ADAPTIVE FEEDBACK ANC SYSTEM USING VIRTUAL MICROPHONES

Nobuhiro Miyazaki, Yoshinobu Kajikawa

Kansai University
Faculty of Engineering Science
Yamate-cho, Suita-shi, Osaka, 564-8680, Japan
k365960@kansai-u.ac.jp, kaji@kansai-u.ac.jp

ABSTRACT

In this paper, we apply a virtual sensing technique to a head-mounted ANC system we have already proposed. Adaptive feedback ANC system can reduce noise with periodicity or having narrow band components. However, since quiet zones are formed only at the locations of error microphones, an adequate noise reduction cannot be achieved at the locations where error microphones cannot be placed. In the head-mounted ANC system, the error microphones are placed near the opening of the ear canals. However, the intended locations at which the maximum noise reduction should be achieved are near eardrums. Hence, we try to introduce a virtual sensing technique to a head-mounted ANC system to achieve higher noise reduction. Experimental results demonstrate that the proposed system can achieve higher noise reduction than the conventional system. Moreover, a subjective assessment result demonstrates that the proposed system can also give higher noise reduction to human auditory.

Index Terms— Active noise control, Adaptive feedback ANC, Virtual sensing, Head-mounted ANC system, MR noise

1. INTRODUCTION

Recently, 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 [1] was introduced to conduct the microwave coagulation therapy assisted by near-realtime MR imaging. 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 [2].

We have already proposed an active noise control (ANC) system [3, 4] for reducing MR noise to solve this problem [5]. The proposed system was based on the internal model control (IMC) principle [6, 7]. The IMC-based feedback ANC system can reduce noise which has periodicity. In the head-mounted ANC system, the error microphones are placed near the opening of the ear canals. However, the intended locations at which the maximum noise reduction should be achieved are near eardrums where we cannot place the error microphones. A solution of this problem is to utilize a virtual sensing technique [8–16]. Virtual sensing ANC system can achieve higher noise reduction at the desired locations by measuring the system models from physical sensors to virtual sensors, which will be used in the online operation of the virtual sensing ANC algorithm. Hence, we try to achieve the maximum noise reduction near eardrums by using virtual sensing. However, it is impossible to place the microphone near eardrums. Therefore, we measure the system models from physical sensors to virtual sensors using HATS instead of human ears and try to achieve the maximum noise reduction near eardrums by implementing the virtual sensing technique to the head-mounted ANC system.

In this paper, we examine a virtual sensing ANC system and demonstrate the effectiveness through some experimental results and subjective assessment.

2. HEAD-MOUNTED ANC SYSTEM

2.1. 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 [1].

A typical time waveform of MR noise, measured with an optical microphone in an MR room, is shown in Figure 1 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.

2.2. 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 [17–19], Filtered-x NLMS (FXNLMS) [20] algorithm is the most popular. The block diagram of the adaptive feedback ANC system with the FXNLMS algorithm is illustrated in Figure 2. 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 $\boldsymbol{S}$ in the figure, is modeled by the secondary path model $\hat{\boldsymbol{S}}$, before the ANC system starts operating. The updating algorithm of the noise control filter is expressed as follows.

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

$$r(n) = \hat{\mathbf{s}}^{\mathrm{T}}\hat{\mathbf{d}}(n-1)$$

$$\mathbf{w}(n) = [w_1(n)\; w_2(n) \cdots w_i(n) \cdots w_N(n)]^{\mathrm{T}}$$

$$\mathbf{r}(n) = [r(n)\; r(n-1) \cdots r(n-N+1)]^{\mathrm{T}}$$

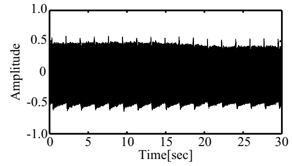
$$\hat{\mathbf{d}}(n) = [\hat{d}(n)\; \hat{d}(n-1) \cdots \hat{d}(n-N+1)]^{\mathrm{T}}$$

$$\hat{\mathbf{s}} \;\; = [\hat{s}_1\; \hat{s}_2 \cdots \hat{s}_i \cdots \hat{s}_N]^{\mathrm{T}}$$

