Audio Integrated Active Noise Control System with Auto Gain Controller

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***Abstract*—This paper proposes an audio integrated active noise control (ANC) system with an auto gain controller. An ANC system is one of the techniques for reducing unwanted noise and used to reduce the factory noise, engine noise, and so forth. In general, the ANC system cannot completely reduce the unwanted noise due to its principle. To solve this problem, an audio integrated ANC (AIANC) system has been proposed. The AIANC system uses the additional audio signal to mask the residual noise called error signal. Also, the AIANC system can be used for telecommunication under noisy environment, in which the voice is treated as the audio signal of the AIANC system. However, in the conventional AIANC system, the power of the audio signal cannot be adjusted to that of the error signal and it causes that the audio signal is too larger or smaller than the error signal. To solve this problem, the AIANC with an auto gain controller is proposed. The proposed AIANC system has the auto gain controller to adjust the power of the audio signal and that of the error signal. Then, the audio signal is emitted with the same power as the error signal. Simulation results shows that the proposed AIANC system can reduce the unwanted noise and adjust the power of the audio signal to that of the error signal.**

I. Introduction

An active noise control (ANC) system is one of the techniques for reducing unwanted noise [1]–[6]. The ANC system is based on the superposition between the unwanted noise and anti-noise which has same amplitude and opposite phase of the unwanted noise. The structures of ANC systems are classified into a feedforward control [1]–[3], [6] and feedback control [7]–[9]. In this paper, the feedforward ANC sytem is focused on.

The feedforward ANC system consists of a reference microphone, error microphone, and secondary loudspeaker and can reduce the various noises of industrial equipments, engines, air conditioners, and so forth. The noise reduction ability of the feedforward ANC system is determined by the coherence between the reference signal obtained at the reference microphone and the unwanted noise obtained at the error microphone [3]. In general, the reference signal and the error signal differ from each other and the ANC system cannot completely reduce the unwanted noise.

An audio integrated ANC (AIANC) system [10], [11] is one of the solutions for this problem. The AIANC system uses the additional audio signal to mask the residual noise called error signal. Also, the AIANC system can be used for telecommunication under noisy environment, in which the voice is treated as the audio signal of the AIANC system. However, in the conventional AIANC system, the power of the audio signal cannot be adjusted to that of the error signal and it causes that the audio signal is too larger or smaller than the error signal.

To solve this problem, the AIANC with an auto gain controller is proposed. The proposed AIANC system has the auto gain controller to adjust the power of the audio signal and that of the error signal. Then, the audio signal can propagate with the same level as the error signal. Computer simulation was conducted to show that the proposed AIANC system can reduce the unwanted noise and adjust the power of the audio signal to that of the error signal.

II. Conventional AIANC system

An AIANC system [10], [11] is one of the ANC systems with additional audio signal. The AIANC systems in [10], [11] can identify the secondary path model and reduce the unwanted noise at the same time. In this paper, the online secondary path modeling is omitted to simplify the system.

Block diagram of the AIANC system with the feedforward ANC structure is shown in Fig. 1. In Fig. 1, P, R, and S represent the primary path, reference path, and secondary path, $\hat{S}$ is the secondary path model of S, W is the noise control filter, respectively. The AIANC system aims to reduce the unwanted noise $d(n)$ and mask the error signal $e(n)$ by the additional audio signal $a(n)$ at the same time.

In the AIANC system, the reference signal $x(n)$ is obtained by the reference microphone. Then, the control signal $y'(n)$ is generated by the noise control filter as

$$y'(n) = \mathbf{w}\ \ (n)\mathbf{x}(n), \tag{1}$$

where $\mathbf{w}(n)$ is the filter coefficient vector of the noise control filter. The audio signal $a(n)$ is mixed to $y'(n)$ and mixed control signal is emitted from the secondary loudspeaker. This mixed control signal and that obtained at the error microphone are as

$$y'_{\mathrm{M}}(n) = y'(n) + g_{\mathrm{C}}a(n), \tag{2}$$

$$y(n) = \mathbf{s}\ \ (n)\mathbf{y}'(n), \tag{3}$$

where g_{C} is the constant gain. The error signal $e_{\mathrm{A}}(n)$ is obtained at the error microphone and can be represented as

