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Three-dimensional resonance control based on spatial wave synthesis with parametric array loudspeaker

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ABSTRACT

A parametric array loudspeaker (PAL) realizes a sharper directivity utilizing a demodulation of an intense amplitude modulated (AM) wave. The demodulation is modeled as spatial sound sources generated with the demodulated audible sound in the acoustic beam. Spatial sound sources form resonance and anti-resonance positions because of the interference between spatial sound sources demodulated with the non-linear interaction in the air. Thus, the sound pressure level (SPL) of the PAL is reduced at anti-resonance positions. However, it suggests that resonance positions might be reformed by spatial wave synthesis with phased array processing. Therefore, we propose three-dimensional resonance control based on spatial wave synthesis with the PAL. In this paper, we employ the PAL consists of line-type PALs corresponding to column. The PAL can apply different delay filters to each column. The proposed method designs inverse filters with a multichannel adaptive algorithm as delay filters for line-type PALs under the constraint condition that the SPL is maximized by spatial wave synthesis at the target position. The proposed method enables to influence the non-linear interaction and improve the SPL at the target position by applying designed filters to each column. Finally, we confirmed the effectiveness of the proposed method through evaluation experiments.

Keywords: Parametric array loudspeaker, Resonance control, Spatial wave synthesis

1. INTRODUCTION

Recently, an electrodynamic loudspeaker is widely used to emit audible sound such as music and voice. The electrodynamic loudspeaker has a wider directivity and can transmit the audible sound to a wide area. However, it is difficult for the electrodynamic loudspeaker to emit the audible sound to an only target listener who requires the audible sound. Therefore, the audible sound emitted by the electrodynamic loudspeaker becomes a noise for the non-listener [1]. To solve this problem, a parametric array loudspeaker (PAL) [2, 3] which has a sharper directivity has recently been focused on. The PAL realizes the sharper directivity by utilizing an ultrasonic wave. The PAL emits an intense amplitude modulated (AM) wave which is designed by modulating the ultrasonic wave (carrier wave) with an audible sound. The emitted intense AM wave from the PAL is gradually demodulated into the original audible sound by the non-linear interaction in the process of propagation in the air [4]. In the process of the demodulation, the PAL generates spatial sound sources with the demodulated audible sound and constructs a vertical array in the acoustic beam with the non-linear interaction. Therefore, the PAL can realize the sharper directivity and emits the audible sound in a particular area [4, 5, 6].

The sound pressure level (SPL) of the audible sound emitted from the electrodynamic loudspeaker decay by distance right after it is emitted. On the other hand, the SPL of the demodulated audible sound gradually increases with the occurring demodulation. The demodulated audible sound has the maximum SPL at the distance which ceases the demodulation. After ceasing the demodulation, the SPL of the demodulated audible sound decays by distance. Furthermore, spatial sound sources form

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resonance and anti-resonance positions of the demodulated audible sound in the acoustic beam because of the interference between spatial sound sources demodulated with the non-linear interaction. Thus, the SPL of the demodulated audible sound is attenuated at anti-resonance positions. To control the resonance positions, we propose the method to reform them by spatial wave synthesis with phased array processing. In the preliminary experiment, we confirmed that resonance positions can be reformed by emitting AM waves imparted different delays from each line-type PAL constituting the PAL. It is considered that imparting delays to each column influences the composite waves demodulated by the non-linear interaction in the air. In addition, we confirmed that the SPL of the demodulated audible sound increases in the preliminary experiment. The proposed method designs inverse filters with a multichannel adaptive algorithm as delay filters for line-type PALs to improve the SPL of the demodulated audible sound at the target position. We design filters under constraint condition that the SPL of the demodulated audible sound is maximized by spatial wave synthesis. By applying designed filters which control phases to each column, we change the sound pressure distribution of resonance positions and improve the SPL at the target position. As a result of the evaluation experiment, we confirm the effectiveness of the proposed method.

