IMPROVEMENT OF CANCELLATION PERFORMANCE FOR ANC SYSTEM USING THE SIMULTANEOUS PERTURBATION METHOD

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ABSTRACT

In this paper, we propose an updating algorithm considering the frequency characteristic of the secondary path in order to improve the cancellation performance of active noise control (ANC) system using the perturbation method. Since the ANC system using the perturbation method does not need the secondary path model, it has an advantage that can track the path changes. However, the conventional perturbation method has a problem that the cancellation performance deteriorates as the frequency of noise becomes high. The proposed method can improve the cancellation performance for the noise at the higher frequency to consider the damping of acoustic wave. The effectiveness of the proposed method is demonstrated through experimental results.

1. INTRODUCTION

In active noise control (ANC) systems [1], the filtered-x LMS (FXLMS) algorithm [2] is widely used as an algorithm to update the coefficients of noise control filters. This algorithm has an advantage such as its small computational complexity and simplicity of implementing to hardware. However, FXLMS algorithm needs an estimation of the secondary path from the secondary source generating anti-noise to the error sensor detecting residuals in advance. Therefore, it has a problem that the ANC system becomes unstable when the secondary path changes and the modeling error increases. In order to solve this problem, we have already proposed a technique called “the perturbation method” [3] which does not need the secondary path models [4]. The perturbation method is an algorithm that calculates an estimate of gradient vector by adding a little perturbation to the coefficients of the noise control filter. This method has an advantage that the computational complexity is small compared with the FXLMS algorithm. Although the convergence speed is quite slow, the frequency domain time difference simultaneous perturbation (FDTDSP) method [5, 6] has been proposed and then the problem of convergence speed has been remarkably improved. However, the FDTDSP method has a problem that the cancellation performance deteriorates in higher frequency band. This arises due to the tendency that a sound wave propagating in a room is more significantly attenuated as its frequency becomes higher. Since the present FDTDSP method does not consider this damping characteristic at each frequency, the updating amount will become small and then the convergence speed will be slow in higher frequency band.

In this paper, we propose an updating algorithm in consideration of the frequency characteristic of secondary path in order to improve the cancellation performance. By using the proposed method, since the exact updating amount is obtained, the cancellation performance of the noise can be improved in higher frequency band.

This research was financially supported by MEXT KAKENH(17686018).

2. FDTDSP METHOD

The FDTDSP method has the fastest convergence speed in the perturbation methods now. The ANC system using the FDTDSP method [5, 6] is shown in Fig. 1. In the perturbation method, a little perturbation is always added to the coefficients of the noise control filter and the coefficients are updated once every N samples. The updating algorithm is defined as follows:

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

where $\mathbf{w}_n$ is the coefficient vector of the noise control filter, $\mathbf{E}_n$ the frequency domain error vector, the superscript * the complex conjugate, n the block time, μ the step-size parameter, c_n the magnitude of the perturbation, $\mathbf{S}_n$ the frequency domain perturbation vector which is a complex vector whose elements are -1 or 1 in both real and imaginary parts and is generated by Maximum Length Sequence. $\mathbf{S}_n$ is generated so that it has the conjugate symmetry and has following characteristics:

$$\begin{array}{l} E[\mathrm{Re}\{S_n(i)\}] = 0,\ E[\mathrm{Im}\{S_n(i)\}] = 0 \\ \mathrm{Re}\{S_n(i)\}^2 = 1,\ \mathrm{Im}\{S_n(i)\}^2 = 1 \\ E[\mathrm{Re}\{S_n(i)\}\mathrm{Re}\{S_m(i)\}] = 0,\ n \neq m \\ E[\mathrm{Im}\{S_n(i)\}\mathrm{Im}\{S_m(i)\}] = 0,\ n \neq m \\ E[\mathrm{Re}\{S_n(i)\}\mathrm{Re}\{S_n(j)\}] = 0,\ i \neq j \\ E[\mathrm{Im}\{S_n(i)\}\mathrm{Im}\{S_n(j)\}] = 0,\ i \neq j \\ E[\mathrm{Re}\{S_n(i)\}\mathrm{Im}\{S_n(j)\}] = 0 \end{array} \tag{2}$$

where $E[\cdot]$ is an expected value. Moreover, the perturbation added to the noise control filter is $c_n\mathbf{s}_n$, and c_n and $\mathbf{s}_n$ are respectively defined as follows:

$$c_n = \sqrt{\frac{\alpha P_{e,n-1}}{G^2 P_{x,n-1}}} \tag{3}$$

$$\mathbf{s}_n = \text{first } N \text{ elements of IFFT}[\mathbf{S}_n] \tag{4}$$

where G is the gain of secondary path, $P_{x,n}$ and $P_{e,n}$ are respectively the power of the input signal and the error signal at the n-th block, and α is a coefficient that defines a ratio of the power of the perturbation to the error signal. Although the FDTDSP method has the fastest convergence speed in the perturbation methods now, it has a problem that the cancellation performance deteriorates in higher frequency band.

