Proceedings of 20th International Congress on Acoustics, ICA 2010

23-27 August 2010, Sydney, Australia

A Study of Dynamic Determination of Filter Length for Multi-channel Active Noise Control

Masanori Morise (1), Takayuki Iriki (2), Yasuhiro Kuratani (1), Hiroaki Ishii (2) and Takanobu Nishiura (1)

(1) College of Information Science and Engineering, Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, 5258577, Shiga, Japan
(2) Graduate School of Science and Engineering, Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, 5258577, Shiga, Japan

PACS: 43.60.Ac

ABSTRACT

The environmental noises such as the room air conditioner are recently one of the most major problems in urban cities. Several noise reduction methods for the environmental noise have been proposed to maintain the comfortable environment. ANC (Active Noise Control) is utilized to reduce the sound pressure level of the environmental noise by using two microphones (Reference microphone and Error microphone) and one noise-cancellation loudspeaker. The noise-cancellation loudspeaker emits the sound wave with the same amplitude but with the inverse phase at the error microphone. The noise and the sound wave emitted by the loudspeaker are cancelled at the error microphone position. Designing a filter to efficiently reduce the noise is required to compensate the transfer function from the noise-cancellation loudspeaker to the error microphone. Multi-channel ANC system with several microphones and loudspeakers has been proposed to improve the noise reduction performance. As it uses several microphones and loudspeakers, a lot of filters are required to efficiently reduce the noise. Although the suitable filter lengths of each filter are determined by each transfer function, the same filter length is used to design all filters in multi-channel ANC. In this paper, a dynamic filter length determination method for multi-channel ANC is proposed to avoid the use of the unsuitable filter lengths. The objective experiment in a real environment was conducted to demonstrate the effectiveness of the proposed method. As a result, we confirmed that the proposed method can determine the suitable filter length of each filter.

INTRODUCTION

There are a lot of environmental noises such as air conditioner, road noise and train noise, in urban cities. Since such noises are disagreeable to the ear, a method to remove these noises is indispensable to maintain the comfortable environment. PNC [1], [2] (Passive Noise Control) and ANC [3], [4] (Active Noise Control) have generally been used to overcome the noise problem.

PNC is the noise reduction method by using sound insulation walls and is utilized in various places such as cinemas and concert halls. Thick walls for sound insulation can reduce the noise by diffraction, interference and attenuation of the sound waves. Although PNC can effectively reduce the noise in higher frequency region, reduction of the noise in lower frequency region is difficult because of the sound transmission loss in the wall.

ANC is utilized to reduce the noise in lower frequency region by using two microphones (Reference microphone and error microphone) and one noise-cancellation loudspeaker. The noise-cancellation emits the sound wave with the same amplitude but with the inverse phase. The emitted sound wave is created based on the sound wave observed at the reference microphone. The noise and the sound wave emitted by the loudspeaker are cancelled at the error microphone position. Designing a filter to efficiently reduce the sound pressure level of the noise is required for compensation of the transfer function from the noise-cancellation loudspeaker to the error microphone. Although ANC is useful to reduce the one noise source at one error microphone position, the control of the wide area is impossible because the reduction performance is limited to near the error microphone.

Multi-channel ANC system has been proposed to overcome this problem by using several reference microphones, error microphones and noise-cancelling loudspeakers. Since the control point to reduce the noise increases, multi-channel ANC can reduce a lot of noise sources. Designing many filters is required to achieve the efficient performance because there are a lot of transfer functions between each microphone and loudspeaker.

The filter length for the multi-channel ANC is crucial to determine the noise reduction performance because the transfer function isn't compensated by using the filter that has the shorter length than the impulse response of the transfer functions. The suitable filter length depends on the transfer functions between a noise-cancelling loudspeaker and an error microphone. Although suitable lengths for each filter are different, the same length is used for multi-channel ANC.

