{\partial n_1},$$
$$\left(j = 1, \ldots, M_1, \quad k = 1, \ldots, N_1\right).$$

Here, $\mathbf{p}_{\mathrm{E}}$ is the column vector of the sound pressures in all the small areas $\Delta S_{\mathrm{E},j}$, $\mathbf{H}_{\mathrm{E}}$ is an $M_1 \times 2N_1$ matrix, $\mathbf{S}_d$ is a diagonal matrix $diag(\Delta S_{1,1,}, \ldots, \Delta S_{1,N})$, and $[\cdot]^T$ denotes the transpose.

According to equation 6, the sound pressure vector of the surface S_1 is represented by the following equation using the inverse matrix of $\mathbf{H}_{\mathrm{E}}$:

$$\mathbf{p}_1 = \mathbf{H}_{\mathrm{E}}^{-1}\mathbf{p}_{\mathrm{E}}. \quad (7)$$

Equation 7 implies that we can obtain the sound pressures and particle velocities on the surface S_1 from the sound pressures on the surface S_{E} by solving the inverse problem. From equation 2, we also find that the radiation from the sound source is obtained through equation 6.

Next, we consider a reproduction of the sound source radiation in a shared sound field. Let V_1' and S_1' be a volume and surface in the shared sound field that are congruent with V_1 and S_1, respectively. On the basis of the outer HIE, the sound pressures in a volume $V_{O'}$ which is outside of V_1' are given by an equation similar to equation 1 using the sound pressures and particle velocities on the surface S_1'. Considering the congruency, we find that the radiation from the sound source in V_1 is reproduced in $V_{O'}$ when the sound pressures and particle velocities on S_1' correspond to those on S_1: $\mathbf{p}_1 = \mathbf{p'}_1$, where $\mathbf{p_1'}$ is the column vector of the sound pressures on S_1'. That is, when this equation is satisfied, there is a virtual sound source in V_1' in the shared sound field.

In the shared sound field, we consider a volume V_2 where a virtual listener is located and which is bounded by a surface S_2. In the reproduced sound field, we also consider a volume V_2' and surface S_2'

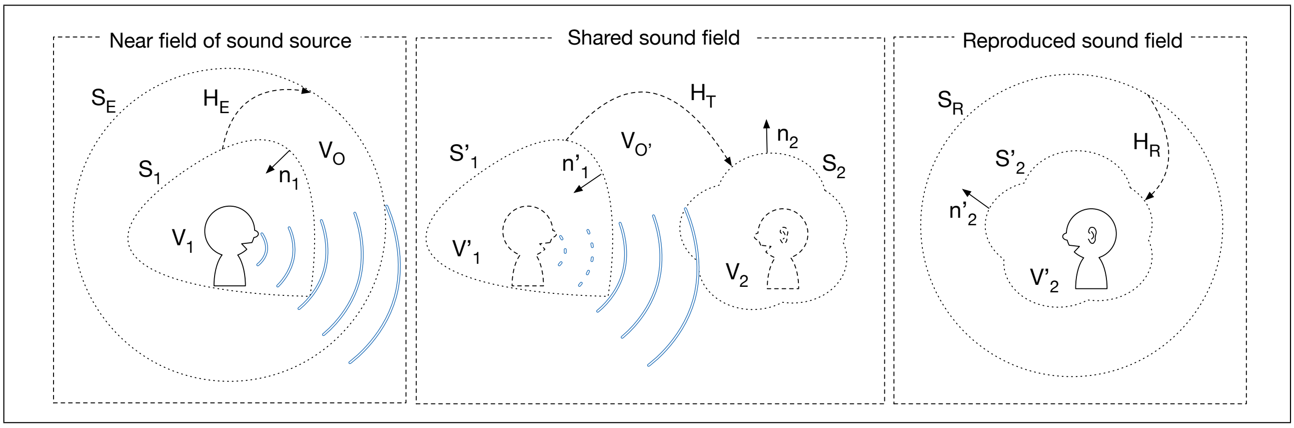

Figure 1. Concept of sound directivity reproduction

that are congruent with V_2 and S_2, respectively. On the basis of BoSC, the sound pressures in V_2' correspond to those in V_2 when the sound pressures on S_2' are matched with those on S_2 using secondary sources located on a surface S_R outside of S_2': $\mathbf{p}_2 = \mathbf{p'}_2$ where $\mathbf{p_2}$ and $\mathbf{p_2'}$ are the vector of the sound pressure in the small areas obtained by discretising the surfaces S_2 and S_2' into N_2, respectively.

