Head-mounted active noise control system to achieve speech communication

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***Abstract*—In this paper, we propose a head-mounted active noise control (ANC) system with speech communication. Magnetic resonance imaging (MRI) device is one of medical equipment. Recently, MRI device is utilized for microwave coagulation therapy. However, MRI device generates a serious noise (over 100 dB SPL). Hence, surgeons and other medical staff are exposed to the large MR noise for many hours and cannot verbally communicate with each other. We have therefore proposed a head-mounted ANC system for reducing MR noise and realizing verbal communication under such a loud noise environment. However, MRI device is generally controlled by the operator outside the MRI room. Hence, speech communication between inside and outside the room is needed. We therefore integrate the speech communication function with the head-mounted ANC system. Concretely, the error microphones and secondary loudspeakers are also used as an interface to realize the speech communication. In this case, the outside voice may be returned through the error microphone, so an audio-integrated ANC system based on the echo cancellation is utilized. Linear prediction filter is also utilized for separating the inside voice from residual noise. In this paper, we demonstrate the validity of the proposed ANC system through some noise reduction experiments and subjective assessment tests on phoneme articulation.**

I. Introduction

Acoustic noise problems are becoming increasingly serious with the increasing use of industrial equipment. Active noise control (ANC) [1], [2] has been studied as a means of solving such acoustic noise problems. ANC is a technique based on the principle of superposition, i. e., an anti-noise with the same amplitude and opposite phase is generated and combined with an unwanted noise, thus resulting in the cancellation of both noises. The control structure of ANC is classified into two groups. One is a feedforward structure and the other is a feedback structure. Feedforward ANC systems are very commonly used and can reduce all classes of noise, but the system scale is likely to be large. On the other hand, feedback ANC systems [3], [4] have a small scale in comparison with feedforward ANC systems. Feedback ANC systems are effective for narrowband noise and are widely used in headset applications [5], [6] because of their small scale. An application example of this system is to reduce MR noise generated from MRI device.

Magnetic resonance imaging (MRI) devices, which are used to take images inside the patient's body, have been introduced in many medical institutions on the grounds of safety and convenience. In particular, an open-configuration MR system [7] was introduced to conduct the microwave coagulation therapy assisted by near-realtime MR imaging.

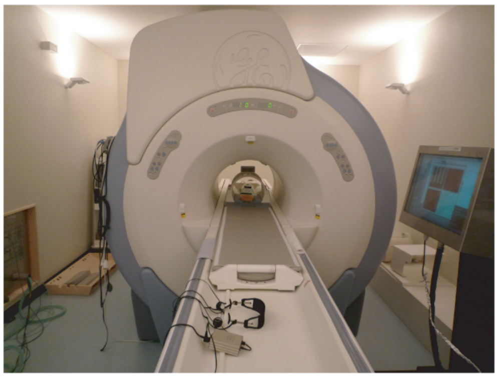


Fig. 1. Open-configuration MR system.

However, taking images with an MRI device leads to intense noise (referred to as MR noise in this paper) because the gradient coil in the MRI device vibrates owing to the Lorentz force. Exposure to the intense noise may cause operators and other medical staff to suffer extreme stress and prevent verbal communication between the staff members. This may lead to accidents [8].

We have already proposed a head-mounted active noise control (ANC) system [9], [10] for reducing MR noise to solve this problem [11]. The proposed system was based on the internal model control (IMC) principle [12], [13]. The IMC-based feedback ANC system can reduce noise which has periodicity. In the head-mounted ANC system, the secondary sources and error microphones are placed near the opening of the ear canals. However, MRI device is generally controlled by the operator outside the MRI room. Hence, speech communication between inside and outside the room is needed. We therefore integrate the speech communication function with the head-mounted ANC system. Concretely, the error microphones and secondary loudspeakers are also used as an interface to realize the speech communication. In this case, the outside voice may be returned through the error microphone, so an audio-integrated ANC system based on the echo cancellation is utilized. However, the residual noise is included in the inside voice sent to the outside room. Hence, linear prediction filter is also utilized for separating the inside voice from residual noise.

In this paper, we demonstrate the validity of the proposed ANC system through some noise reduction experiments and subjective assessment tests on phoneme articulation.

II. HEAD-MOUNTED ANC SYSTEM

A. MR Noise

MRI devices are used to take anatomical images of inside the human body without exposing subjects to harmful radiation and have been introduced in many medical institutions for the diagnosis of diseases and for biomedical research because of their safety and convenience. Recently, an open-configuration MR system has been introduced to conduct microwave coagulation therapy with the help of near-realtime MR imaging [7]. Figure 1 shows an open-configuration MR system.

