Multi-channel Feedforward ANC System Combined with Noise Source Separation

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Abstract—**In this paper, we examine the effectiveness of Multi-channel feedforward active noise control (ANC) system combined with noise source separation through some simulations. Multi-channel feedforward ANC system can reduce various noise such as the broad-band noise by arranging the reference microphones close to noise sources. However, the performance of ANC system deteriorates when the reference microphones cannot be arranged close to noise sources because each microphone catches undesired noise in addition to desired noise. The proposed system separates noises by using microphone arrays as reference microphones and the separated signals are used as the reference signals of the ANC system. In simulation results, the proposed system can improve the noise reduction performance in case where the ANC system uses two reference microphones and the reference microphones do not need to be arranged close to the noise sources.**

I. INTRODUCTION

Active noise control (ANC) is an effective approach to reduce the noise level at the noise control point using anti-noise [1]–[4]. Anti-noise with the same amplitude and opposite phase of the unwanted noise is generated and emitted to reduce the noise at the noise control point based on the principle of superposition.

Feedforward ANC system, which is one of the ANC system, can reduce various noise such as the broad-band noise by arranging the reference microphones close to noise sources [5]. However, it is difficult to control of noises when the reference microphones cannot be arranged close to the noise sources due to various noises generate from the various noise sources or noise source moves. To solve this problem, Hase *et al.* [6] proposed an ANC system with noise source localization. In this system, some microphones were arranged around the noise control point and the nearest microphone to noise source is used as the reference microphone to reduce the noise. However, the performance of ANC system deteriorates when noise generates from different noise sources and the multiple microphones are used because each microphone catches undesired noise in addition to desired noise.

Therefore, we propose a multi-channel feedforward ANC system combined with noise source separation in this paper. This system uses microphone arrays as reference microphones to separate noises and the separated signals is used as the reference signals of the ANC system. In this paper, we examine the effectiveness of the proposed system using delay-and-sum (DS) beamformer [7] or generalized sidelobe canceller (GSC) [8] as noise source separation method through some simulations.

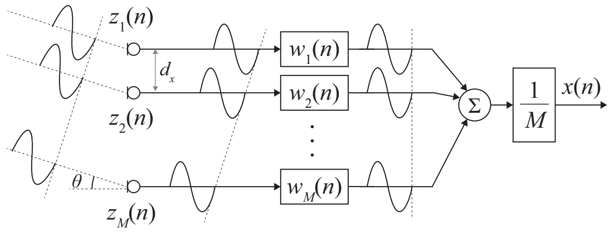

Fig. 1. The principle of DS beamformer.

II. SOUND SOURCE SEPARATION BY MICROPHONE ARRAY

Microphone array can obtain spatial information of the sound such as time difference of arrival and amplitude difference. Beamforming is the basis of the array signal processing and can be applied to sound source separation by directionality control. In this Section, we explain two beamformers: delay-and-sum (DS) beamformer and generalized sidelobe canceller (GSC).

A. DS beamformer

DS beamformer is one of the fixed beamformer composed of multiple channel fixed filters. The principle of DS beamformer is shown in Fig. 1. Assuming that the desired signal of single plane wave arrives from a certain direction. The desired signal $s(n)$ observed by the mth microphone has the following propagation delay:

$$z_m(n) = s(n - \tau_m), \quad (1)$$

$$\tau_m = \frac{d_x \sin\theta}{c}, \quad (2)$$

where $z_m(n)$ is the observed signal of the mth microphone, τ_m is the propagation delay of the mth microphone, c is the sound speed in air, and d_x is the distance between adjacent microphones, respectively. DS beamformer delays each observed signal by using filters for compensating propagation delay and adds them together. In consequence, the desired signal arriving from a certain direction is emphasized because the phases of desired signals at all microphones are matched. On the other hand, undesired signals arriving from other directions are not emphasized because the phases of desired signals at all microphones are not matched. The mth compensation filter

$w_m(n)$ is expressed by

$$w_m(n) = \frac{1}{M}\delta(n + \tau_m), \tag{3}$$

where M is the number of microphones, and $\delta(\cdot)$ is the dirac delta function, respectively.