The relationship between the sound pressures at the secondary sources and those on the surface S_2' is

$$\mathbf{p'}_2 = \mathbf{H}_\mathrm{R}\mathbf{p}_\mathrm{R}, \tag{8}$$

where $\mathbf{p}_\mathrm{R}$ is the column vector of the sound pressures in the small areas obtained by discretising the surface S_R into M_2, and $\mathbf{H}_\mathrm{R}$ is an $N_2 \times M_2$ matrix corresponding to the transfer matrix between the two surfaces.

Finally, we consider the relationship between the sound pressures on the surfaces S_1' and S_2 to reproduce the virtual sound source for the virtual listener. This relationship can be derived in the same way as equation 6, and the column vector of the sound pressures on the surface S_2 is

$$\mathbf{p}_2 = \mathbf{H}_\mathrm{T}\mathbf{p'}_1, \tag{9}$$

where $\mathbf{H}_\mathrm{T}$ is an $N_1 \times N_2$ matrix.

Therefore, from equations 7–9, we can obtain

$$\mathbf{p}_\mathrm{R} = \mathbf{H}_\mathrm{R}^{-1}\mathbf{H}_\mathrm{T}\mathbf{H}_\mathrm{E}^{-1}\mathbf{p}_\mathrm{E}. \tag{10}$$

That is, when we control the sound pressures at the secondary sources on the surface S_R to satisfy equation 10, the radiation of the sound source in the volume V_1 is reproduced in the volume V_1' and then reproduced in the volume V_2' after propagating in the shared sound field.

Well-known methods that trace the sound radiation back by solving the inverse problem are acoustical holography[8] and nearfield acoustical holography[7]. In this paper, an inverse problem that traces back to the sound source is applied to a telecommunication system using an immersive sound reproduction system based on BoSC. To apply it to a telecommunication system, we install secondary sources between the measurement surface and the sound source and solve the inverse problem using the measured impulse response matrix. Note that solving the inverse problem makes it easier to remove the effect of the characteristic features of the loudspeakers, microphones, and room acoustics.

2.2. System

In this section, we consider a telecommunication system based on the concept described in the preceding section. Figure 2 shows the system, which reproduces a speaker's or player's original sound directivity into the other party's system.

The sounds produced by the player are recorded with a microphone array that is installed so as to enclose the player. Let the signal recorded by the k-th microphone of the array be $s_k (k = 1, \ldots, N_1)$ in the time domain. First, the signals $[s_k]$ are convolved with the inverse matrix $[g_{kj}]^{-1}$ to estimate the radiation from the sound source. This inverse matrix is derived from an impulse response matrix $[g_{kj}] (j = 1, \ldots, M_1)$, which is measured using the microphone array, and M_1 secondary sound sources, which are placed so as to enclose the position of the original sound source. By this convolution, we obtain the signals of secondary sources to reproduce the radiation of the sound source.

Next, in the shared sound field, a loudspeaker array and a microphone array for BoSC sound reproduction are installed so as to enclose the position of a virtual player and a virtual listener, respectively. We measure an impulse response matrix $[h_{ij}] (i = 1, \ldots, N_2)$ from the loudspeaker array to the microphone array. On the basis of BoSC, we reproduce the sound field at the virtual listener into the area where a real listener is. Therefore, the reproduction of the sound field using the inverse system $[g'_{im}]^{-1}$ requires the loudspeaker