2. PARAMETRIC ARRAY LOUDSPEAKER

2.1 Principle of Parametric Array Loudspeaker

The PAL realizes a sharper directivity by utilizing an ultrasonic wave and a parametric array effect. Generally, the high frequency wave has the straightness than the low frequency wave. For this reason, the PAL utilizing the ultrasonic wave, which has high frequency, can transmit the sound straightly. In addition, the PAL utilizes the parametric array effect by consisting of multiple small ultrasonic transducers with close-packed structure. When AM waves are emitted from each ultrasonic transducer simultaneously, they are synthesized spatially and realize the directivity. The PAL emits an intense AM wave which is designed by modulating the carrier wave with an audible sound. Figure 1 shows the overview of the demodulation of the emitted AM wave in time domain. The carrier wave $v_{\mathrm{c}}(t)$ and the audible sound $v_{\mathrm{s}}(t)$ are indicated as follows:

$$v_{\mathrm{C}}(t) = A_{\mathrm{C}} \cos(2\pi f_{\mathrm{C}} t), \tag{1}$$

$$v_{\mathrm{S}}(t) = A_{\mathrm{S}} \cos(2\pi f_{\mathrm{S}} t), \tag{2}$$

where t represents a time index, A_{c} and A_{s} represent the maximum amplitudes of the carrier wave and the audible sound, f_{c} and f_{s} represent the frequencies of the carrier wave and the audible sound. The AM wave $v_{\mathrm{AM}}(t)$ is designed by modulating the carrier wave $v_{\mathrm{c}}(t)$ with the audible sound $v_{\mathrm{s}}(t)$. From Eq. (1) and (2), the AM wave $v_{\mathrm{AM}}(t)$ is indicated as follows:

$$v_{\mathrm{AM}}(t) = \{1 + m \cdot v_{\mathrm{S}}(t)\} v_{\mathrm{C}}(t), \tag{3}$$

$$m = A_{\mathrm{S}} / A_{\mathrm{C}}, \tag{4}$$

where m $(0 < m \leq 1)$ is an amplitude modulation index. Equation (3) can be transformed as follows by trigonometric functions.

$$\begin{aligned} v_{\mathrm{AM}}(t) &= \{1 + m \cdot v_{\mathrm{S}}(t)\} v_{\mathrm{C}}(t) \\ &= \{1 + m A_{\mathrm{S}} \cos(2\pi f_{\mathrm{S}} t)\} A_{\mathrm{C}} \cos(2\pi f_{\mathrm{C}} t) \\ &= A_{\mathrm{C}} \cos(2\pi f_{\mathrm{C}} t) + m A_{\mathrm{S}} A_{\mathrm{C}} \cos(2\pi f_{\mathrm{S}} t) \cos(2\pi f_{\mathrm{C}} t) \\ &= A_{\mathrm{C}} \cos(2\pi f_{\mathrm{C}} t) + m A_{\mathrm{S}} A_{\mathrm{C}} \cos\{2\pi (f_{\mathrm{C}} + f_{\mathrm{S}}) t\}/2 + m A_{\mathrm{S}} A_{\mathrm{C}} \cos\{2\pi (f_{\mathrm{C}} - f_{\mathrm{S}}) t\}/2. \end{aligned} \tag{5}$$

From Eq. (5), we can confirm that the AM wave consists of the carrier wave (frequency: f_{C}), the higher sideband wave (frequency: $f_{\mathrm{C}} + f_{\mathrm{S}}$) and the lower sideband wave (frequency: $f_{\mathrm{C}} - f_{\mathrm{S}}$). When the intense AM wave as a primary wave is emitted from the PAL, the sum and difference frequency waves of the carrier wave and sideband waves are secondarily demodulated along the primary beam in the process of propagation in the air due to the non-linear interaction. Generally, the high frequency wave decay more rapidly than the low frequency wave because the sound absorption increases with frequency. Therefore, only the difference frequency wave, which has low frequency and is perceived by humans, can be transmitted an appreciable distance and remain sufficient amplitudes than the high frequency wave such as the carrier wave.