B. Generalized Sidelobe Canceller (GSC)

GSC is one of the adaptive beamformer composed of multiple channel fixed filters and adaptive filters. The block diagram of GSC is shown in Fig. 2. In the GSC, the processing flow is as follows. First, the observed signals of each microphone $z(n)$ are compensated for propagation delay by using the fixed filters $w_c(n)$. Next, those output signals are branched to the upper and the lower sections. In the upper section, the output signals of the fixed filters are added together to emphasize the desired signal. On the one hand, in the lower section, the difference between adjacent output signals of the fixed filters is calculated. In consequence, the input signals of the adaptive filters $z_b(n)$ contain only noises rather than the desired signal. The output signal of the GSC $x(n)$ is obtained by subtracting the total output signal of the adaptive filters $y_a(n)$ from the sum of the output signals of the fixed filters $y_c(n)$. Hence, GSC has higher noise reduction performance compared with the fixed beamformer.

In this paper, we utilize the normalized least mean square (NLMS) algorithm as the updating algorithm of the adaptive filters. The updating equation of the adaptive filters is given by

$$\mathbf{w}_{a,m}(n+1) = \mathbf{w}_{a,m}(n) + \frac{\alpha'}{\beta' + ||\mathbf{z}_{b,m}(n)||^2}\mathbf{z}_{b,m}(n)x(n), \quad (m = 1, 2, \cdots, M-1), \tag{4}$$

where $\mathbf{w}_{a,m}(n)$ is the filter coefficient vector of the mth adaptive filter, $\mathbf{z}_{b,m}(n)$ is the input signal vector of the mth adaptive filter, α' is the step size parameter, and β' is the regularization parameter, respectively.

III. Feedforward ANC system with the FXNLMS algorithm

Feedforward ANC system consists of an reference microphone, an error microphone, and a secondary source. Filtered-x NLMS (FXNLMS) [9] algorithm used to update the noise control filter is the most popular update algorithm. Block diagram of the feedforward ANC system with the FXNLMS algorithm is shown in Fig. 3. In Fig. 3, the primary path P represents the noise propagation path from the noise source to the error microphone, the reference path R represents the noise propagation path from the noise source to the reference microphone, and the secondary path S represents the acoustic path from the secondary source to the error microphone including some analog devices (ADC, DAC, LPF, and so on). Feedforward ANC system generates anti-noise by convolving the reference signal $x(n)$ obtained at the reference microphone to the noise control filter $w(n)$ and emits it from the secondary

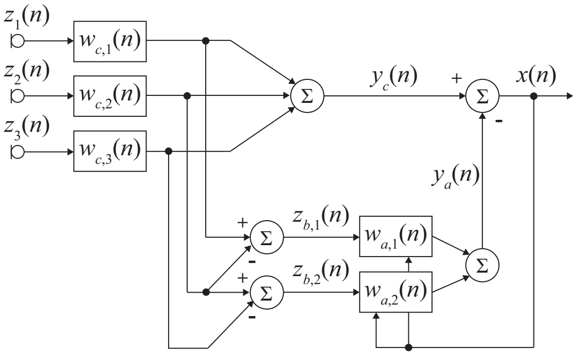

Fig. 2. Block diagram of GSC using three microphones.