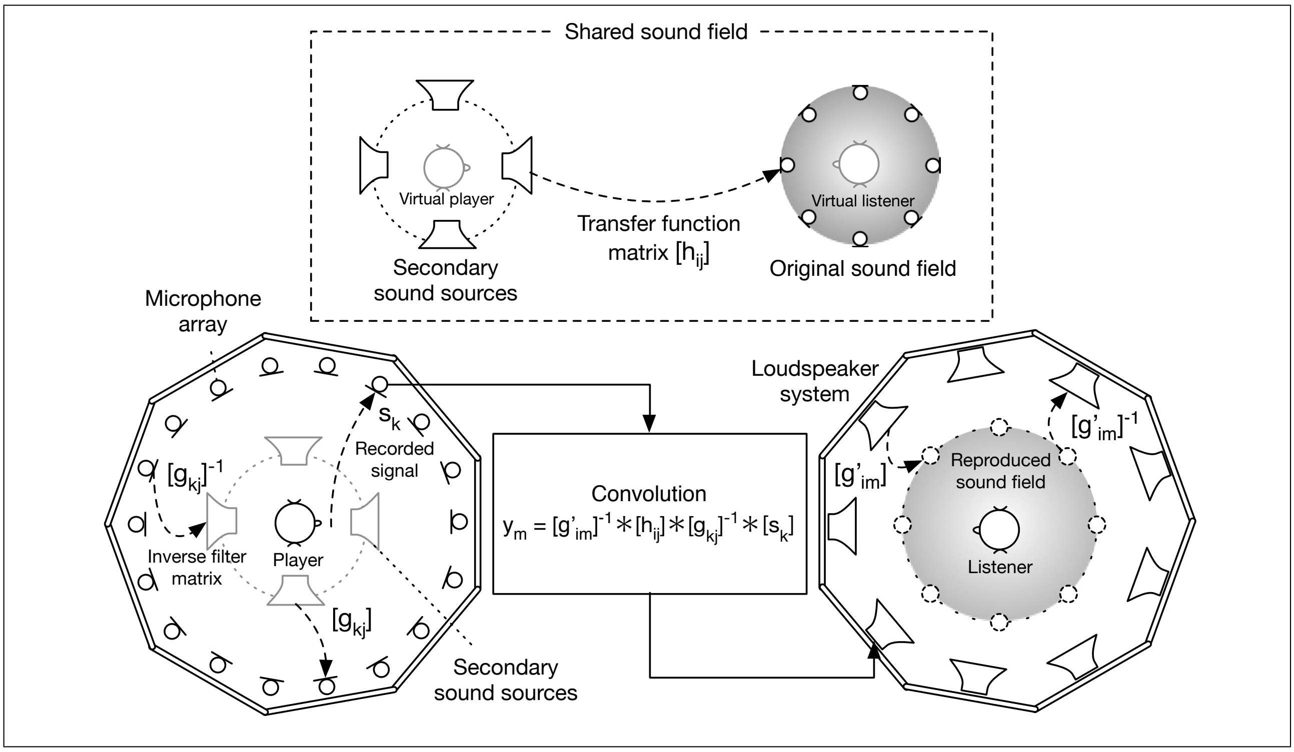

Figure 2. Sound directivity reproduction system for use in telecommunication

system signals $y_m (m = 1, \ldots, M_2)$ in the reproduction area to form the following equation:

$$y_m = [g'_{im}]^{-1} * [h_{ij}] * [g_{kj}]^{-1} * [s_k]. \tag{11}$$

Considering that the system is time invariant, and that secondary sources for the reproduction of the sound source's radiation are used only to measure impulse responses, the loudspeaker array can be replaced by a single loudspeaker that is moved to achieve the same result. The configuration of the secondary sources must be determined so as to control the sound pressures and particle velocities on the specific closed surface.

In this section, we consider only a one-way telecommunication system. Therefore, there is no feedback in the system. However, in an actual sound field sharing system, we must consider a feedback canceller[9].

3. Simulation of the free field

We verify the validity of our proposed method using a simulation in a free sound field.

In this paper, we consider only a horizontal plane. Figure 3 shows the positions of the sound sources and measurement points. The original sound source is set at the centre, and 18 observation points are placed at the side centres and vertices of a regular nonagonal array that inscribes a circle with a radius of 1.0 m and has the same size as the inside dimensions of the sound cask. Four hundred secondary sound sources

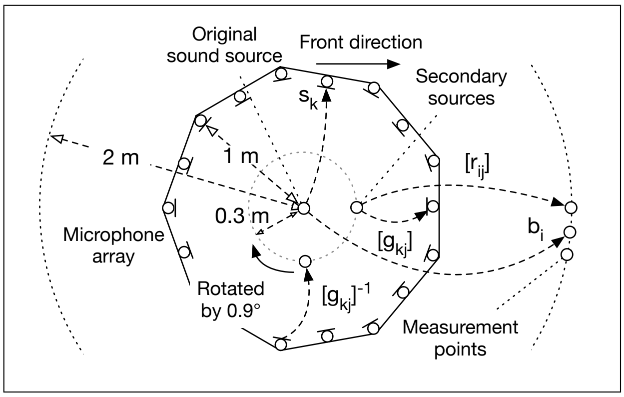

Figure 3. Conditions for simulations

are set in a circle of radius 0.3 m and directed toward the outside. These sound sources and measurement points are placed at the same height.