$$e_{\mathrm{A}}(n) = d(n) - y_{\mathrm{M}}(n). \tag{4}$$

In the basic feedforward ANC system, the noise control filter is updated by the error signal. However, in the AIANC system, the error signal $e_{\mathrm{A}}(n)$ includes the audio signal through the secondary path $a_{\mathrm{s}}(n)$ and $a_{\mathrm{s}}(n)$ disturbs the update of the noise control filter. Therefore, the error signal $e(n)$

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

is estimated and used for the update of the noise control filter. The estimated error signal is obtained by

$$\hat{e}(n) = e_{\mathrm{A}}(n) - \hat{a}_{\mathrm{s}}(n), \tag{6}$$

where $\hat{a}_{\mathrm{s}}(n)$ is the filtered audio signal calculated by

$$\hat{a}_{\mathrm{s}}(n) = \hat{\mathbf{s}}\ \ (n)\mathbf{a}(n), \tag{7}$$

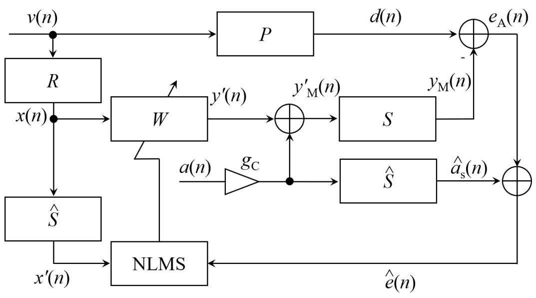


Fig. 1. Block diagram of conventional audio integrated ANC system.

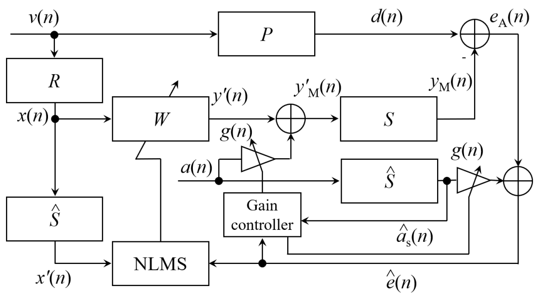


Fig. 2. Block diagram of audio integrated ANC system with gain controller.

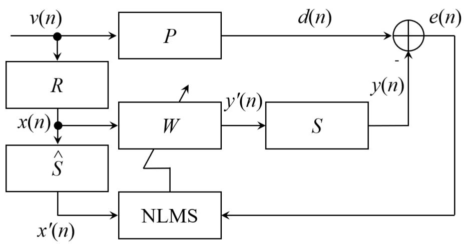


Fig. 3. Block diagram of basic feedforward ANC system.

and $\hat{\mathbf{s}}(n)$ is the impulse response vector of the secondary path model. Using the estimated error signal $\hat{e}(n)$, the noise control filter is updated by the filtered-x NLMS algorithm [12], [13] as

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

where $\mathbf{x}'(n)$ is the filtered reference signal vector calculated by

$$x'(n) = \hat{\mathbf{s}}\ \ (n)\mathbf{x}(n), \tag{9}$$

$\|\cdot\|$ is the l_2 norm, α is the step size parameter, and β is the regularization parameter, respectively.

However, the conventional AIANC system cannot adjust the power of the audio signal to the error signal. In other words, the noise reduction performance of the ANC system cannot be generally known in advance and the power of the error signal cannot be estimated. This is because the noise reduction performance of the ANC system depends on the response of each path which is time variant system. Hence, it is difficult to adjust the audio signal to the error signal in advance and the audio signal may be too larger or smaller than the error signal.