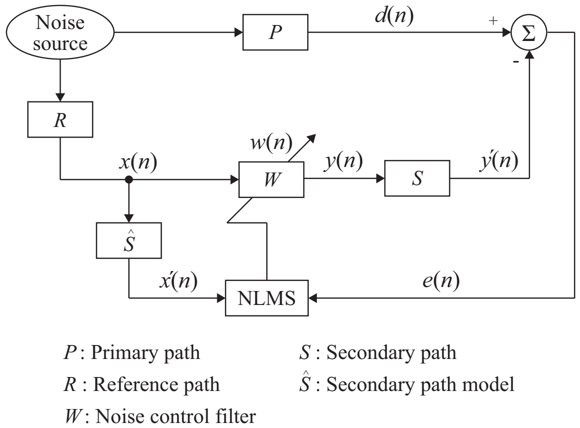

Fig. 3. Block diagram of feedforward ANC system with the FXNLMS algorithm.

source to cancel the unwanted acoustic noise at the noise control point based on the principle of superposition. Noise control filter is updated so that the mean square error for the error signal $e(n)$ is minimized. The error signal $e(n)$ is obtained at the error microphone and given by

$$e(n) = d(n) - y'(n), \tag{5}$$

$$= d(n) - \mathbf{s}^{\mathrm{T}}\mathbf{y}(n), \tag{6}$$

where $d(n)$ is the noise signal propagated through the primary path, $\mathbf{s}$ is the secondary path, $\mathbf{y}(n)$ is the output signal vector of noise control filter, and $y'(n)$ is the output signal of noise control filter propagated through the secondary path, respectively. From Eq. (6), the output signal (the anti-noise) is changed due to the characteristics of the secondary path. In the FXNLMS algorithm, the noise control filter is updated using the secondary path model in consideration for the characteristics of the secondary path. The updating equation of the noise control filter is given by

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\alpha}{\beta + ||\mathbf{x}'(n)||^2}\mathbf{x}'(n)e(n), \tag{7}$$

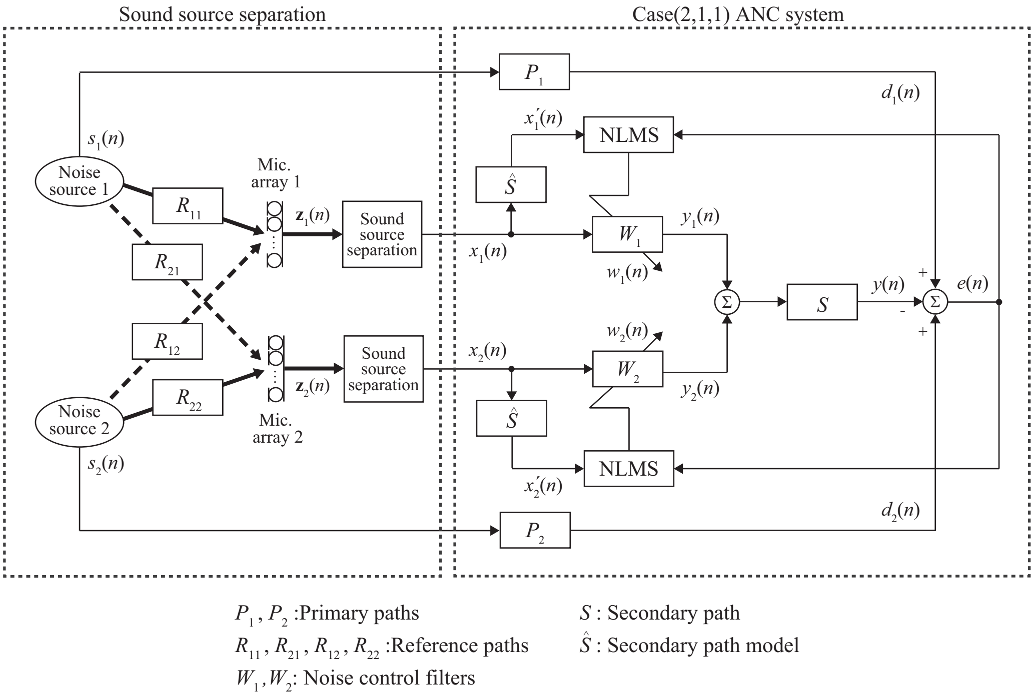

Fig. 4. Block diagram of a Case(2,1,1) ANC system using microphone arrays for noise source separation.

where $\mathbf{w}(n)$ is the filter coefficient vector of the noise control filter, $\mathbf{x}'(n)$ is the filtered reference signal vector, $e(n)$ is the error signal, α is the step size parameter, and β is the regularization parameter, respectively.

