MULTI-CHANNEL ACTIVE NOISE CONTROL WITH FREELY MOVABLE ERROR MICROPHONES

Yoshinobu Kajikawa and Yasuo Nomura

Department of Electronics, Faculty of Engineering, Kansai University
3-3-35, Suita-shi, Osaka 564-8680, Japan

ABSTRACT

In this paper, we present a novel multi-channel active noise control (ANC) system with freely movable error microphones. This ANC system uses a simultaneous perturbation algorithm and has an advantage that secondary path models (estimation of secondary paths) are not required unlike the conventional MEFX (Multiple Error Filtered-X) based ANC. This system can consequently control noise stably because there are not modeling errors which cause system instability. The computational complexity is also very small. We demonstrate that the proposed multi-channel ANC system can operate stably under the environment where the error microphones always move.

1. INTRODUCTION

Active noise control (ANC) [1, 2] has been recently applied to a wide variety of acoustic noise problems. In the single-channel feedfoward ANC system, the FX (Filtered-X) based algorithms are usually used as an algorithm to update the adaptive filter coefficients. The FX based algorithms require an estimate of the secondary path (from the secondary source generating anti-noise to the error sensor detecting residuals) prior to the operation of the ANC system. The estimated model is often called a secondary path model. However, the secondary path model generally differs from the physical one. In this case, the errors between the secondary path and its model lead to system instability or suboptimal performance [3, 4]. The modeling error is consequently the most significant problem in the ANC system.

Some approaches to solving this problem include ANC systems without secondary path models. One of these approaches involves a technique that employs the simultaneous equations method [5]. In this technique, another auxiliary filter is used to create the noise control filter. This technique can converge faster than the conventional FX based ANC systems. However, this technique requires a complex system composition.

This research was financially supported by MEXT KAKENHI(14750320).

Another technique uses SP (Simultaneous Perturbation) algorithms [6, 7, 8, 9]. In the SP algorithms, the coefficients of the adaptive filter are updated by two error signals: one is the error between the output of the adaptive filter and the desired signal; the other is the error between the output of the adaptive filter with a little perturbation and the desired signal. This technique can update the filter coefficients using only error signals. Hence, the ANC system using this technique does not require the secondary path model. Moreover, the system composition is smaller than that of the conventional FX based ANC systems.

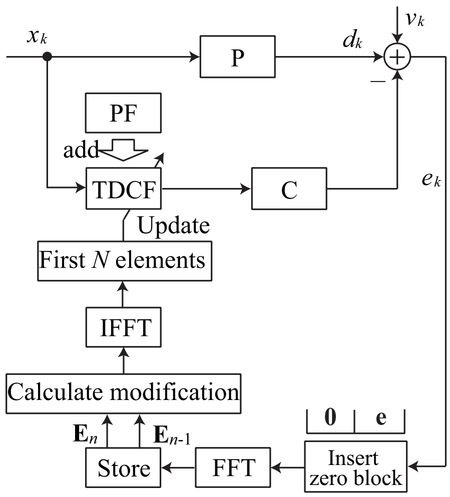
This paper presents a multi-channel active noise control (ANC) system using the SP algorithm. This system has an advantage that secondary path models (estimation of secondary paths) are not be required and can consequently control noise stably, that is, this system has a grate advantage that the error microphones can be freely moved while the ANC system controls noise because there are not modeling errors which cause system instability. The computational complexity is also very small.

2. SIMULTANEOUS PERTURBATION ALGORITHM

Simultaneous perturbation (SP) algorithms calculate an instantaneous estimate of gradient vector by adding a little perturbation to the coefficient of noise control filter. In this paper, the frequency domain time difference simultaneous perturbation (FD-TDSP) algorithm [9] is explained.

2.1. FD-TDSP

FD-TDSP is an algorithm which updates the filter coefficients once every N samples by using only one kind of error signal. Perturbations are always added to the coefficients of the noise control filter, and the error signals are stored within the block period N. The filter coefficients at the nth block are updated by using two error vectors obtained within the nth and $(n-1)$th block periods. The ANC system using the FD-TDSP is shown in Fig. 1. In this system, the filtering of the noise control filter is done on the time



P: Primary Path, C: Secondary Path
TDCF: Time Domain Control Filter
PF: Perturbation Filter

Fig. 1. Block diagram of the ANC system using the frequency domain time difference simultaneous perturbation method.