A typical time waveform of MR noise measured with an optical microphone in an MRI room is shown in Figure 2 along with the corresponding spectrum. Note that the envelop of the time waveform of the MR noise varies with the time, i.e., the MR noise is nonstationary. To make things even more difficult, the measured signal has periodically occurring discontinuities. On the other hand, the MR noise has a very regular spectral structure, that is, it consists of many harmonically related periodic components with frequencies that are multiplies of the same fundamental frequency. Therefore, we use an adaptive feedback ANC system for reducing MR noise.

B. Adaptive Feedback ANC System

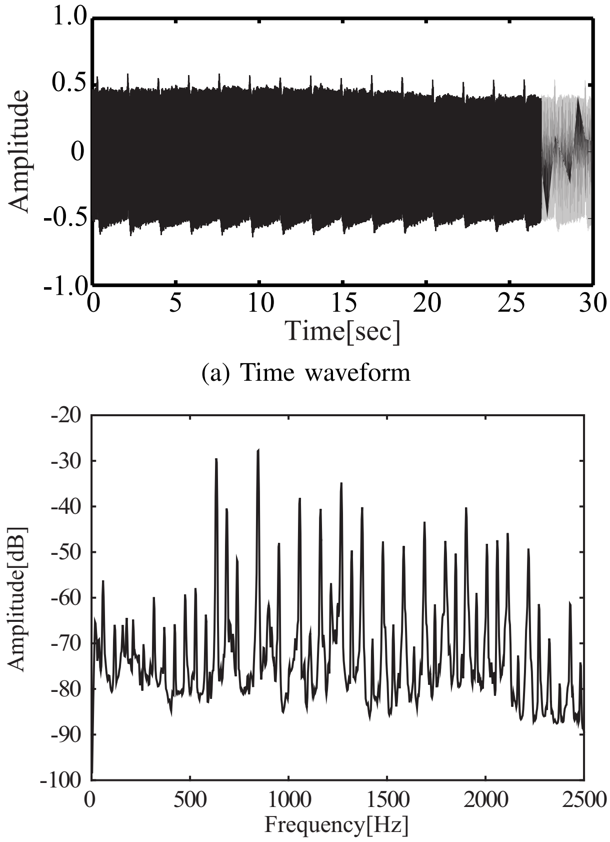
Adaptive feedback active noise control system consists of an error microphone and a secondary source (loudspeaker). For the updating algorithm of noise control filter in the adaptive feedback ANC system [14]–[16], Filtered-X NLMS (FXNLMS) [17] algorithm is the most popular. The block diagram of the adaptive feedback ANC system with the FXNLMS algorithm is illustrated in Figure 3. In this system, the linear prediction of the primary noise is used instead of using reference microphones. Hence, the system can control noise with periodicity or having narrow band components regardless of the direction of arrival. In this algorithm, an unknown system between secondary source and error microphone shown as S in the figure, is modeled by the secondary path model $\hat{S}$, before the ANC system starts operating. The updating algorithm of the noise control filter is expressed as follows.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\alpha}{\|\mathbf{r}(n)\|^2 + \beta}\mathbf{r}(n)e(n), \qquad (1)$$

where $\mathbf{w}(n) = [w_1(n)\ \ w_2(n)\cdots w_i(n)\cdots w_N(n)]^{\mathrm{T}}$ and $\mathbf{r}(n) = [r(n)\ r(n-1)\cdots r(n-N+1)]^{\mathrm{T}}$ are a tap-weight vector of the noise control filter and filtered reference signal vector, respectively. Moreover, $e(n)$ is the error signal picked up at the error microphone. Futhermore, α and β are a step size and regularization parameters, respectively.

C. Head-Mounted Structure

Figure 4 shows a head-mounted ANC system we developed [11]. In this system, compact loudspeakers are located near the user's ears and microphones are also located at the opening of the external auditory canal. Hence, since the user's ears are not covered, clear verbal communication can be realized under loud noise environments. In addition, since the user's head is located between each microphone and loudspeaker pair, the crosstalk does not arise. Hence, the left and right channels can be independently controlled with a single-channel feedback ANC system.



(b) Spectrum

Fig. 2. An example of time waveform and spectrum of MR noise.