IV. Multi-channel feedforward ANC system combined with noise source separation

Multi-channel feedforward ANC system consists of several reference microphones, error microphones, and perhaps even several secondary sources. According to the common naming rule, a multi-channel ANC system consisting of two reference microphones, one error microphone, and one secondary source is called a Case(2,1,1) ANC system.

Block diagram of a Case(2,1,1) ANC system combined with noise source separation is shown in Fig. 4. The proposed system uses microphone arrays as reference microphones. The observed signals of the two reference microphone arrays are separated into two noise sources using a sound source separation method. After that, the separated signals are used as the reference signals of the Case(2,1,1) ANC system. By using the proposed system, it is expected that the noise reduction performance can be improved when the reference microphones are arranged away from the noise sources.

In this paper, we utilize the FXNLMS algorithm as the updating algorithm of the noise control filters. The updating equation of the noise control filters is given by

$$\mathbf{w}_j(n+1) = \mathbf{w}_j(n) + \frac{\alpha}{\beta + ||\mathbf{x}'_j(n)||^2}\mathbf{x}'_j(n)e(n), \quad (j = 1, 2), \tag{8}$$

where j is the index number of the reference microphone array, $\mathbf{w}_j(n)$ is the filter coefficient vector of the jth noise control filter, $\mathbf{x}'_j(n)$ is the jth filtered reference signal vector, $e(n)$ is the error signal, α is the step size parameter, and β is the regularization parameter, respectively.

V. Simulation results

In this section, we demonstrate the effectiveness of the proposed Case(2,1,1) ANC system using the DS beamformer or the GSC as noise source separation method. We compare the noise reduction performance and the noise separation performance of the proposed Case(2,1,1) ANC system. Moreover, we compare the noise reduction performance of the Case(2,1,1) ANC system in case where the number of microphones in the microphone array is changed. Figure 5 shows the arrangement of noise sources, reference microphone arrays, an error microphone, and a secondary source in the simulation. In Fig. 5, the mic. array 1 suppresses the noise signal of the noise source 2 and the mic. array 2 suppresses the noise signal of the noise source 1. Table I shows the basic simulation conditions. The fixed filters of the DS beamformer and the GSC were created by using the acoustic path models

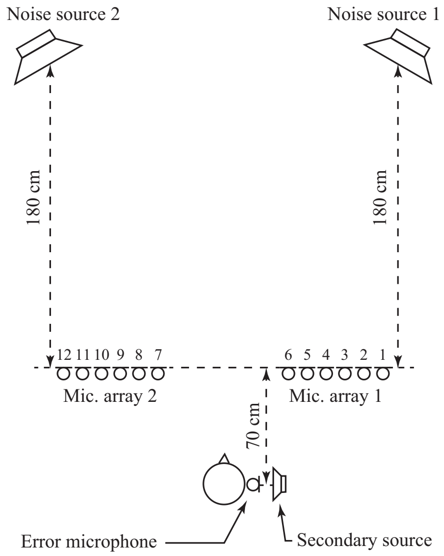

Fig. 5. Arrangement of noise sources, reference microphone arrays, an error microphone, and a secondary source in the simulation.

from each noise source to each microphone. Noise reduction performance is evaluated by the "Reduction" defined by

$$\text{Reduction} = 10\log_{10}\frac{\sum\left(d_1\left(n\right)+d_2\left(n\right)\right)^2}{\sum e^2\left(n\right)}, \tag{9}$$

where $d_1(n)$ and $d_2(n)$ are the noise signals at the error microphone, and $e(n)$ is the error signal, respectively. Noise separation performance is evaluated by the "Magnitude-squared coherence (MSC)" defined by

$$\text{MSC} = \frac{|P_{xy}(f)|^2}{P_{xx}(f)P_{yy}(f)}, \tag{10}$$

where $P_{xy}(f)$ is the cross power spectral density between two signals, and $P_{xx}(f)$ and $P_{yy}(f)$ are the power spectral densities of each signal, respectively. The MSC indicates the degree of coincidence of the two signals at each frequency. Therefore, it is possible to confirm the separation performance at each frequency by comparing the noise source and the separated signal using MSC.