domain and the updating is done on the frequency domain so that the ANC system can reduce noise in real time. The updating algorithm at the nth block is defined as follows:

$$
\begin{aligned}
\mathbf{w}_{n+1} &= \mathbf{w}_n - \mu \Delta \mathbf{w}_n & (1)\\
\Delta \mathbf{w}_n &= \text{first } N \text{ elements of IFFT}[\mathbf{U}_n] & (2)\\
\mathbf{U}_n &= \text{diag}[\mathbf{S}_n] \frac{\text{diag}[\mathbf{E}_n^*]\mathbf{E}_n - \text{diag}[\mathbf{E}_{n-1}^*]\mathbf{E}_{n-1}}{c_n} & (3)\\
\mathbf{E}_n &= \text{FFT}[0 \cdots 0\ e_{nN+1} \cdots e_k \cdots e_{(n+1)N}]^T\\
\mathbf{S}_n &= [S_n(1)\ S_n(2) \cdots S_n(i) \cdots S_n(2N)]^T
\end{aligned}
$$

where μ is the step-size parameter, c_n is the magnitude of the perturbation, and $\mathbf{S}_n$ is a complex vector whose elements are -1 or 1 in both real and imaginary parts and has the following characteristics:

$$
\begin{aligned}
&E[\text{Re}\{S_n(i)\}] = 0,\ E[\text{Im}\{S_n(i)\}] = 0\\
&\text{Re}\{S_n(i)\}^2 = 1,\ \text{Im}\{S_n(i)\}^2 = 1\\
&E[\text{Re}\{S_n(i)\}\text{Re}\{S_m(i)\}] = 0,\ n \neq m\\
&E[\text{Im}\{S_n(i)\}\text{Im}\{S_m(i)\}] = 0,\ n \neq m\\
&E[\text{Re}\{S_n(i)\}\text{Re}\{S_n(j)\}] = 0,\ i \neq j\\
&E[\text{Im}\{S_n(i)\}\text{Im}\{S_n(j)\}] = 0,\ i \neq j\\
&E[\text{Re}\{S_n(i)\}\text{Im}\{S_n(j)\}] = 0
\end{aligned}
$$

2.2. Multi-channel FD-TDSP

In this paper, the updating algorithm in the experiment is the multi-channel FD-TDSP algorithm applying the FD-TDSP algorithm to multi-channel ANC systems. Multi-channel ANC systems with J reference microphones, L secondary sources, and M error microphones are defined as CASE (J,L,M).

The updating algorithm of CASE(J,L,M) is defined as follows:

$$
\begin{aligned}
\mathbf{w}_{lj,n+1} &= \mathbf{w}_{lj,n} - \mu \Delta \mathbf{w}_{lj,n} & (4)\\
\Delta \mathbf{w}_{lj,n} &= \text{first } N \text{ elements of IFFT}[\mathbf{U}_{lj,n}] & (5)\\
\mathbf{U}_{lj,n} &= \text{diag}[\mathbf{S}_{lj,n}] \sum_{m=1}^{M} \big\{ \text{diag}[\mathbf{E}_{m,n}^*]\mathbf{E}_{m,n}\\
&\quad - \text{diag}[\mathbf{E}_{m,n-1}^*]\mathbf{E}_{m,n-1} \big\} / c_n & (6)
\end{aligned}
$$

$$
\begin{aligned}
\mathbf{E}_{m,n} &= \text{FFT}[0 \cdots 0\ e_{m,nN+1} \cdots e_{m,k} \cdots e_{m,(n+1)N}]^T\\
\mathbf{w}_{lj,n} &= [w_{lj,n}(1)\ w_{lj,n}(2) \cdots w_{lj,n}(i) \cdots w_{lj,n}(N)]^T\\
\mathbf{S}_{lj,n} &= [S_{lj,n}(1)\ S_{lj,n}(2) \cdots S_{lj,n}(i) \cdots S_{lj,n}(2N)]^T
\end{aligned}
$$

where $\mathbf{w}_{lj,n}$ is the coefficient vector of the noise control filter and $\mathbf{S}_{lj,n}$ is the frequency domain simultaneous perturbation vector. Moreover, c_n is the magnitude of the perturbation and defined as follows:

$$
c_n = \sqrt{\frac{\alpha J \sum_{m=1}^{M} \sum_{k=(n-1)N+1}^{nN} e_{m,k}^2}{G^2 M \sum_{j=1}^{J} \sum_{k=(n-1)N+1}^{nN} x_{j,k}^2}} \qquad (7)
$$

where G is the mean gain of all secondary paths and α is a coefficient that defines a ratio of the power of the perturbation to the error signal.

3. COMPUTATIONAL COMPLEXITY

In the case of controlling the noise in the real-time processing, one of the most significant elements is the computational complexity in the algorithm. Therefore, the computational complexity in the multi-channel FD-TDSP is compared with the MEFX based algorithms. By the way, the perturbation algorithm is realized by the block processing of the block period N and the computational complexity required in the algorithm generally depends on the number of multiplication. Hence, the number of multiplication required within sample period N is compared in each algorithm. We assume that the tap lengths of the noise control filters and the secondary path models are N.