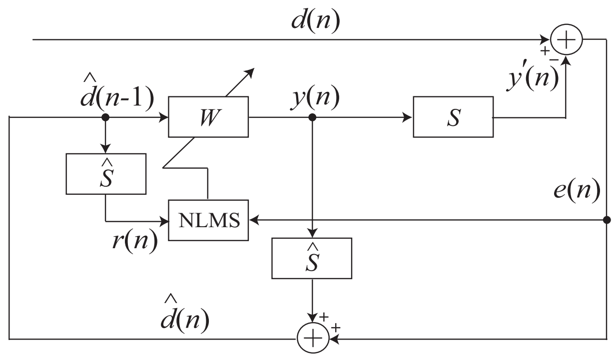


Fig. 3. Block diagram of adaptive feedback ANC system with the FXNLMS algorithm.

III. HEAD-MOUNTED ANC SYSTEM WITH SPEECH COMMUNICATION

A. Audio-Integrated ANC System

The ANC systems using the FXNLMS algorithm is effective in reducing low-frequency noise. In the ANC system with speech communication, the secondary loudspeaker is also used as an emitter of speech signal while reducing noise. In this case, the noise reduction performance may be degraded due to the speech signal. To prevent the ANC system from canceling the desired speech signal and to avoid the speech signal acting as interference to degrade the ANC performance, the audio-integrated ANC algorithm was proposed [18]–[26].

The block diagram of the audio-integrated ANC system is illustrated in Figure 5. In Figure 5, the audio signal $a(n)$ is added to the adaptive filter output $y(n)$, and the mixed signal $y'(n)$ is output to the secondary loudspeaker for canceling the primary noise $d(n)$. Thus, the signal $e(n)$ picked up by the error microphone contains both the residual noise and the desired speech component. To estimate the audio component picked up by the error microphone, $a(n)$ is filtered by the secondary path estimation filter $\hat{S}(z)$. The estimated audio component $a'(n)$ is subtracted from $e(n)$ to get the true error signal $e'(n)$ if $\hat{S}(z) = S(z)$. This audio-free error signal is then used to update the adaptive filter $W(z)$. Therefore, the performance of the FXNLMS algorithm will not be degraded by the additional audio signal, and the ANC system will not cancel the desired audio component because the audio component is not fed back to the NLMS algorithm.

B. Linear Prediction Filter

Figure 6 shows a block diagram of the linear prediction filter [27], [28] with the NLMS algorithm. This filter produces no output in the case of an unpredictable broadband signal and passes only a predictable narrowband signal. The update algorithm of the linear prediction filter at sample time n is expressed as

$$\mathbf{h}(n+1) = \mathbf{h}(n) + \frac{\alpha_h}{\|\mathbf{x}(n-\Delta)\|^2 + \beta_h}\mathbf{x}(n-\Delta)f(n), \quad (2)$$

where α_h is the step size parameter, β_h is the regularization parameter, $\mathbf{h}(n) = [h_1(n)\ h_2(n)\cdots h_i(n)\cdots h_K(n)]^{\mathrm{T}}$ is the coefficient vector of the linear prediction filter, $\mathbf{x}(n) = [x(n)\ x(n-1)\cdots x(n-i+1)\cdots x(n-K+1)]^{\mathrm{T}}$ is the input signal vector, $f(n)$ is the prediction error, and K is the tap length of the linear prediction filter, respectively. Δ is the delay of the input signal to the linear prediction filter and is determined by the autocorrelation characteristic of the removed signal. That is, Δ is set to a small value for white noise and pink noise and to a large value for speech signals because speech signals have a stronger autocorrelation characteristic than white and pink noises.

C. Problem of Feedback ANC System to Achieve Speech Communication

The block diagram of the adaptive feedback ANC system with the FXNLMS algorithm to achieve speech communication between the inside and outside MRI room is illustrated in Figure 7. In this figure, $v(n)$, $a(n)$ and $f(n)$ are the inside speech signal, the outside speech signal, and the output of the loudspeaker outside the MRI room, respectively. Moreover, $d(n)$ and $e(n)$ are the unwanted MR noise and the error signal picked up by the error microphone, respectively.

This system can reduce MR noise at the error microphone inside the MRI room. However, the error microphone picks up the outside speech signal radiated from the secondary loudspeaker inside the MRI room. Therefore, the outside loudspeaker outputs the returned outside speech signal just like echo signal. Moreover, the signal from the outside loudspeaker also includes the residual MR noise picked up by the error microphone and prevent us from clear verbal communication. Hence, the only speech signal must be extracted.