A. Effectiveness of the proposed Case(2,1,1) ANC system

In this section, we show the effectiveness of the proposed Case(2,1,1) ANC system. We compare three ANC systems, without the noise source separation (conventional method) and with the noise source separation using the DS beamformer or the GSC in this simulation. Table II shows the step size parameters in each ANC system. Each step size parameter was set to optimal values based on some preliminary simulation results.

Figure 6 shows the comparison of the noise reduction performance in each ANC system. From Fig. 6, we can confirm that the noise reduction performances of the ANC systems with the noise source separation are higher than that of the

TABLE I
BASIC SIMULATION CONDITIONS.

Noise source 1	White noise
Noise source 2	White noise
Sampling frequency f_s	12000 Hz
Cut-off frequency f_c	2500 Hz
Tap length of primary path P	1000
Tap length of reference path R	1000
Tap length of secondary path S	200
Tap length of secondary path model $\hat{S}$	200
Tap length of noise control filter W	1500
Tap length of DS beamformer filter W_c	230
Tap length of adaptive filter of GSC W_a	500
Regularization parameter β	1.0×10^{-6}

TABLE II
STEP SIZE PARAMETERS IN EACH ANC SYSTEM.

	Adaptive filter of GSC	Noise control filter
Conventional method	-	0.05
DS beamformer	-	0.01
GSC	0.01	0.05

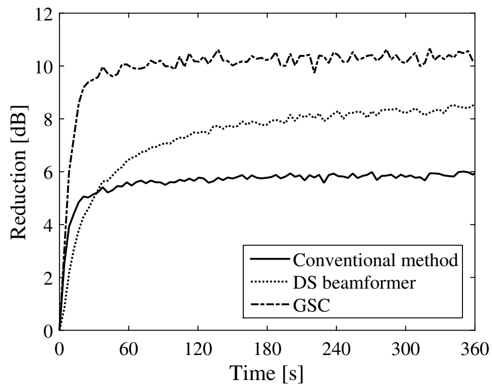

Fig. 6. Comparison of the noise reduction performance in each ANC system.

ANC system without the noise source separation. Moreover, the GSC can yield higher noise reduction performance and faster convergence speed than the DS beamformer.

Figure 7 shows the comparison of the noise separation performance in each method. Figure 7 (a) shows the MSC between the noise signal of the noise source 2 and the observed signal in the mic. 10, Figure 7 (b) shows the MSC between the noise signal of the noise source 2 and the output signal of the DS beamformer in the mic. array 2, and Figure 7 (c) shows the MSC between the noise signal of the noise source 2 and the output signal of the GSC in the mic. array 2. From Fig. 7, we can confirm that the noise separation performance of the GSC is overall higher than that of the DS beamformer. In particular, the GSC has considerably higher noise separation performance than the DS beamformer in the frequency band from 500 to 750 Hz.