The amplitudes of the sound directivity of the original and secondary sound sources are represented by

$$Amp_\theta = \frac{1 + cos\theta}{2}, \tag{12}$$

where θ is the direction angle of the measurement point against the direction of the sound source. The direction angle of the secondary source in particular corresponds to the rotational angle.

Let the signal b_i be the recorded signal from the original sound source at the i-th measurement point.

We calculate the reproduced signals b'_i, which are represented by

$$b'_i = [r_{ij}] * [g_{kj}]^{-1} * [s_k], \qquad (13)$$

where $[r_{ij}]$ is the impulse response matrix from the secondary sources to the measurement points, and $[g_{kj}]^{-1}$ is the inverse matrix between the secondary sources and the observation points. Using the regularisation method, we calculate the inverse filter matrix $[g_{kj}]^{-1}$ from the impulse response matrix $[g_{kj}]$. The sampling rate of all the signals is 48 kHz, and the length of the inverse filter matrix is 4096 points. The measurement points are set in a circle and separated by 5° intervals.

The inverse filter matrix is calculated in the frequency domain with a Fourier transform of length 8192. In the time domain, the inverse filter matrix is multiplied by the Hanning function to shorten it to a sample length of 4096. The SNR of the inverse filter is defined as

$$\begin{aligned} SNR_{inv} =& 10\log_{10}\frac{M\sum_{n=1}^{N}|\delta(n-L)|^2}{\sum_{k=1}^{M}\sum_{n=1}^{N}|\delta(n-L)-y_k[n]|^2}, \\ y_k[n] =& [g_{kj}] * [\hat{f_{jk}}], \end{aligned} \qquad (14)$$

where δ is a delta function, $[\hat{f_{ij}}]$ is an inverse filter matrix calculated using the regularisation method, L is the half-length of the inverse filter, and N and M is the signal length and the number of the observation points, respectively. In this experiment, the SNR of the calculated inverse filter is 34.0 dB.

The relative sound pressure level (SPL) is defined as

$$RSPL_i[dB] = 10\log_{10}\frac{\sum^N |b_i[n]|^2}{\sum^N |b_{i'}[n]|^2}, \qquad (15)$$

where the index i is a measurement point index, and the index i' refers to the measurement point facing the original source direction; $N(=8192)$ is the signal length for evaluation.

Figure 4 shows the relative SPLs at measurement points positioned at a 2.0 m distance. In this case, the original sound source is directed toward the front of the BoSC microphone array. We found as many peaks and dips as the number of observation points in this figure. However, the difference in the relative SPLs between the original and reproduced signals in the front 270° is less than 3.0 dB. Further, the mean of the difference is 1.0 dB.

Similarly, figures 5 and 6 show the results for other directions of the original sound source, where the original sound source is rotated clockwise by 45° and 90°, respectively. These reproduced directivities have features similar to the 0° angle sound source. Each difference in the relative SPLs between the original and reproduced signals in the front 270° of the each speaker's direction is less than 2.1 dB and 4.3 dB, respectively. Further, the means of the difference are 0.8 dB and 1.8 dB. For a source direction angle of 90°, the difference between the SPLs is largest because the basis measurement points are placed at the dip of the SPLs.