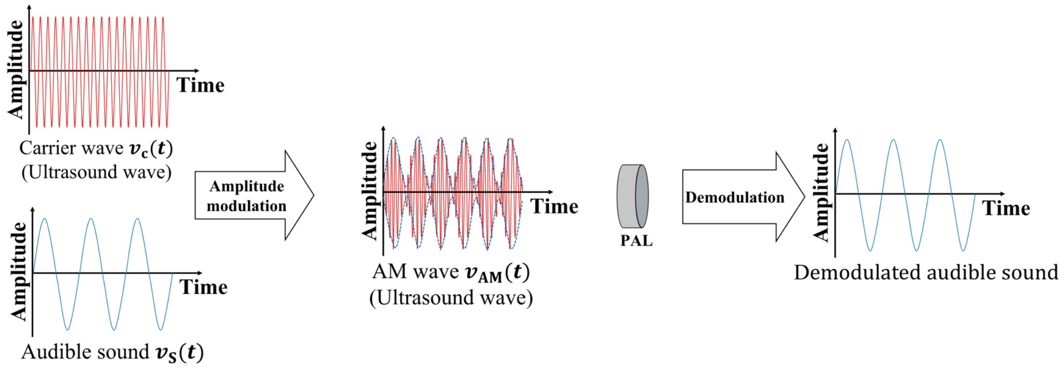

Figure 1 – Overview of the demodulation of the emitted AM wave (Time domain).

2.2 Generation of Virtual Sound Source

Figure 2 shows the generation model of a parametric array [1]. As mentioned in the previous section, the intense AM wave as a primary beam emitted from the PAL is demodulated in the process of the propagation in the air due to the non-linear interaction. In the primary beam, secondary spatial sound sources of the difference frequency wave are virtually created and distribute along a narrow beam. This generation model is referred as the parametric array. In the parametric array, each spatial sound source operates as a point sound source. Therefore, they form resonance and anti-resonance positions in the acoustic beam because of the interference between spatial sound sources demodulated with the non-linear interaction in the air. Figure 3 shows the relationship between the distance and the SPL of the demodulated audible sound. In the Fig. 3, a horizontal axis shows the distance from the PAL and a vertical axis shows the power of the demodulated audible sound. From Fig. 3, we can confirm that the several resonance and anti-resonance positions are generated and the power of the demodulated audible sound mottles in the audible area. Hence, the listener who stands at the anti-resonance position may obtain the demodulated audible sound with insufficient power. In this paper, we propose the method to control the generation process of resonance positions and improve the SPL of the demodulated audible sound at the target position.

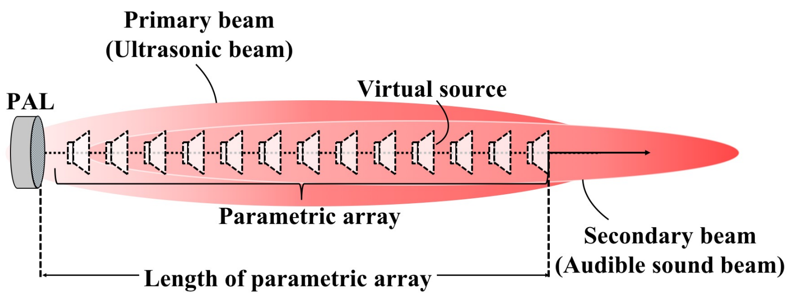

Figure 2 – Generation model of parametric array.

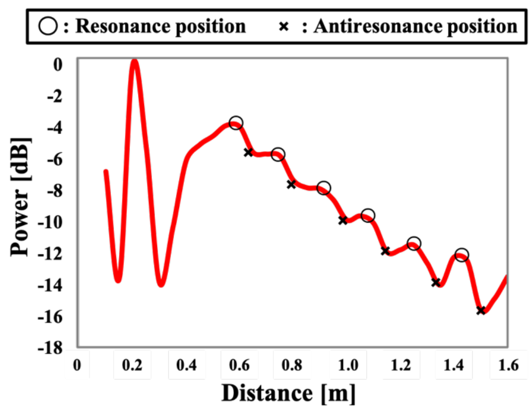

Figure 3 – Relationship between the distance and the SPL of the demodulated audible sound.