From these results, the proposed Case(2,1,1) ANC system

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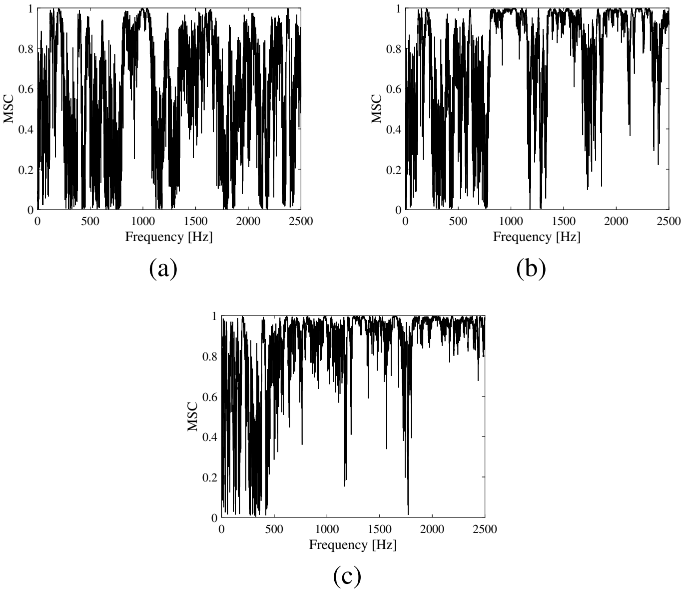

Fig. 7. Comparison of the noise separation performance between the noise signal of the noise source 2 and each signal. (a) Observed signal of the mic. 10, (b) output signal of the DS beamformer in the mic. array 2, and (c) output signal of the GSC in the mic. array 2.

TABLE III
STEP SIZE PARAMETERS FOR VARIOUS COMBINATION NUMBERS OF MICROPHONES.

	Adaptive filter of GSC	Noise control filter
DS beamformer	-	0.01
GSC ($M = 2$)	0.04	0.03
GSC ($M = 3$)	0.03	0.05
GSC ($M = 4$)	0.03	0.05
GSC ($M = 5$)	0.03	0.05
GSC ($M = 6$)	0.01	0.05

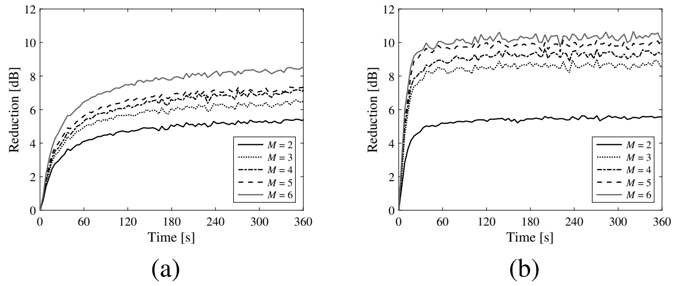

Fig. 8. Comparison of the noise reduction performance of the proposed Case(2,1,1) ANC system in case where the number of microphones in the microphone array is changed. (a) Using the DS beamformer and (b) using the GSC.

can realize superior noise reduction performance through the use of noise source separation. Additionally, the case of using the GSC has higher noise reduction performance and faster convergence speed than the case of using the DS beamformer because the GSC has higher noise separation performance than the DS beamformer.

B. Noise reduction performance in case where the number of microphones in the microphone array is changed

In this section, we examine the noise reduction performance of the proposed Case(2,1,1) ANC system in case where the number of microphones in the microphone array is changed. The number of microphones was changed from two to six. Table III shows the step size parameters for each combination number of microphones. Each step size parameter was set to optimal values based on some preliminary simulation results.

Figure 8 shows the comparison of the noise reduction performance of the proposed Case(2,1,1) ANC system using the various numbers of microphones. Figure 8 (a) shows that of the DS beamformer and Figure 8 (b) shows that of the GSC. From Fig. 8, we can confirm that the noise reduction performance of the proposed Case(2,1,1) ANC system can be improved by using more than two microphones for forming the microphone array. Moreover, we can confirm that the case of using the GSC has higher noise reduction performance with smaller number of microphones than the case of using the DS beamformer.

VI. CONCLUSIONS

In this paper, we have proposed the Case(2,1,1) ANC system combined with noise source separation and confirmed the effectiveness of the proposed ANC system through some simulations. As a result, this system can improve the noise reduction performance in case where the ANC system uses two reference microphones and the reference microphones cannot be arranged close to the noise sources.

In the future, we will examine the noise reduction performance of the ANC system combined with noise source separation based on the noise source localization.

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