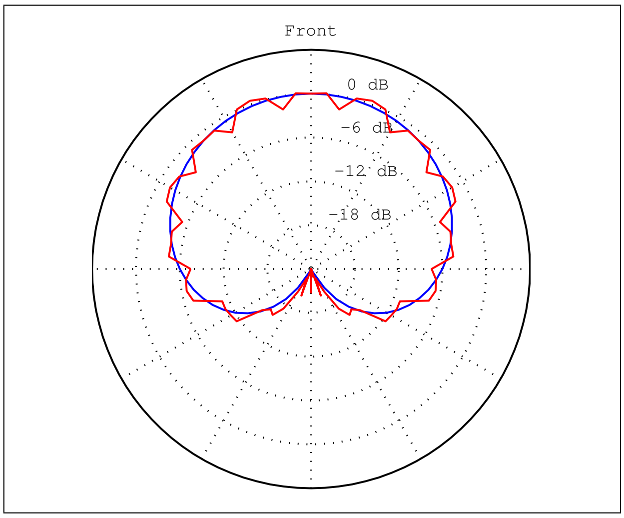

Figure 4. Simulation of sound directivity reproduction (Original source direction: 0°, measurement radius: 2.0 m, original: blue line, reproduced: red line)

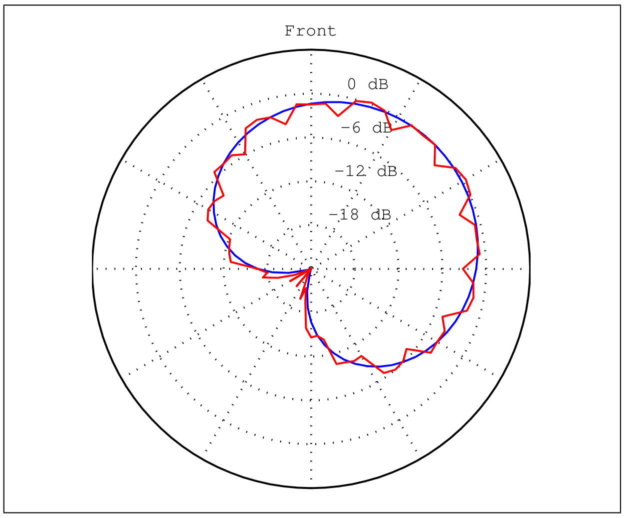

Figure 5. Simulation of sound directivity reproduction (Original source direction: 45°, measurement radius: 2 m, original: blue line, reproduced: red line)

For all the original sound source directions, the differences in the relative SPLs are less than 4.3 dB, and the mean of the differences is 1.1 dB in the front 270° of the speaker's direction.

Next, we consider the difference in the distance between measurement points. Figure 7 shows the result for measurement points at a 3.0 m distance. We found a difference in the positions of the peaks and dips between figs. 4 and 7. Figure 8 shows the result for measurement points at a 0.5 m distance. In this case, the measurement points are placed between the sec-

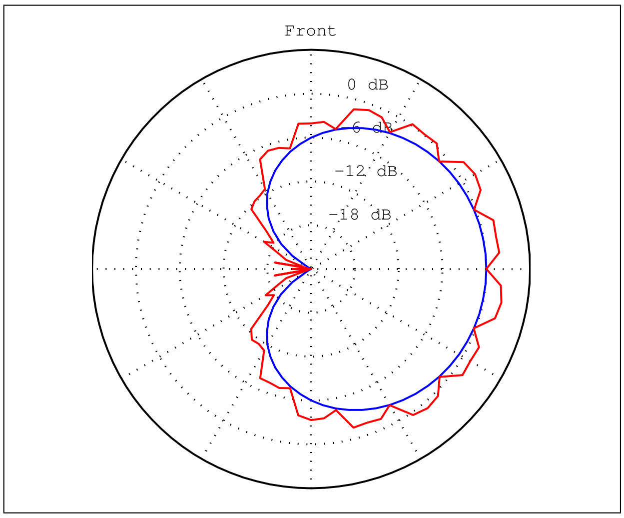

Figure 6. Simulation of sound directivity reproduction (Original source direction: 90°, measurement radius: 2 m, original: blue line, reproduced: red line)