The dynamic filter length determination method based on the iterative algorithm is proposed to overcome this problem. The all filter lengths to compensate the all transfer functions are automatically determined by the proposed method. In this paper, an evaluation experiment is conducted to demonstrate the effectiveness of the proposed method. As a result, we confirmed that the proposed method is superior to the conventional one in the filter length, provided that the noise reduction performance is the same.

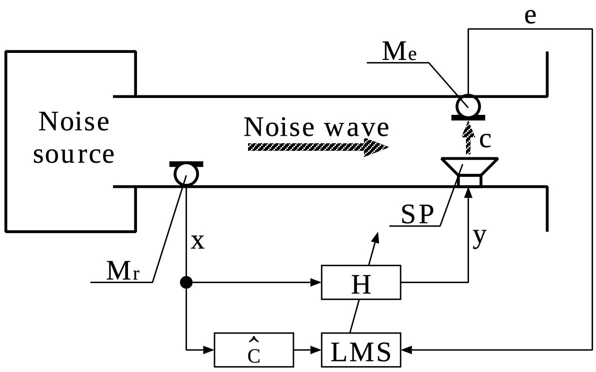

Figure 1. The ANC system based on the Filtered-X algrithm

ANC BASED ON THE FILTERED-X LMS ALGORITHM

Figure 1 represents an ANC system consisted of one reference microphone (Mr), one error microphone (Me) and one noise-cancelling loudspeaker (SP). It required estimating the transfer function c. The noise reduction performance depends on the estimation performance in the estimated transfer function $\hat{c}$. The noise source is reduced by cancelling the sound wave emitted by the noise cancelling loudspeaker. Filtered-X algorithm [5] is generally utilized to develop the ANC system. LMS algorithm is utilized to design the adaptive filter H. Filter H is updated to reduce the sound wave observed by the error microphone Me until the filter converges. Basic ANC system can reduce the noise at one error microphone position.

Multi-channel ANC

Multi-channel ANC [3] has been proposed to reduce a number of noise sources at the error microphone positions. A lot of reference microphones, error microphones and noise-cancelling loudspeakers are used in this system. The number of filters used for the ANC is the product of the number of reference microphones, error microphones and noise-cancelling loudspeakers. Since LMS algorithm does not include the change of the filter length, the determination of suitable filter length is crucial to achieve the high noise reduction performance. The suitable lengths for each filter are different because it depends on the transfer function. Therefore, a long length is utilized to design all filters. However, the noise reduction performance decreases, provided that the unsuitable filter length is utilized.

PROPOSED METHOD

The filter length dynamic determination method is proposed to improve the noise reduction performance for the multi-channel ANC. As shown in Figure 2, the proposed method utilizes the iterative algorithm. ANC can control one noise source, while MISO (Multiple-Input and Single-Output)-ANC can reduce a number of noise sources.

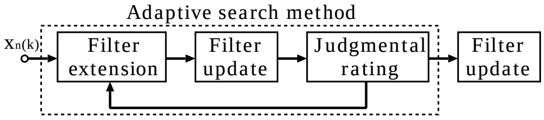

Figure 2. Block diagram of the proposed method.

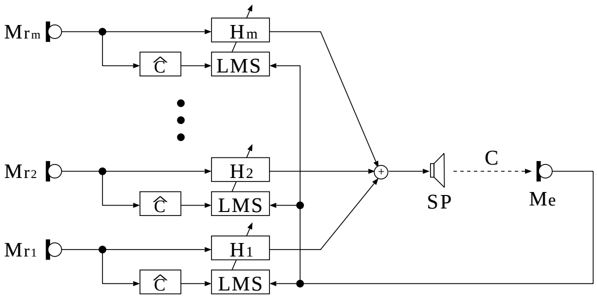

Figure 3. Multi-channel ANC system used for the experiment.