III. Proposed AIANC system with auto gain controller

The proposed AIANC has an auto gain controller to adjust the power of the audio signal and that of the error signal. Figure 2 shows the proposed AIANC system, where $g(n)$ is the time variant gain for the audio signal. The proposed AIANC system adjusts the power of the audio signal to the error signal by the time variant gain. Here, the mixed control signal $y'(n)$ is rewritten as

$$y'_{\mathrm{M}}(n) = y'(n) + g(n)a(n), \tag{10}$$

where $g(n)$ is the time variant gain. Here, the error signal $e_{\mathrm{A}}(n)$ is rewritten as

$$\begin{aligned} e_{\mathrm{A}}(n) &= d(n) - y_{\mathrm{M}}(n) \\ &= d(n) - \mathbf{s}\ \ (n)\mathbf{y}'_{\mathrm{M}}(n) \\ &= d(n) - \mathbf{s}\ \ (n)\left\{\mathbf{y}'(n) + \mathbf{a}_{\mathrm{G}}(n)\right\} \\ &= e(n) - \mathbf{s}\ \ (n)\mathbf{a}_{\mathrm{G}}(n), \end{aligned} \tag{11}$$

where $\mathbf{a}_{\mathrm{G}}(n) = [g(n)a(n)\ \ g(n-1)a(n-1)\ \ \cdots\ \ g(n-i)a(n-i)\ \ \cdots]$ is the audio signal vector with the gain. One of the goals of the AIANC system is to adjust the power of the audio signal to that of the error signal. In this situation, the relationship between the audio signal and error signal is written as

$$\sum_{l=0}^{L-1} \hat{e}^2(n-l) = \sum_{l=0}^{L-1} a_{\mathrm{s}}^2(n-l), \tag{12}$$

where $a_{\mathrm{s}}(n)$ is the audio signal emitted from the secondary loudspeaker represented by

$$a_{\mathrm{s}}(n) = \mathbf{s}\ \ (n)\mathbf{a}(n). \tag{13}$$

Also, when the error signal should be masked, the audio signal should be larger than the error signal. In this situation, the relationship between these signals is written as

$$\sum_{l=0}^{L-1} \hat{e}^2(n-l) < \sum_{l=0}^{L-1} \hat{a}_{\mathrm{s}}^2(n-l), \tag{14}$$

or

$$\sum_{l=0}^{L-1} \hat{e}^2(n-l) < c\sum_{l=0}^{L-1} \hat{a}_{\mathrm{s}}^2(n-l), \tag{15}$$

where $c\ (>1)$ is the constant.

In this paper, the situation of (12) is considered and the gain $g(n)$ is calculated by comparison of the average power of the audio signal $a_{\mathrm{s}}(n)$ and that of the estimated error signal $\hat{e}(n)$. The update equation of the gain $g(n)$ is shown as

$$g(n) = \sqrt{\frac{\sum_{l=0}^{L-1} |\hat{e}(n-l)|^2}{\sum_{l=0}^{L-1} |\hat{a}_{\mathrm{s}}(n-l)|^2}}, \tag{16}$$

where L is the update interval of the gain. The gain $g(n)$ is updated every L samples.

TABLE I
SIMULATION CONDITIONS.

Noise source	White noise
Additional audio signal	01 Brook (Lch is only used)
Sampling frequency	48000 Hz
Frequency range	70 – 12000 Hz
Tap length of primary path P	8192
Tap length of reference path R	4096
Tap length of secondary path S	512
Tap length of secondary path model $\hat{S}$	512
Tap length of noise control filter W	1024
Step size parameter α	0.01
Regularization parameter β	1.0×10^{-6}
Fixed gain for conventional AIANC g_C	2.0