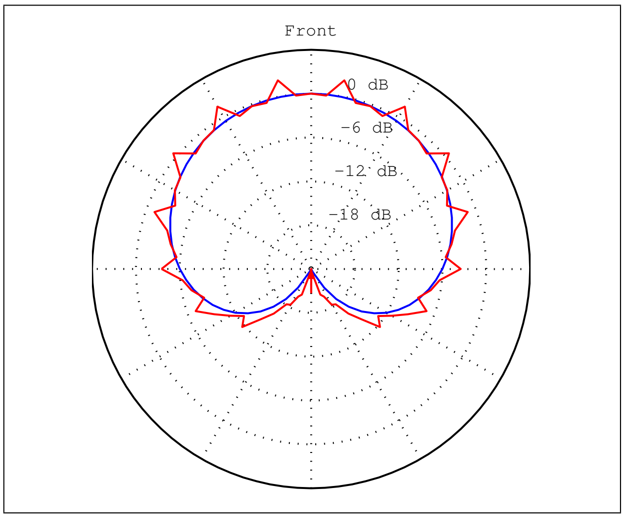

Figure 8. Simulation of sound directivity reproduction (Original source direction: 0°, measurement radius: 0.5 m, original: blue line, reproduced: red line)

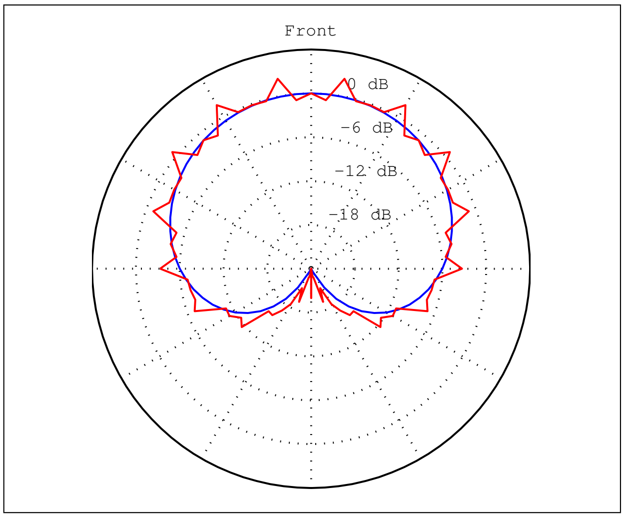

Figure 7. Simulation of sound directivity reproduction (Original source direction: 0°, measurement radius: 3.0 m, original: blue line, reproduced: red line)

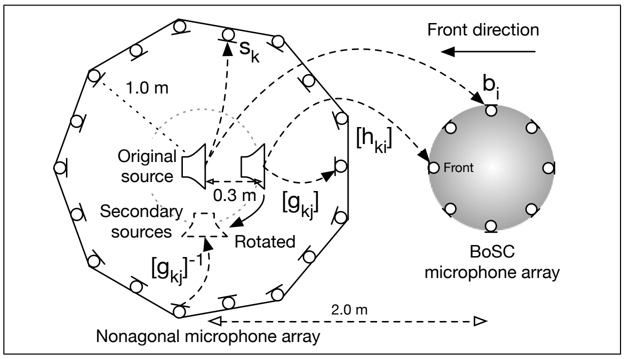

Figure 9. Conditions for experiments in an anechoic chamber

ondary sources and the observation points. The figure shows that there is no significant difference between the inside and the outside of the observation surface. Similarly, in these two cases, the relative SPL difference between the original and reproduced signals in the front 270° is less than 3.0 dB.

Consequently, we found that our proposed method can reproduce the original sound directivity independently of the original sound source's direction and the measurement positions.

4. Experimental sound directivity reproduction

We evaluate the proposed method with impulse responses measured in an anechoic chamber (Chiba, Tokyo Denki University). Figure 9 shows the measurement conditions, such as the positions of the sound sources and microphones. A loudspeaker (TOA BST-246) that produces the original sound directivity is placed at the centre and surrounded by a regular nonagonal microphone array. The regular nonagonal microphone array is the size of a regular nonagon inscribed in a circle with a radius of 1.0 m. Its size is also the same as the inside dimensions of the sound cask. Eighteen omnidirectional microphones (DPA 4060) are installed at the vertices and side centres of the regular nonagon. Identical loudspeakers are used for both the secondary and original sound sources. We measured the transfer functions at a total of 400 secondary source positions by rotating the loudspeaker by 0.9° in a circle with a radius of 0.3 m. The measurement points used for evaluation are positioned at a 2.0 m distance from the centre, and an 80-ch fullerene-shaped microphone array (the BoSC microphone array) is installed at the points. The BoSC microphone array is about 0.46 m in diameter, and it is positioned facing toward the sound source.