Fig. 4. Head-mounted ANC system.

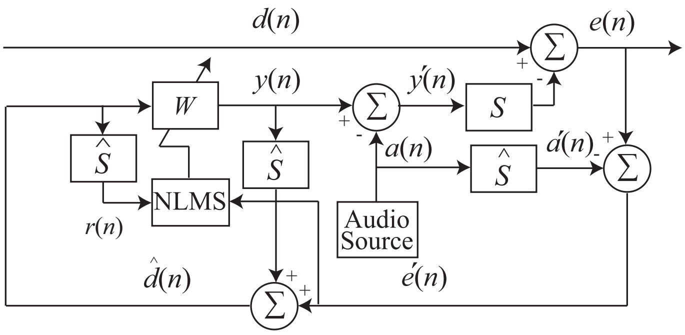


Fig. 5. Block diagram of the audio-integrated ANC system with the NLMS algorithm.

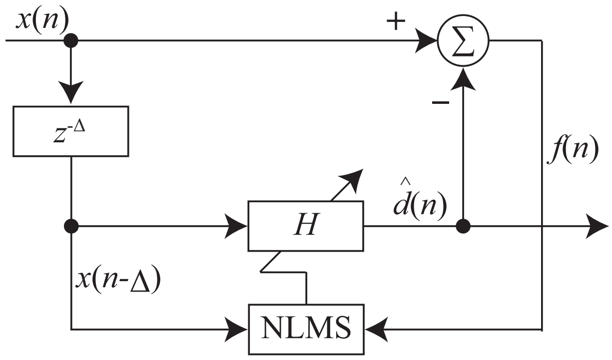


Fig. 6. Block diagram of the linear prediction filter with the NLMS algorithm.

D. Proposed ANC System for Speech Communication

To solve the problem of the feedback ANC system with speech communication, we propose a novel audio-Integrated ANC system with linear prediction filter. The block diagram of the proposed ANC system is illustrated in Figure 8. The update algorithm of the proposed system at sample time n is

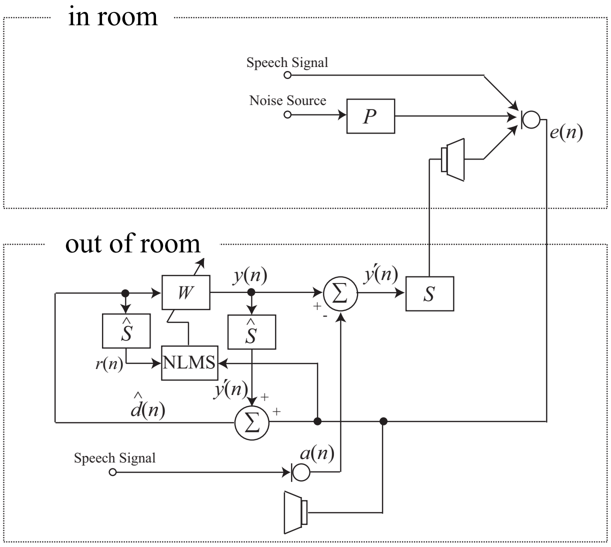


Fig. 7. Block diagram of feedback ANC system with speech communication.

expressed as

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\alpha_w}{||\mathbf{r}(n)||^2 + \beta_w}\mathbf{r}(n)x(n), \quad (3)$$

$$\mathbf{h}(n+1) = \mathbf{h}(n) + \frac{\alpha_h}{||\mathbf{x}(n-\Delta)||^2 + \beta_h}\mathbf{x}(n-\Delta)f(n),$$

Linear prediction filter produces no output in the case of an unpredictable broadband signal and passes only a predictable narrowband signal. Therefore, linear prediction filter can remove the residual MR noise from the signal $x(n)$ and realize more clear speech communication because loudspeaker outputs only speech signal.

IV. EXPERIMENTAL RESULTS

A. Noise Reduction and Separate Performance

We demonstrate the effectiveness of the head-mounted ANC system with speech communication through some experimental results. Figure 9 shows an arrangement of ANC system used in this experiment. The DSP used in this experiment is TMS320C6713 (Texas Instruments Co.product). Table I shows basic measurement conditions.

Figures 10 and 11 show the time waveform and the spectra at the error microphone, respectively. It can be seen from these figures that the proposed ANC system can operate stably and reduce MR noise about 25 dB effectively. Next, Figure **??** shows the comparison of the spectra between the input and output signal of linear prediction filter. It can be seen from this figure that the linear prediction filter can estimate MR noise and suppress speech signal about 10 dB. From the above results, the proposed ANC system can effectively reduce MR noise inside the MRI room and separate the residual noise and speech signal with linear prediction filter.