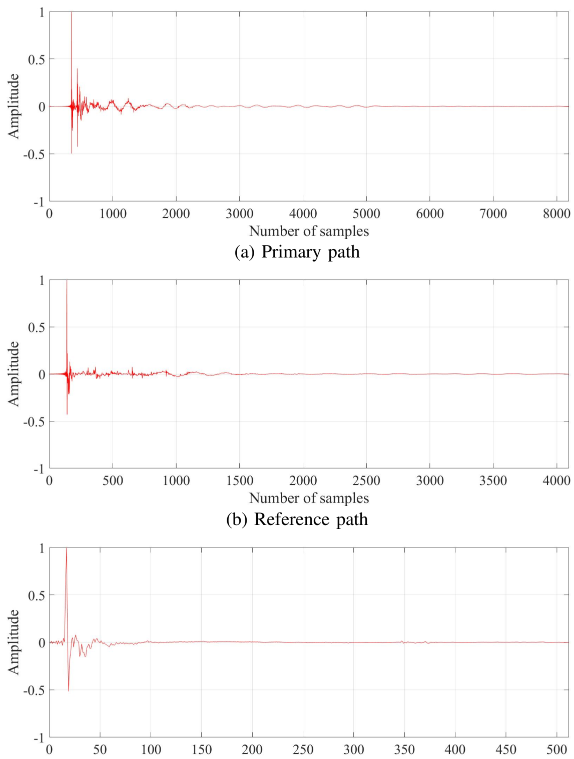

(a) Primary path
(b) Reference path
(c) Secondary path

Fig. 4. Impulse responses of each path.

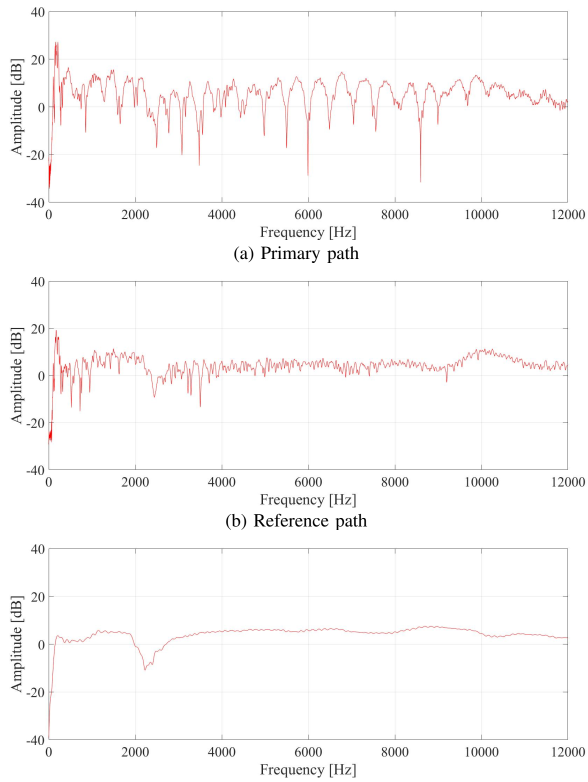

(a) Primary path
(b) Reference path
(c) Secondary path

Fig. 5. Frequency responses of each path.

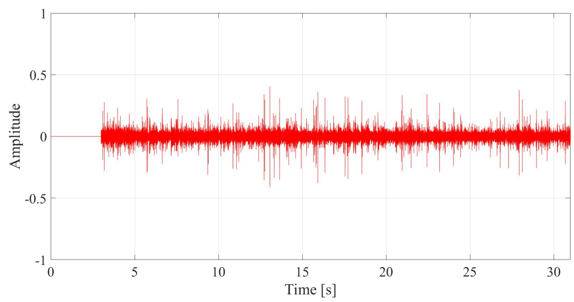


Fig. 6. Time waveform of audio signal $a(n)$.

IV. SIMULATION RESULTS

Computer simulation was conducted to evaluate the noise uction performance and power adaptation ability of the proposed AIANC system. In the simulation, the proposed AIANC system shown in Fig. 2, conventional AIANC system shown in Fig. 1, and basic feedforward ANC system shown in Fig. 3 were used. Impulse responses of each path were identified in advance by adaptive digital filter with NLMS algorithm. Impulse responses and frequency responses of each path are shown in Figs. 4 and 5, respectively. Simulation conditions are shown in Table I. In this simulation, the sound of water stream was chosen as the additional audio signal from BBC sound effect library [14]. Here, the audio signal in the library was recorded with sampling frequency of 44100 Hz as stereo signal. Then, the audio signal was resampled with 48000 Hz and left channel of the signal was only used. ANC systems were activated at 3 seconds. The time waveform and spectra of the audio signal are shown in Figs. 6 and 7.