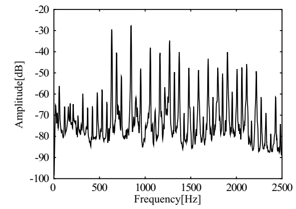
where $\mathbf{w}(n)$, $\mathbf{r}(n)$, and $\hat{\mathbf{s}}$ are a tap-weight vector of the noise control filter, filtered reference signal vector and tap-weight vector of secondary path model, respectively. Moreover, $\hat{d}(n)$ and $e(n)$ are a prediction of the primary noise $d(n)$ and error signal at the sample time n. Futhermore, α and β are a step size and regularization parameters, respectively.

2.3. Head-Mounted Structure

We developed a head-mounted ANC system [5]. 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.



(a) Time waveform



(b) Spectrum

Fig. 1. An example of time waveform and spectrum of MR noise.

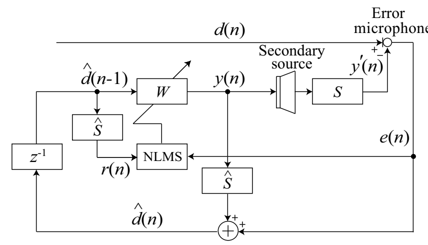
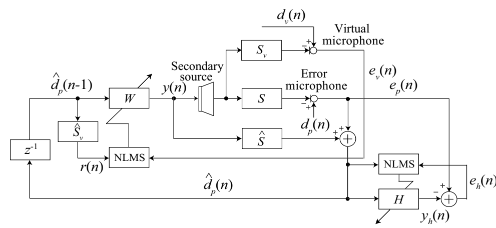
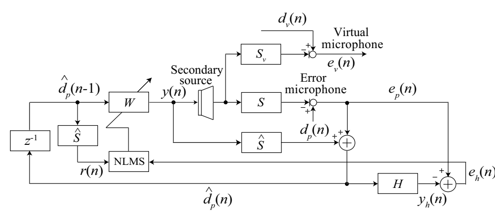


Fig. 2. Block diagram of adaptive feedback ANC system with the FXNLMS algorithm.



(a) Tuning stage



(b) Control stage

Fig. 3. Block diagram of adaptive feedback ANC system using virtual microphone.

3. FEEDBACK ANC SYSTEM USING VIRTUAL MICROPHONE

Feedback ANC system can reduce noise with periodicity or having narrow band components regardless of the direction of arrival. However, since quiet zones are formed only at the locations of error microphones, an adequate noise reduction cannot be achieved at the locations where error microphones cannot be placed. A solution of this problem is to utilize a virtual sensing technique. Currently, two classes of algorithms have been developed for virtual sensing. The first class of algorithms requires no offline training to obtain the system model. However, there are some constraints between microphones. Therefore, we apply the second class of algorithms, which requires system models. The block diagram of an adaptive feedback ANC system using virtual microphone [8] is illustrated in Figure 3. This system consists of two stages. In the tuning stage, the optimal noise control filter at the virtual microphone is estimated and an auxilary filter $\boldsymbol{H}$ which preserves the information of the optimal noise control filter is determined. In the control stage, the maximum noise reduction can be realized at the desired location by using the auxiliary filter $\boldsymbol{H}$ determined in the tuning stage. In this figure, $\boldsymbol{S_v}$ shows the secondary path from the secondary source to the virtual microphone.

The updating algorithms of the noise control filter $\boldsymbol{W}$ and auxiliary filter $\boldsymbol{H}$ in the tuning stage are expressed as follows.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\alpha_w}{||\mathbf{r}(n)||^2 + \beta_w}\mathbf{r}(n)e_v(n) \tag{2}$$

$$\mathbf{h}(n+1) = \mathbf{h}(n) + \frac{\alpha_h}{||\hat{\mathbf{d}}_p(n)||^2 + \beta_h}\hat{\mathbf{d}}_p(n)e_h(n) \tag{3}$$

$$r(n) = \mathbf{s}_v^{\mathrm{T}}\hat{\mathbf{d}}_p(n-1) \tag{4}$$

where α_w and α_h show the step size parameters of the noise control filter $\boldsymbol{W}$ and the auxiliary filter $\boldsymbol{H}$ and β_w and β_h show the reguralization parameters of the noise control filter $\boldsymbol{W}$ and the auxiliary filter $\boldsymbol{H}$, respectively.