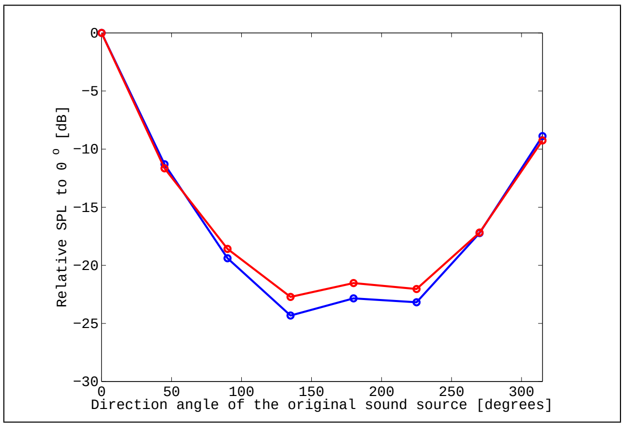

Figure 10. Relative SPLs at the front of the BoSC microphone array (Blue: original, red: reproduced)

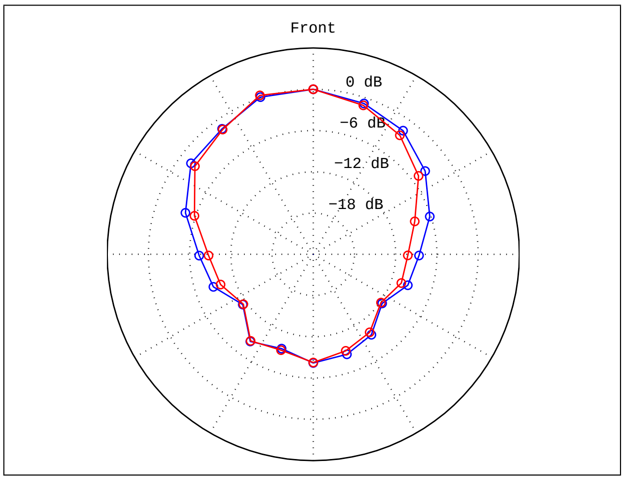

Figure 11. Relative SPLs of the BoSC microphone array (Original speaker direction: 0°, blue: original, red: reproduced)

For the original sound field, we measured the impulse responses from a loudspeaker at the centre to the regular nonagonal microphone array and the BoSC microphone array by rotating the loudspeaker by 45°.

We measured the impulse responses between the secondary sources and the nonagonal microphone array in three times using a sweep signal. We derived the inverse system $[g_{ji}]^{-1}$ using a transfer function matrix obtained in the first measurement. The inverse system was derived using the regularisation method. The transfer functions obtained in the second measurement were used to optimise the regularisation parameters.

We measured all the impulse responses with 48 kHz sampling. Then, we use 2048-sample-length impulse responses to calculate the inverse filter matrix. The SNR of the calculated inverse matrix is 26.5 dB according to equation 14.

The microphones of the fullerene-shaped microphone array were used for evaluation by comparing the signal $[b_i]$ of the original sound and the signal $[b_i']$ of the reproduced sound with the inverse system on the basis of the sound directivity reproduction:

$$b_i' = [h_{ij}] * [g_{kj}]^{-1} * [s_k]. \tag{16}$$

When a microphone index at the front of the BoSC microphone array is zero, the relative sound pressure level $RSPL_\alpha(\alpha = 0, 45, 90, ..., 315[°])$ is defined as

$$RSPL_\alpha[dB] = 10\log_{10}\frac{\sum^N |b_{0,\alpha}[n]|^2}{\sum^N |b_{0,\mathrm{front}}[n]|^2}, \tag{17}$$

where $b_{0,\alpha}$ is the signal of the sound source directed at angle α, and $b_{0,\mathrm{front}}$ is the signal of the sound source directed toward the front of the BoSC microphone array.

Figure 10 shows the change in the relative SPL of a single microphone placed in the front of the BoSC microphone array by rotating the original sound source by 45°. The figure shows that the relative SPLs of both the original and the reproduced signals decrease as the direction angle of the original sound source approaches 180°. The difference between the SPLs of the original and reproduced signals is less than 1.6 dB.