Time waveforms and the power transitions of the error signals are shown in Figs. 8 and 9, respectively. From Fig. 8, each ANC system can reduce the unwanted noise about 9 dB. In the conventional AIANC system, the error signal was similar to that of the basic ANC system. On the other hand, in the proposed AIANC system, the error signal was larger than that of the basic ANC system. From Fig. 9, the error signal had the power about -25 dB by the basic ANC system. The audio signal with the constant gain had the power about -20 dB and was little bigger than that of the error signal. On the other hand, the audio signal with the time variant gain had the power about -25 dB. Hence, it can be said that the proposed AIANC system adjusted the power of the audio signal to that

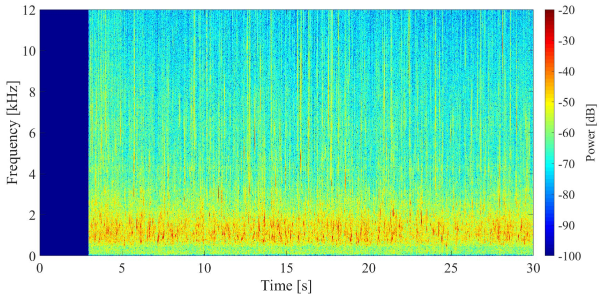
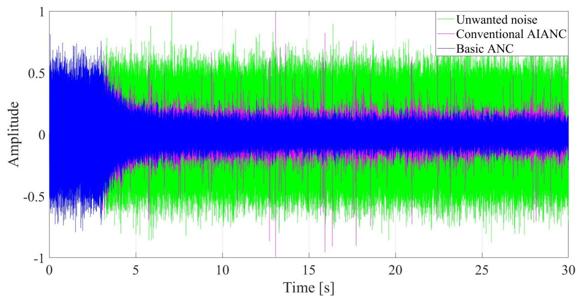
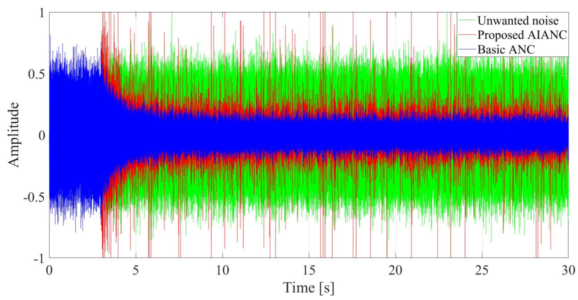


Fig. 7. Spectrogram of audio signal $a(n)$.

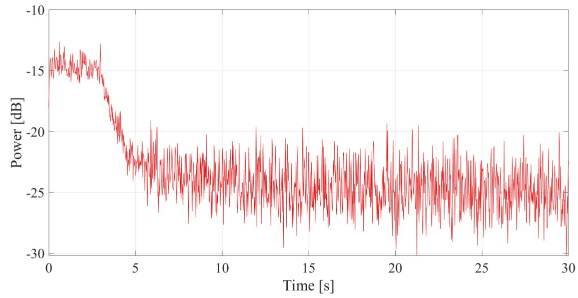


(a) Conventional AIANC system

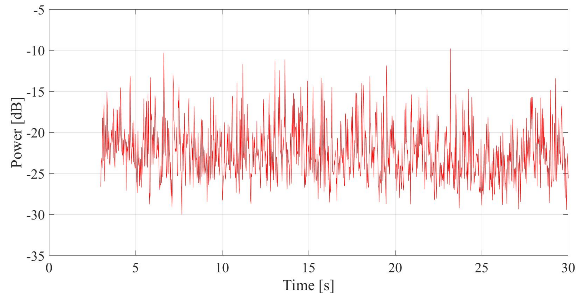


(b) Proposed AIANC system

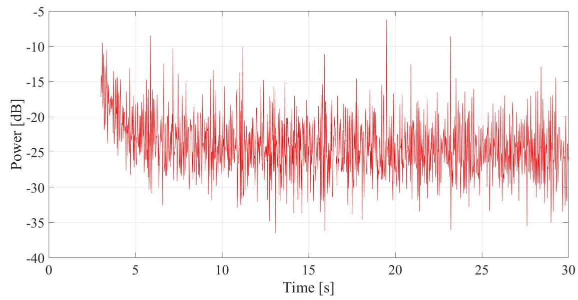
Fig. 8. Time waveforms of error signals in each ANC system.