3. PROPOSAL OF THREE-DIMENSIONAL RESONANCE CONTROL BASED ON SPATIAL WAVE SYNTHESIS WITH PARAMETRIC ARRAY LOUDSPEAKER

3.1 Overview of Proposed Method

In this paper, we propose the method to control three-dimensional resonance positions of the demodulated wave. We reform resonance positions by spatial wave synthesis with phased array processing with the PAL. Figure 4 shows the overview of the proposed method. In Fig. 4, $x(t)$ shows the input signal and $g_i(t)(i = 1, 2, \cdots, M)$ shows the transfer function, where i represents a column index and M is a number of line-type PALs. $h_i(t)(i = 1, 2, \cdots, M)$ shows the inverse filter. Figure 5 shows the PAL which we use in this paper. In Fig. 4 and 5, $S_i(i = 1, 2, \cdots, M)$ shows the line-type PAL of ith column. The PAL is consisted of some line-type PALs corresponding to column and enables array processing. In the proposed method, we design some delay filters for each column to improve the SPL of the demodulated audible sound at the target position. By imparting delays to each column, it is possible to influence the composite waves demodulated by non-linear interaction in the air. As delay filters, the proposed method designs inverse filters for line-type PALs under the constraint condition that the SPL is maximized by spatial wave synthesis at the target position. The inverse filter has the inverse characteristics of a system. In this paper, we utilize inverse filters in order to correct transfer functions between each line-type PAL and the target position. It is possible to change the phase of the sound emitted from some line-type PALs by designing inverse filters in consideration of phases.

In the proposed method, we first measure transfer functions $g_i(t)$ between each line-type PAL and the target position. Transfer functions can be measured using the TSP method [8]. Next, in the Step2, we design inverse filters $h_i(t)$ with a multichannel adaptive algorithm in order to add different delays to each line-type PAL. Details on calculation of the inverse filter coefficient are described in the next section. The filtered input signal $v_{\mathrm{S}_i}(t)$ is indicated as follows:

$$v_{S_i}(t) = x(t) * h_i(t), \tag{12}$$

where $*$ means convolution operation. In the proposed method, we modulate the carrier wave $v_{\mathrm{AM}_i}(t)$ with the filtered input signal $v_{\mathrm{S}_i}(t)$ and emit the modulated wave $v_{\mathrm{AM}_i}(t)$ from each line-type PAL. The modulated wave $v_{\mathrm{AM}_i}(t)$ is indicated as follows:

$$v_{AM_i}(t) = \{1 + m \cdot v_{S_i}(t)\} v_{\mathrm{C}}(t). \tag{13}$$

By emitting the AM wave applying designed filters which control phases to each column, we enable to influence the non-linear interaction and change the process of the spatial wave synthesis. Therefore, it becomes to change the sound pressure distribution of resonance positions and improve the SPL at the target position.