Table 1. Number of multiplication required during N samples in the multiple-channel ANC systems of CASE(J, L, M).

MEFXLMS	$(2JLM + JL)N^2 + JLN$
FD-MEFXLMS	$JLN^2 + (JL + J + M)8N \log_2 2N + (16JLM + JL)N$
FD-MEFXLMS with Online Modeling	$JLN^2 + (JL + LM + J + M)8N \log_2 2N + (16JLM + 20LM + JL)N$
FD-TDSP	$JLN^2 + (JL + M)8N \log_2 2N + (12JL + 4M)N$

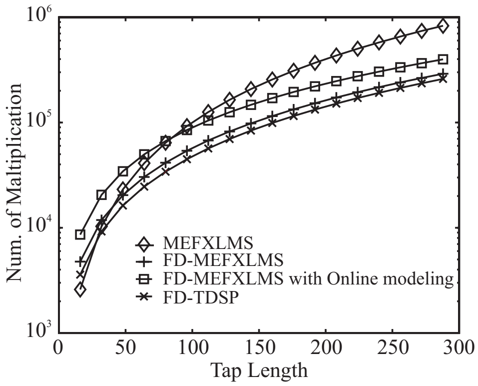


Fig. 2. Comparison of the number of multiplication in each algorithm (CASE(1,2,2)).

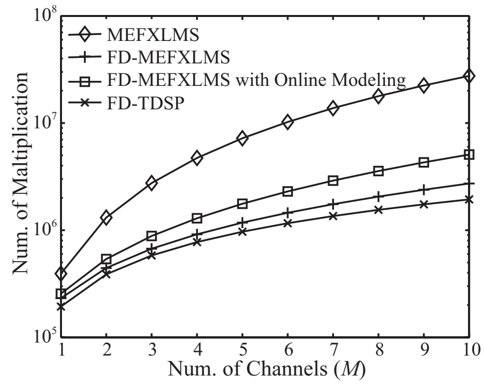


Fig. 3. Comparison of the number of multiplication in each algorithm (CASE(2,M,M)).

The algorithms used in the multi-channel ANC systems are the MEFXLMS, the FD-MEFXLMS updating the coefficients of the noise control filter on the frequency domain, FD-MEFXLMS with the online modeling which updates secondary path models online, and the multi-channel FD-TDSP. **Table 1** shows the computational complexity in each algorithm of CASE (J,L,M). **Figure 2** shows the number of multiplication versus tap length at CASE (1,2,2). **Figure 3** also shows the number of multiplication versus the number of channels at CASE (2,M,M) in case of $N = 128$.

It can be seen from **Fig. 2** that the computational complexity in the FD-TDSP is the smallest when N is more than 32. It can be also seen from **Fig. 3** that the difference of computational complexity between the FD-TDSP and the other MEFX based algorithms becomes larger as the number of channels M increases.

4. EXPERIMENTAL RESULTS

We verify the effectiveness of the proposed multi-channel ANC system. **Figure 4** shows a practical multi-channel ANC system of CASE (1,2,2). Experiment conditions are shown in **Table 2**. The noise source is multi-sinusoidal waves whose frequencies are 100, 130, 180, 240, and 320[Hz], respectively.

Table 2. Experiment conditions.

Sampling frequency	5[kHz]
Tap length of noise control filters	128
Cut-off frequency	1.56[kHz]
DSP	TMS320C6711

We verify the effectiveness of the ANC systems using the multi-channel perturbation method in the environment that the secondary paths always change. Two error microphones are moved like **Fig. 5**. **Figure 5** shows two error microphones moved like the point $\mathrm{A} \rightarrow \mathrm{B} \rightarrow \mathrm{A} \rightarrow \mathrm{C} \rightarrow \mathrm{A} \rightarrow \mathrm{B} \cdots$. The interval of the two error microphones is fixed and the center between the two error microphones is moved to each point. Moreover, the system operates without moving the two error microphones for $20 \sim 30$ seconds at the points A, B and C.

Figure 6 shows error signals of the ANC system to multi-sinusoidal waves obtained at error microphone 1. In **Fig. 6**, (a) is the case that the ANC system is stopped and (b) is the case that the ANC system is executed. It can be seen from **Fig. 6** that the ANC system can operate stably in the environment that secondary paths always change. Therefore, this system has a great advantage that error microphones can be freely moved while the ANC system controls noise.