Filter length determination method

A short filter length is assigned as the initial value. The filter length is then updated after a certain number of iteration γ . The parameter to determine the filter length is calculated by the following equations.

$$P = \sum_{m=1}^{n} \mathbf{w}_m(k)\mathbf{w}_m(k)^T , \tag{1}$$

$$Q_n = \left[wl_{n-\alpha} \dots wl_n\right]\left[wl_{n-\alpha} \dots wl_n\right]^T , \tag{2}$$

where, P represents the total gain of the filter, Q_n represents the gain of the extended part of the filter and α represents the length used for the extension. $\mathbf{w}(k)$ represents the filter coefficients in the time k , l represents the filter length and n represents the channel number of the reference microphone. The filter length is determined by the following equation.

$$l_n = \begin{cases} l_n + \alpha & \text{if } Q_n / P > \beta \\ l_n & \text{otherwise.} \end{cases} , \tag{3}$$

where β represents the threshold value. The filter length is extended while the conditional equation is fulfilled. The initial coefficient of the extended part is 0. Therefore, the filter $w_n(x)$ is extended based on the following equation.

$$w_n(x) = \begin{cases} w_n(x) & \text{if } x < l_n - \alpha \\ 0 & \text{otherwise} \end{cases} , \tag{4}$$

where the range of x is from 0 to l_n . The conclusive filter is obtained, provided that the conditional equation is not fulfilled. The proposed method has three parameters (the iteration update number γ , threshold β and the extension filter length α). Although these parameters should be optimized, this is not focused in this paper. The parameters empirically determined by the exploratory experiment are used for the experiments.

EXPERIMENTS AND RESULTS

The experiment to verify the effectiveness of the proposed method was conducted by using the sound recorded in a real environment. As shown in Fig. 3, the number of the reference microphone is 1, the number of the error microphone is 7 and the number of the noise-cancelling loudspeaker is 1. The noise reduction performance and the filter length are examined in this experiment. The proposed method is compared with the multi-channel ANC that utilizes the same length filters.

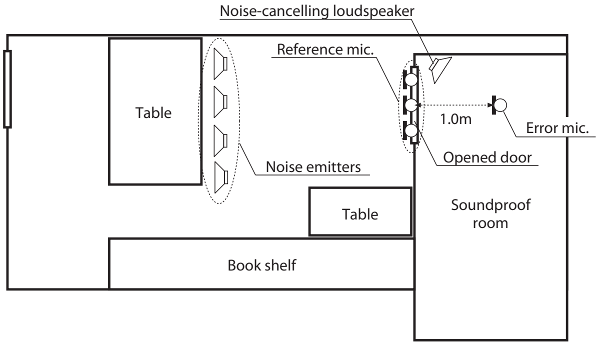

Figure 4. Experimental environment.

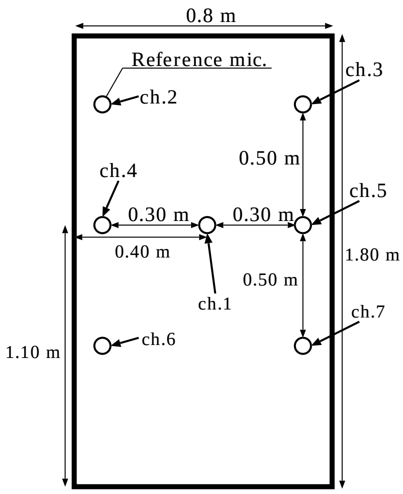

Figure 5. The reference microphone positions for the expeiment.

Table 1. Conditions for the proposed method.

Conditions of ANC	
Secondary path	500
Secondary path model	200
Sampling frequency	6000
Step size	10^{-6}
Conditions of the proposed method	
Extension filter length	16
Threshold	$10^{-6} < \beta < 10^{-2}$
Iteration updata number	2000

Conditions for the experiment

Environmental noises recorded in a building are utilized as the noise for reduction. The experimental environment is shown in Fig. 4. The noise emitters are utilized to create a number of noise sources. Figure 5 shows the reference microphone positions.