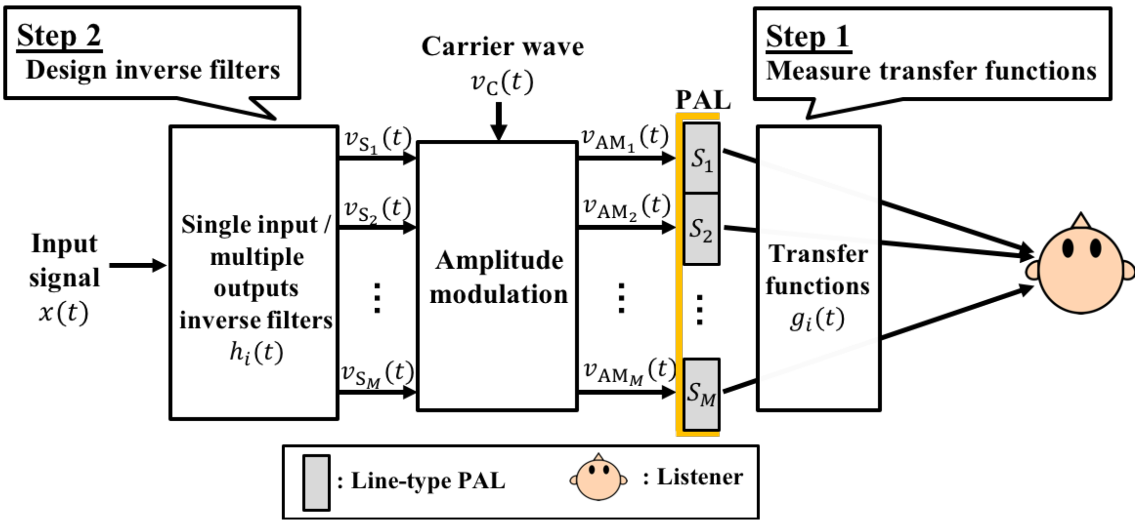

Figure 4 – Overview of the proposed method.

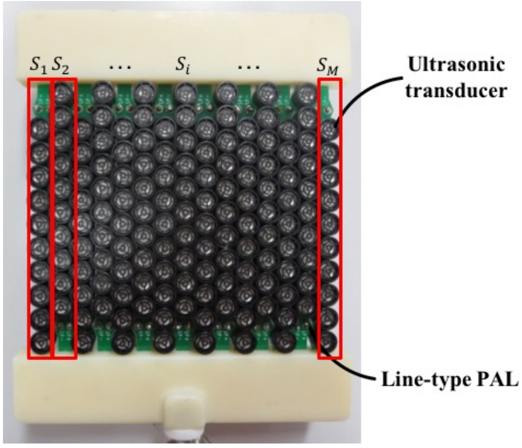

Figure 5 – PAL consisted of some line-type PALs.

3.2 Calculation of inverse filter coefficient

The proposed method utilizes the NLMS for the adaptive algorithm [9]. A filter design procedure is following:

i. Calculate the input signal $r_i(t)(i = 1, 2, \cdots, M)$ of the adaptive algorithm as follows:

$$r_i(t) = x(t) * g_i(t). \tag{6}$$

ii. Set initial coefficient $h_i(0)$ as time $t = 0$.

iii. Calculate the output signal $y(t)$ and error signal $e(t)$ as follows:

$$y(t) \approx \sum_{i=1}^{M} \boldsymbol{h}_i(t)^T \boldsymbol{r}_i(t), \tag{7}$$

$$\mathrm{e}(t) = d(t) - y(t), \tag{8}$$

where $\boldsymbol{r}_i(t)$ and $\boldsymbol{h}_i(t)$ are given as follows:

$$\boldsymbol{r}_i(t) = [r_i(t), r_i(t-1), \cdots, r_i(t-L+1)]^T, \tag{9}$$

$$\boldsymbol{h}_i(t) = \left[h_{i,1}(t), h_{i,2}(t), \cdots, h_{i,L}(t)\right]^T, \tag{10}$$

where L represents the length of the filter coefficient. In the proposed method, $e(t)$ is calculated by the output single $y(t)$ and the desired signal $d(t)$. The output signal $y(t)$ in Eq. (9) is the approximation of the output signal emitted from the PAL.

iv. Estimate the coefficient $\boldsymbol{h}_i(t+1)$ as follows:

$$\boldsymbol{h}_i(t+1) = \boldsymbol{h}_i(t) + \alpha e(t) r_i(t) / (\boldsymbol{r}_i(t)^T \boldsymbol{r}_i(t) + \beta), \tag{11}$$

where α represents the step size which controls the size of the coefficient correction, β represents the small positive constant which prevents denominator in Eq. (11) from becoming zero.

v. Increment a variable by 1 and repeat procedures ⅲ and ⅴ.