(a) Error signal



(b) Audio signal with constant gain



(c) Audio signal with time-variant gain

Fig. 9. Power transitions of each signal.

of the error signal.

Figure 10 shows the spectrograms of the error signal with each ANC system. Compared with Fig. 10 (a) and (b), the spectrograms of the error signals obtained by both systems are almost similar and the audio signal was not added with sufficient power by the conventional AIANC system. On the other hand, compared with Fig. 10 (a) and (c), the spectrogram of the error signal obtained by the proposed AIANC system is different from that obtained by the basic ANC system. This is because the audio signal was larger than the error signal by the time variant gain. However, the power may be too large below 2000 Hz. This is because the time variant gain was calculated by the average powers of the audio signal and error signal. To prevent the power bias, the gain should be calculated in each subbands.

From these results, the proposed AIANC system can effectively reduce the unwanted noise and mask the error signal at the same time. However, the masking ability of the proposed AIANC system is insufficient and the improvement of the proposed AIANC system is required.

V. Conclusion

In this paper, the AIANC system with the auto gain controller was proposed. The conventional AIANC system cannot adjust the power of the audio signal to that of the error signal and it causes that the audio signal is too louder or smaller than the error signal. To solve this problem, the AIANC with the auto gain controller is proposed. The proposed AIANC system has the auto gain controller to adjust the power of the audio signal and that of the error signal. Then, the power of the audio signal can be adjusted to that of the error signal. From the simulation results, the proposed AIANC system achieves both reducing the unwanted noise and adjust the audio signal to the error signal. Future works of this system is to adopt the subband ANC system [15]–[19] to adjust the power of the audio signal to that of the error signal in each subband.

Acknowledgment

This work was supported by KDDI Foundation Research Grant Program 2019.

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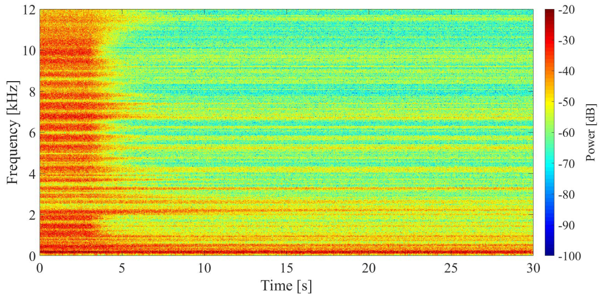
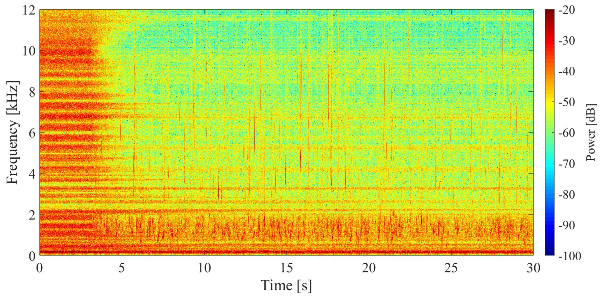
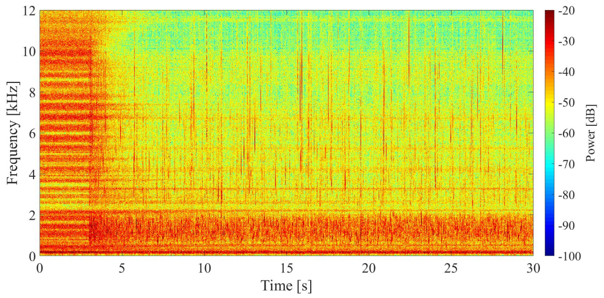


(a) Basic ANC system



(b) Conventional AIANC system



(c) Proposed AIANC system

Fig. 10. Spectrograms of error signals in each ANC system.

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