B. Subjective Assessment Test

We demonstrate whether the proposed ANC system can improve verbal conversation inside the MRI room and speech

TABLE I
MEASUREMENT CONDITIONS.

Input signal	MR noise
Tap length of secondary path model	200
Tap length of noise control filter W	300
Tap length of linear prediction filter H	200
Step size parameter W	0.01
Step size parameter LPF H	0.001
Regularization parameter β	1.0×10^{-6}
Sampling frequency	12000 Hz
Cut-off frequency of low-pass filter	2500 Hz
Delay	150

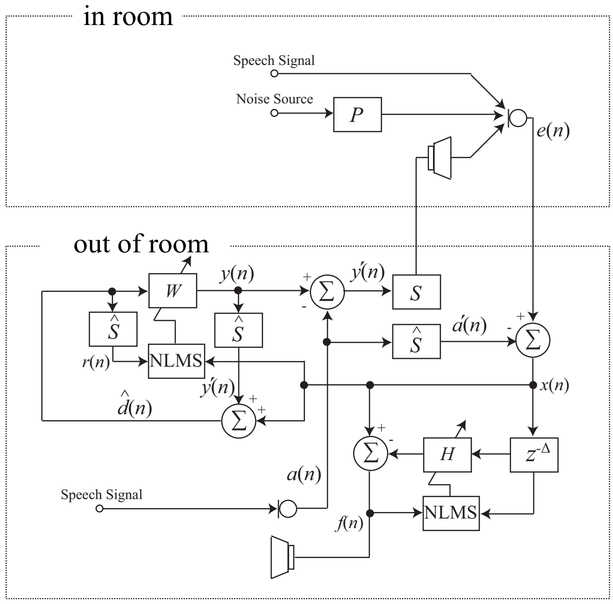


Fig. 8. Block diagram of the proposed ANC system including linear prediction filter for speech communication.

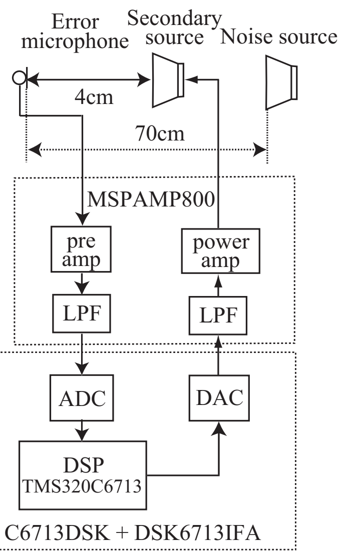


Fig. 9. Arrangement of the proposed ANC system implemented to DSP (TMS320C6713).

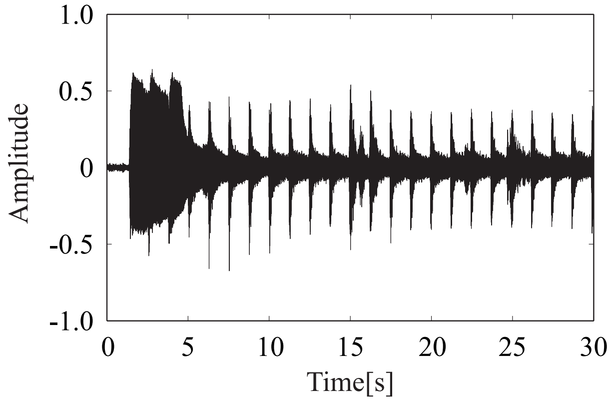


Fig. 10. Time waveform of the error signal picked up by the error microphone.

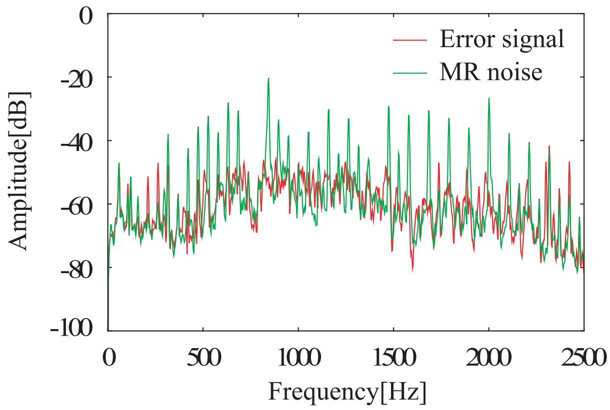


Fig. 11. Comparison of error spectra between before and after ANC.