The updating algorithm of the noise control filter $\boldsymbol{W}$ in the control stage is expressed as follows.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\alpha_w}{||\mathbf{r}(n)||^2 + \beta_w}\mathbf{r}(n)e_h(n) \tag{5}$$

$$r(n) = \mathbf{s}^{\mathrm{T}}\hat{\mathbf{d}}_p(n-1) \tag{6}$$

We explain the principle of this system. The purpose of the tuning stage is to obtain the auxiliary filter $\boldsymbol{H}$, which includes the optimal noise control filter at the virtual microphone. First, the noise control filter is updated by placing the microphone at a desired location to reduce noise at the location. After the noise control filter converges, the auxiliary filter $\boldsymbol{H}$ is updated. By this way, the noise control filter converges to

$$W(z) = 1/S_v(z), \tag{7}$$

where $W(z)$ and $S_v(z)$ represent the transfer functions of the noise control filter and the secondary path for the virtual microphone, respectively. Moreover, the auxliary filter converges to

$$H(z) = 1 - S(z)/S_v(z) = 1 - S(z)W(z), \tag{8}$$

where $H(z)$ and $S(z)$ represent the transfer functions of the auxiliary filter and the secondary path for the real error microphone, respectively. It can be seen from this equation that the auxiliary filter contains the information of the optimal noise control filter for the virtual microphone.

On the other hand, in the control stage, the modified erorr $e_h(n)$ is represented by

$$\begin{aligned} E_h(z) &= E_p(z) - H(z)D_p(z) \\ &= \left\{D_p(z) - S(z)W(z)\hat{D}_p(z)\right\} \\ &\quad - \{1 - S(z)/S_v(z)\}\, D_p(z), \end{aligned} \tag{9}$$

where $E_h(z)$, $E_p(z)$, $D_p(z)$, and $\hat{D}_p(z)$ are the Z transfoms of $e_h(n)$, $e_p(n)$, $d_p(n)$, and $\hat{d}_p(n)$, respectively. If

$$D_p(z) = \hat{D}_p(z), \tag{10}$$

then

$$\begin{aligned} E_h(z) &= \{1 - S(z)W(z)\}\, D_p(z) \\ &\quad - \{1 - S(z)/S_v(z)\}\, D_p(z) \\ &= \{S(z)/S_v(z) - S(z)W(z)\}\, D_p(z). \end{aligned} \tag{11}$$

Hence, the noise control filter converges to the optimal transfer function given by (7) using the modified error signal $e_h(n)$ for the update algorithm of the noise control filter. In other words, the optimal noise control filter for the virtual microphone can be obtained.

In this paper, we try to achieve the maximum noise reduction near eardrums. However, it is impossible to place the microphone near eardrums. Therefore, the implanted microphone of HATS (Head and Torso Simulator) is used in the tuning stage, and then the auxiliary filter $\boldsymbol{H}$ is obtained. Since the distance between the error microphone and eardrums does not almost change in the head-mounted system, the noise can be reduced at the desired location (near user's eardrum) using the auxiliary filter $\boldsymbol{H}$ determined for HATS.

4. EXPERIMENTAL RESULTS

4.1. Noise Reduction Performance

We demonstrate that the head-mounted ANC system with virtual microphones can improve the noise reduction performance for MR noise compared with the conventional one (without virtual sensing) through some experiments. The DSP used in this experiment is TMS320C6713(Texas Instruments Co.product). Table 1 shows basic measurement conditions. As the virtual microphones, the microphones implanted in HATS(Head and Torso Simulator) were used and the auxiliary filter $\boldsymbol{H}$ was estimated in the tuning stage.

Figure 4 shows the time waveform at the desired location (the microphone implanted in HATS) and the comparison of the spectra at the desired location between the proposed and the conventional ANC systems. It can be seen from these figures that the proposed adaptive feedback ANC system using virtual microphone can improve the noise reduction performance about 20 dB at the desired location compared with the conventional one.