Next, we compared the sound directivities for the BoSC microphone array by using 20 microphones positioned in the middle part of the array and facing in 10 directions. The relative SPL of the microphone directed toward the front is defined as

$$RSPL_m[dB] = 10\log_{10}\frac{\sum^N |b_i[n]|^2 + |b_{i'}[n]|^2}{2\sum^N |b_0[n]|^2}, \tag{18}$$

where the microphone indices i and i' have the same direction, and index m refers to the microphone direction.

Figure 11 compares the relative SPLs between the original and reproduced signals when the original sound source faces the BoSC microphone array. Because the sound source is located at the front of the BoSC microphone array, we measured the maximum SPL at 0° of the BoSC microphone array. Because of the shape and directivity of the BoSC microphone array, the relative SPL decreases toward 180°. However, the relative SPL of the reproduced signal changes in the same way as the original signal. The difference between these SPLs in each direction is less than 2.3 dB.

Figures 12 and 13 show the results for other directions of the original sound source, where the original sound source is rotated clockwise by 45° and 90°, respectively. That is, when viewed from the BoSC microphone array, the sound source is directed toward the left side. At this directivity of the original sound source, the difference in the relative SPL between the front and the back of the array is smaller than that for a sound source with a direction of 0°. The differences

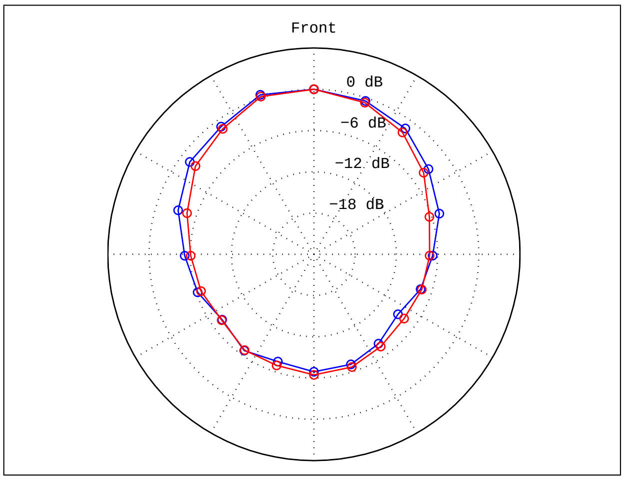

Figure 12. Relative SPLs of BoSC microphone array (Original speaker direction: 45°, blue: original, red: reproduced)

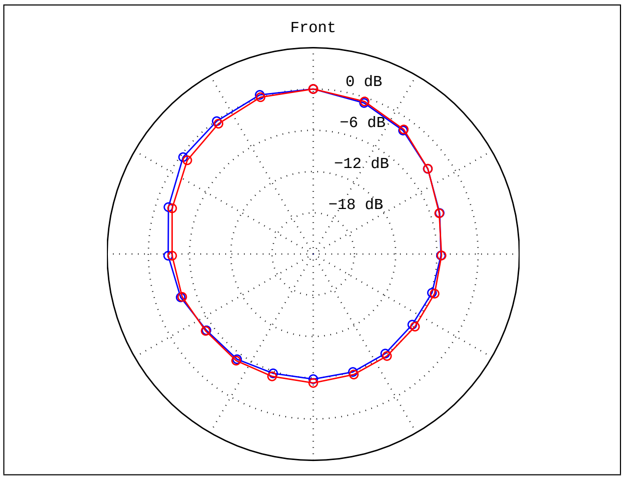

Figure 13. Relative SPLs of BoSC microphone array (Original speaker direction: 90°, blue: original, red: reproduced)

between the SPLs of the original and reproduced signals in each direction are less than 1.5 dB and 0.8 dB, respectively. Similarly, when we change the original source direction from 0° to 315° in 45° increments, the difference between their SPLs in each direction is less than 3.5 dB.

Consequently, our proposed method can accurately reproduce the directivity regardless of the original sound source direction.

5. CONCLUSIONS

We proposed a method of sound source directivity for telecommunication based on BoSC. An evaluation in a free-field simulation showed that our method can reproduce the sound directivity inside and outside the observation surface in a horizontal plane. Experiments in an anechoic chamber demonstrated that our method can reproduce the original sound directivity at a microphone array located at a 2 m distance.

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