communication between inside and outside rooms through some subjective assessment tests. The subjective assessment tests were conducted according to the articulation test in the cases of inside and outside a certain laboratory room assuming an MRI room. First, we conducted the assessment test inside the room (subject is a user putting on the head-mounted ANC system). In this test, subjects hear MR noise (about 95 dB) from a loudspeaker located in front of the subject (the distance is 50 cm) in the room. Moreover, randomized syllabic sound (about 70 dB) is simultaneously radiated with the anti-noise from the head-mounted ANC system. Subjects take down syllabic sound they could catch.

Next, we conducted the assessment test outside the room (subject is an operator outside the MRI room). In this test, randomized syllabic sound (about 70 dB) is simultaneously radiated with MR noise (about 95 dB) from the loudspeaker inside the room and they are picked up by the implanted microphone in HATS (head and torso simulator) and conveyed to a headphone subjects put on outside the room. Subjects take down syllabic sound they could catch. Twenty kinds of syllabic sound recorded in "Familiarity-controlled word lists 2003 (FW03)" in Tohoku University were used in these assessment tests. In the subjective assessment, the number of subjects was 10.

Tables II and III show the experimental results of the subjective assessment tests. It can be seen from these tables that the proposed ANC system can increase the articulation about 30 %. Therefore, the proposed ANC system can improve speech communication under MR noise environment. Moreover, it can be seen from Table III that linear prediction filter can improve the articulation score about 9 % compared with the

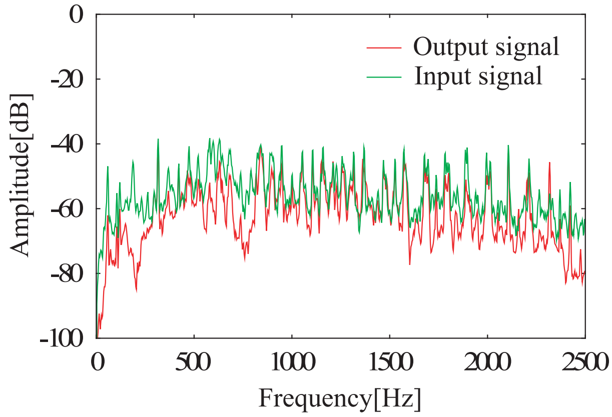


Fig. 12. Comparison of the spectra between the input and output signals of linear prediction filter.

TABLE II
SUBJECTIVE ASSESSMENT RESULT INSIDE THE ROOM (SUBJECT IS A USER PUTTING ON THE HEAD-MOUNTED ANC SYSTEM).

Number	ANC off	ANC on
1	7	13
2	8	11
3	0	13
4	8	12
5	8	11
6	8	11
7	9	13
8	8	12
9	8	11
10	8	13
sum	72	120
articulation score	3%	60%

TABLE III
SUBJECTIVE ASSESSMENT RESULT OUTSIDE THE ROOM (SUBJECT IS AN OPERATOR OUTSIDE THE MRI ROOM).

Number	ANC off	ANC on	LPF on
1	2	11	12
2	4	10	11
3	3	10	11
4	4	7	10
5	0	10	11
6	4	9	11
7	9	11	13
8	5	11	13
9	6	11	14
10	3	10	12
sum	40	100	118
articulation score	20%	50%	59%
t between ANC on and LPF on			-3.25

conventional ANC. Next, we confirm a significant difference between the proposed ANC system and the conventional one by t-test. According to t-distribution table, t is 2.878 in the case where twenty-eight-degree-of-freedom and significance level is 1 %. From Table III, t is 3.25 and less than -2.878. This is comprised in rejection region in the case where significance level is 1 %. Therefore, there is a significant difference between the proposed and conventional ANC systems.

V. CONCLUSION

In this paper, we proposed the head-mounted ANC system with speech communication and have confirmed the effectiveness of the proposed ANC system through some experiments. As a result, this system can operate stably, and reduce MR noise effectively, and separate speech signal from MR noise. Moreover, we conducted subjective assessment tests to confirm whether clear verbal communication can be realized. As a result, the subjective assessment test results indicated that the proposed ANC system have the effectiveness for speech communication under loud MR noise.

In the future, we will explore an appropriate parameter for linear prediction filter to separate speech signal and MR noise more accurately.

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