4. EVALUATION EXPERIMENT

4.1 Experimental Condition

We carried out the objective evaluation experiment to confirm the effectiveness of the proposed method. The proposed method designs some filters for each column to improve the SPL of the demodulated audible sound at the target position. We compared the SPL of the demodulated audible sound in the case of the conventional method which emits AM waves which have coordinate phase and the case of the proposed method. Table 1 shows the experimental conditions in the evaluation experiment, Table 2 shows the experimental equipment and Fig. 7 shows the experimental arrangement. Table 3 shows the conditions for inverse filter design and Fig. 6 shows the experimental arrangement for inverse filter design. In this experiment, we set the desired position at the 0.8 m spot from the PAL as shown in Fig. 6. We utilize the NLMS for adaptive algorithm, add the delay to the desired signal to satisfy causal relation, and set each parameter experimentally. In addition, we utilize the PAL as shown in Fig. 5. The PAL is consisted of fifteen line-type PALs corresponding to column and each of them has ten small ultrasonic transducers. However, the propose method designs inverse filters for the five channel PAL in consideration of easy convergence. Figure 8 shows the five channel PAL utilized in the evaluation experiment. We realize the five channel PAL by dividing the PAL into five as shown in Fig. 8.

We expect that the proposed method changes the sound pressure distribution of resonance positions and improves the SPL at the target position. The SPL of the demodulated audible sound P_{ave} is calculated as follows:

$$P_{\mathrm{ave}} = \left(\sum_{f=f_a}^{f_b} P(f) \right) \Big/ (f_b - f_a) \tag{14}$$

where f_a and f_b represent the maximum and minimum frequency of the evaluation frequency band and $P(f)$ is the SPL of the frequency f. Moreover, the evaluated frequency band is 1 ± 0.01 kHz because we evaluate the pure tone.

Table 1 – Experimental conditions for measuring the sound pressure distribution.

Sampling frequency	96 kHz
Quantization	16 bits
Carrier frequency	40 kHz
Ambient noise level	$L_{\mathrm{A}} = 19.2$ dB
Environmental	Soundproof room($T_{60} = 0.15$ s)
Sound source	White noise (0~8 kHz)
Evaluation distance	0.30～1.30 m (0.02 m spacing)

Table 2 – Experimental equipment.

Ultrasonic transducer	SPL (Hong Kong) Limited, UT1007-Z325R
Power amplifier	VICTOR, PS-A2002
A/D, D/A converter	RME, FIREFACE UFX
Microphone amplifier	THINKNET, MA-2016C
Microphone	SONY, ECM-88B

Table 3 – Experimental conditions for designing inverse filters.

Adaptive algorithm	NLMS
Filter length	3840 points
Step size	$\alpha = 0.10000$
Parameter	$\beta = 0.00001$
Desired signal	White noise (0～8 kHz)

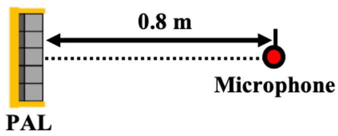

Figure 6 – Experimental arrangement for designing inverse filter.

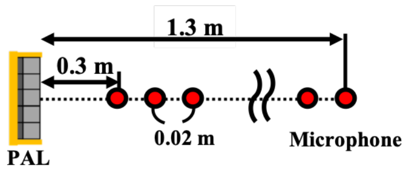

Figure 7 – Experimental arrangement for measuring the sound pressure distribution.