Figure 6. Photograph of the experimenta environment..

The target transfer function c is measured by using TSP (Time-Stretched Pulse) [6]. The length of the TSP signal is 65536 samples (1.4 sec) and the sampling frequency is 48000 Hz. The background noise of the sound proof room is 27 dBA.

The extension filter length α is 16, the thresholds β of each filter are different to determine the filter length and the iteration update filter γ is 2000. These values are determined empirically. The conditions for ANC and the proposed method are shown in Tab. 1. The filter length used for the conventional ANC is determined by the average filter length of the proposed method.

Evaluation indices

NRP (Noise Reduction Performance) is utilized to evaluate the proposed method. In this experiment, the frequency region from 0 Hz to 1000 Hz is used to calculate the NRP because ANC is useful to reduce the low-frequency noise.

$$NRP = 10\log_{10}\left(\frac{\int_0^{1000}\left|N_b(f)\right|^2 df}{\int_0^{1000}\left|N_a(f)\right|^2 df}\right), \quad (5)$$

where, $N_b(f)$ represents the noise spectrum before ANC processing. $N_a(f)$ represents the noise spectrum after ANC processing. f represents the frequency. NRP increases, provided that the high noise reduction performance is achieved.

NMSE (Normalized Mean Square Error) is utilized to evaluate the convergence characteristic. It is defined as the following equation.

$$NMSE = 10\log_{10}\left(\frac{E\left[e^2(k)\right]}{E\left[d^2(k)\right]}\right), \quad (6)$$

where, $E\left[e^2(k)\right]$ represents the average power of the error signal, $E\left[d^2(k)\right]$ represents the average power of the noise signal.

Results

The shortest filter length was 112, while the longest filter length was 320. Therefore, the suitable filter length is different from the microphone positions. The average filter length was 185.1 in the proposed method. The filter length for the conventional ANC is 185 in this experiment.

Figure 6 illustrates the noise reduction performance, and Fig. 7 illustrates the close up in Fig. 6. The horizontal axis represents the average filter length and the vertical axis represents the NRP. The average filter length for the proposed method is determined by controlling the threshold β . The proposed method is superior to the conventional method in NRP, provided that the average filter length is from 100 to 250. NRP of the proposed method is inferior to that of the conventional method, provided that the filter length is less than 100 Hz. The noise reduction performance of the proposed method and the conventional method is the same, provided that the filter length is more than 250.

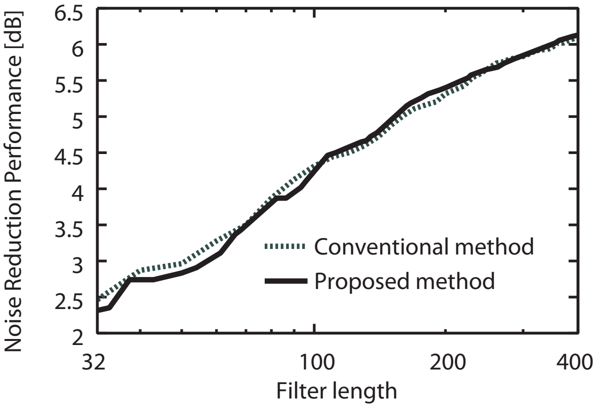

Figure 7. Relationship between the filter length and the noise reduction performance.

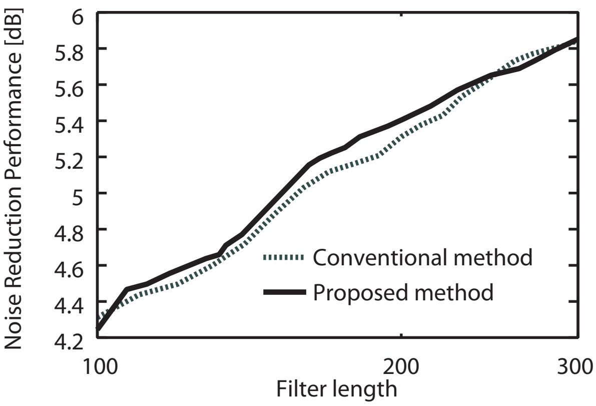

Figure 8. Relationship between the filter length and the noise reduction performance (Filger length is from 100 to 300).