5. CONCLUSIONS

In this paper, we have presented the multi-channel ANC systems using the SP algorithm. We have demonstrated that

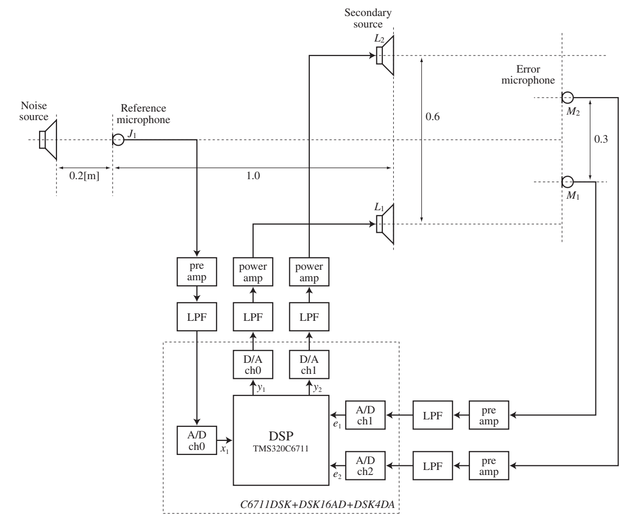


Fig. 4. Practical multi-channel ANC system.

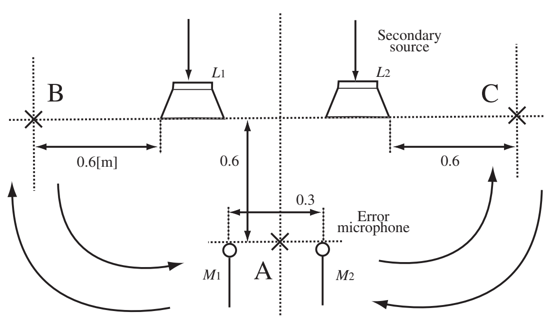


Fig. 5. Position of secondary sources and error microphones.

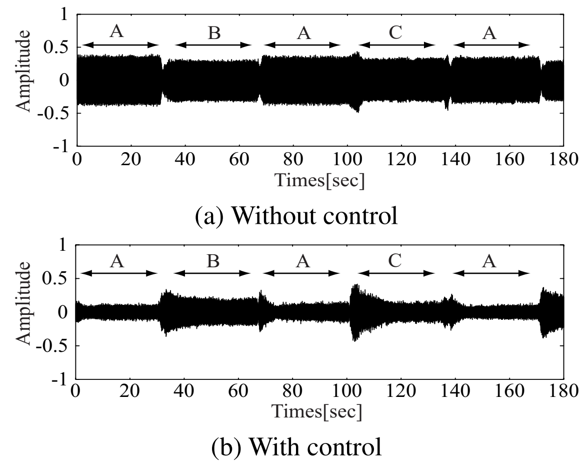


Fig. 6. Error signals of the ANC system to multi–sinusoidal waves.

the proposed ANC system has an advantage which can operate stably even if secondary paths change. In other words, this system has a grate advantage that error microphones can be freely moved while the ANC system controls noise.

6. REFERENCES

[1] P. A. Nelson and S. J. Eliott, *Active Control of Sound*, Academic Press, London, 1992.

[2] S. M. Kuo and D. R. Morgan, *Active Noise Control Systems*, John Willy & Sons, New York, 1996.

[3] C. C. Boucher, S. J. Elliot, and P. A. Nelson, "Effect of errors in the plant model on the performance of algorithms for adaptive feedforward control," *IEE Proc. F. Radar Signal Process.*, vol. 138, no. 8, pp. 313–319, 1991.

[4] S. D. Snyder and C. H. Hansen, "The effect of transfer function estimation errors on the filtered-x lms algorithm," *IEEE Trans. on Signal Process.*, vol. SP-42, no. 4, pp. 950–953, 1994.

[5] K. Fujii, M. Muneyasu, and J. Ohga, "Simultaneous equations method not requiring the secondary path filter," in *Proc. of the 1999 International Symposium on Active Control of Sound and Vibration*, Florida, U.S.A., 1999, pp. 941–948.

[6] Y. Kajikawa and Y. Nomura, "Active noise control system without secondary path model," in *Proc. of the 2000 IEEE Iinternational Symposium on Circuits and Systems*, Geneva, Switzerland, 2000, pp. 349–352.

[7] Y. Kajikawa and Y. Nomura, "Frequency domain active noise control system without a secondary path model via perturbation method," *IEICE Trans. on Fundamentals*, vol. E84-A, no. 12, pp. 3090–3098, 2001.

[8] T. Mori, Y. Kajikawa, and Y. Nomura, "Frequency domain active noise control systems using the time difference simultaneous perturbation method," *IEICE Trans. on Fundamentals*, vol. E86-A, no. 4, pp. 946–949, 2003.

[9] Y. Kajikawa and Y. Nomura, "Active noise control without a secondary path model by using a frequency-domain simultaneous perturbation method with variable perturbation," in *Proc. of the 2003 IEEE International Conference on Acoustic, Speech, and Signal Processing*, Hong Kong, China, 2003, pp. 580–583.