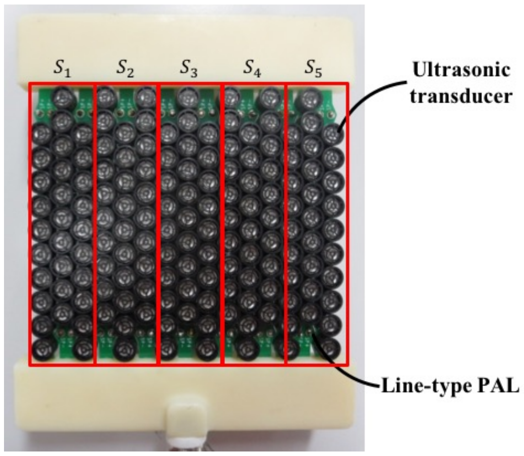

Figure 8 – Channel segment of the PAL in the evaluation experiment.

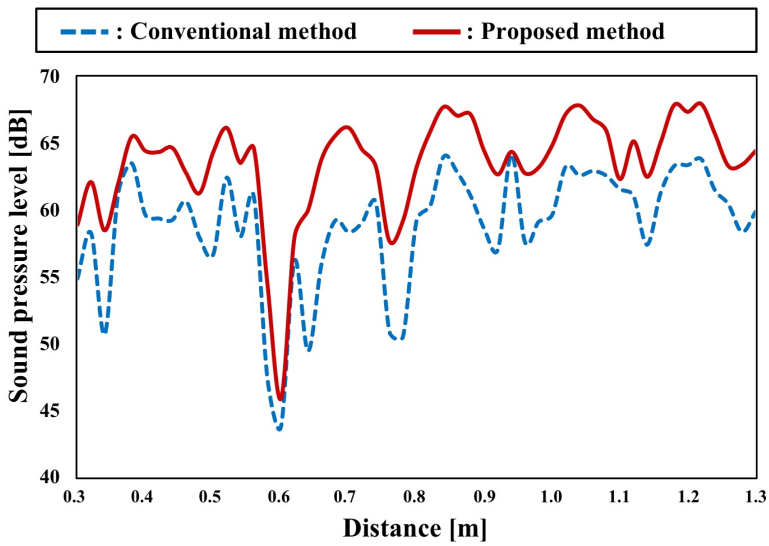

Figure 9 – SPLs of the demodulated audible sound.

4.2 Experimental Result

Figure 9 shows the experimental result. In Fig. 9, a horizontal axis shows the distance from the PAL and a vertical axis shows the power of the demodulated audible sound (1kHz).

As shown in Fig. 9, the SPL of the demodulated audible sound at the target position (0.8 m spot from the PAL) is improved about 4.1 dB compared to the conventional method. This is because inverse filters are designed to correct transfer functions between each line-type PAL and the target position.

In Fig. 9, it is also confirmed that the sound pressure distribution of the resonance positions has changed compared to the conventional method. At the 0.65 m and 0.85 m spots from the PAL, the anti-resonance position of the demodulated wave, which existed in the conventional method, becomes the resonance positions in the proposed method and the SPL of the demodulated audible sound is improved. It is considered that the process of spatial wave synthesis has changed by applying the inverse filters to each line-type PAL and resonance positions have reformed. From the evaluation experimental result, we confirmed the effectiveness of the proposed method.

5. CONCLUSIONS

The PAL realizes a sharper directivity and can transmit the audible sound to only the particular area by utilizing AM wave. However, in demodulation process, some spatial sound sources are generated in the acoustic beam and form some resonance and anti-resonance positions of the demodulated wave because of the interference between spatial sound sources. Thus, the SPL of the demodulated audible sound is attenuated at anti-resonance positions. In this paper, we propose the method to control three-dimensional resonance positions of the demodulated wave. The proposed method reforms resonance positions by spatial wave synthesis with phased array processing. We design inverse filters with a multichannel adaptive algorithm as delay filters for line-type PALs to improve the SPL of the demodulated audible sound at the target position. We conducted the evaluation experiment to confirm the effectiveness of the proposed method. From the result of the evaluation experiment, we confirmed that the proposed method can improve the SPL of the demodulated audible sound at the target position and change the sound pressure distribution of the resonance positions. In the future work, we intend to reconsider the filter design method to change only phase characteristic of inverse filters.

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