Table 1. Measurement conditions.

Tap length of noise control filter $\boldsymbol{W}$	500
Tap length of $\boldsymbol{H}$	500
Tap length of secondary path model	200
Step size parameter of $\boldsymbol{W}$	0.1
Step size parameter of $\boldsymbol{H}$	0.01
Regularization parameter β	1.0×10^{-6}
Sampling frequency	12000 Hz
Cut-off frequency of low-pass filter	2500 Hz

Table 2. The samples for the subjective assessment.

Sample 1	MR noise + ANC with virtual sensing
Sample 2	MR noise + ANC without virtual sensing
Sample 3	MR noise (without ANC)

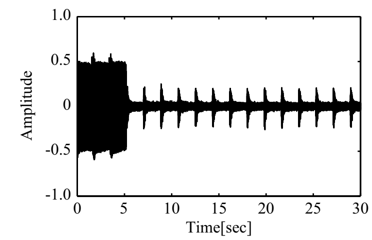
4.2. Subjective Assessment Experiment

We made a subjective assessment of the noise reduction performance between the proposed adaptive feedback ANC with virtual sensing and the conventional one. The subjective assessment was conducted according to the Scheffe's paired comparison method. In this experiment, the subjects select an appropriate sample according to the noise reduction performance and evaluate the sample based on three options (former, same, and latter). In the subjective assessment, the number of subjects was 11 and the sound pressure level were set to 90 dB in SPL at the location where the subjects sat. We compared the three samples shown in Table 2; ANC with virtual sensing, ANC without virtual sensing, and without ANC.

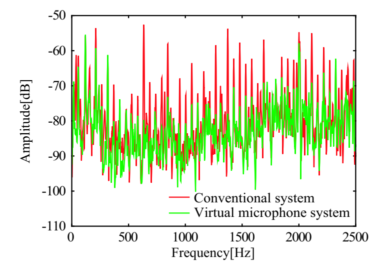
Figure 5 shows the experimental result of the subjective assessment. In this figure, the vertical axis represents the interval scale and the dots and bars of each sample represent the average and the 95% confidence interval. If the average of a certain sample is outside the bar of another sample, then it means a significant difference between the samples. It can be seen from this figure that there is a significant difference between the proposed and conventional ANC systems. Hence, the effectiveness of the adaptive feedback ANC with virtual sensing was also demonstrated in human auditory regardless of using HATS in the tuning stage.

5. CONCLUSION

In this paper, we have confirmed the effectiveness of the adaptive feedback ANC system using virtual microphone through some experiments. As a result, this system can improve the noise reduction performance at the desired location. In the future we will examine the effectiveness of the proposed ANC system in an actual MRI room.



(a) Time waveform at the desired location (the microphone implanted in HATS)



(b) Comparison of the spectra at the desired location between the proposed and the conventional ANC systems

Fig. 4. Experimental results of the proposed adaptive feedback ANC system with virtual sensing.

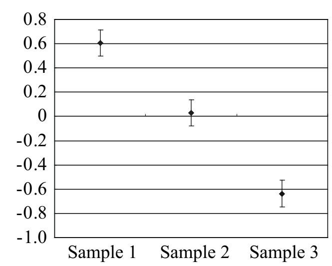


Fig. 5. Experimental result of the subjective assessment.

6. RELATION TO PRIOR WORK

This work presented here has treated the adaptive feedback ANC with virtual sensing implemented to the head-mounted ANC system we have already proposed [5]. The used virtual sensing technique is based on the method proposed by Pawelczyk [8]. The work by Pawelczyk focused on the movements of quiet zones from the error sensor to the desired location in the free field. On the other hand, we focused on the movements of quiet zones from the opening of the external ear canal to the vicinity of the ear drum using the HATS. Although the auxiliary filter $\boldsymbol{H}$, which plays an important role in the virtual sensing technique, was estimated for HATS in the tuning stage because the microphone cannot be placed near the ear drum, the user could perceive noise reduction. This fact wes demonstrated through the subjective assessment. These are our novel contributions.

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