Figure 9 illustrates the NMSEs of both methods. It shows that the convergence of the proposed method was inferior to the conventional method. Since the convergence depends on the threshold β , the parameter optimization for the proposed method is crucial in the future work.

DUSCUSSIONS

The suitable filter lengths for all filters were determined by the proposed method because the proposed method is superior to the conventional method in NRP. However, the proposed method is inferior to the conventional method in NRP provided that the average filter length is under 100 Hz and over 250 Hz. It suggests that the length of the transfer function C for estimation is about 180. The proposed method could not design the suitable filter because the target filter length was shorter than optimal length.

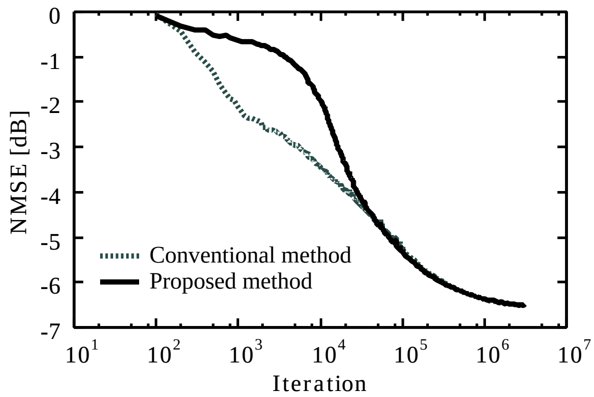

Figure 9.Relation between the number of iteration and NMSE.

The noise reduction performance of the proposed method and the conventional method is the same, provided that the average of the filter length is over 250. It suggests that the filter that is longer than the optimal length could not improve the noise reduction performance.

CONCLUSIONS

The dynamic filter length determination method was proposed to improve the noise reduction performance in multichannel ANC. The experiment by the objective experiments based on the environmental sound to verify the effectiveness of the proposed method. As a result, we confirmed that the proposed method could determine the suitable lengths of all filters. Furthermore, we also confirmed that the proposed method was superior to the conventional method in filter length, provided that the noise reduction performance is the same.

The parameter optimization is crucial for the proposed method to achieve the higher performance. The dynamic determination of the parameter for the proposed method will also be examined in our future work.

ACKNOWLEDGEMENTS

This research was partly supported by Grant-in-Aid by JSPS.

REFERENCES

1 U.J. Kurze, “Noise Reduction by Barriers,” J. Acouust. Soc. Am., vol.55, pp.504-518, 1974.
2 M. Koyasu, H. Tachibana, A. Arimoto, Y. Tokita, J. Igarashi, “Noise and Vibration,” CORONA PUBLISHING CO., LTD., Japan, 1982.
3 S.J. Elliott, I.M. Stothers and P.A. Nelson, “A multiple error algorithm and its application to active control of sound and vibration,” IEEE Trans. Acoust. Speech Signal Process. vol.ASSP-35 pp.1423-1434, 1987.
4 S.J. Elliott and P.A. Nelson, “Active noise control,” IEEE Signal Processing Magazine, vol.10, pp.12-35, 1993.
5 B. Widrow and S. D. Stearns, “Adaptive Signal Processing, Englewood cliffs,” NJ: Prentice-Hall, 1985.
6 A.J. Berkhout, D.de Vries and M.M. Boone, “A new method to acquire impulse responses in concert halls,” J. Acoust. Soc. Am., vol.68, no.1, pp.179-183, 1980.