3. FDTDSP METHOD CONSIDERING THE FREQUENCY RESPONSE OF SECONDARY PATH

A sound wave propagating in a general room tends to significantly decrease as its frequency becomes high. Primary noise and its antinoise which propagate the primary and secondary paths respectively also have the same tendency. Therefore, the perturbation elements contained in the antinoise is more significantly attenuated as its frequency becomes higher while propagating through a secondary path. Moreover, since the magnitude of the perturbation c_n is computed in the time domain for the FDTDSP method as shown in (2), it will be greatly dependent on the error signal components in lower frequency band where the attenuation is small. On the other hand, although the gradient equation of (1) is computed in the frequency domain, c_n computed in the time domain is used in the equation. Consequently, since the gradient in higher frequency band (the error signal components become small) is also computed using c_n depending on lower frequency band, the updating amount in higher frequency band would become a very small value and the convergence speed would be also slow.

In order to solve this problem, we propose a novel gradient equation defined as follows:

$$\mathbf{U}_n = \mathrm{diag}[\mathbf{S}_n]\mathrm{diag}[\mathbf{M}]^{-1} \cdot \frac{\mathrm{diag}[\mathbf{E}_n^*]\mathbf{E}_n - \mathrm{diag}[\mathbf{E}_{n-1}^*]\mathbf{E}_{n-1}}{c_n} \tag{5}$$

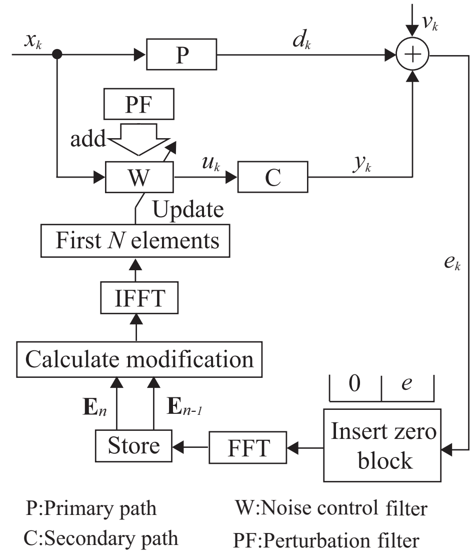


Fig. 1. Block diagram of the ANC system using FDTDSP method.

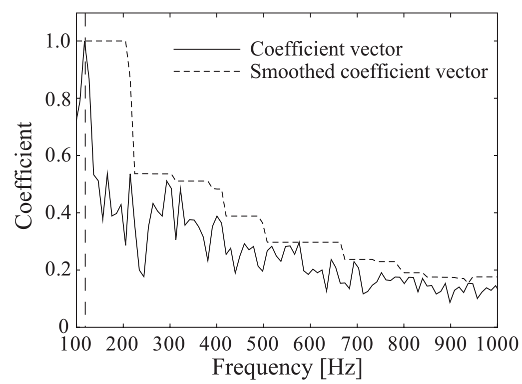


Fig. 2. Coefficient vector $\mathbf{M}$.

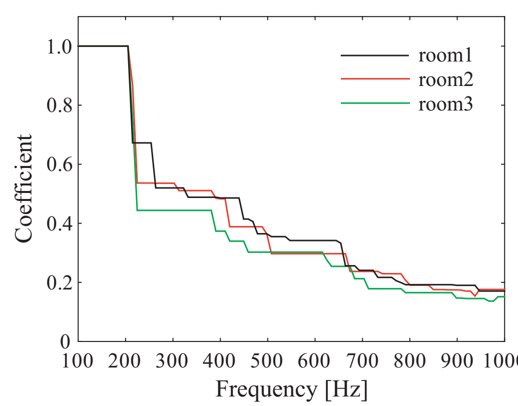


Fig. 3. Coefficient vector $\mathbf{M}$ in the case where the sizes of a room measuring the secondary path are changed. (the distance of the secondary path is 40cm)

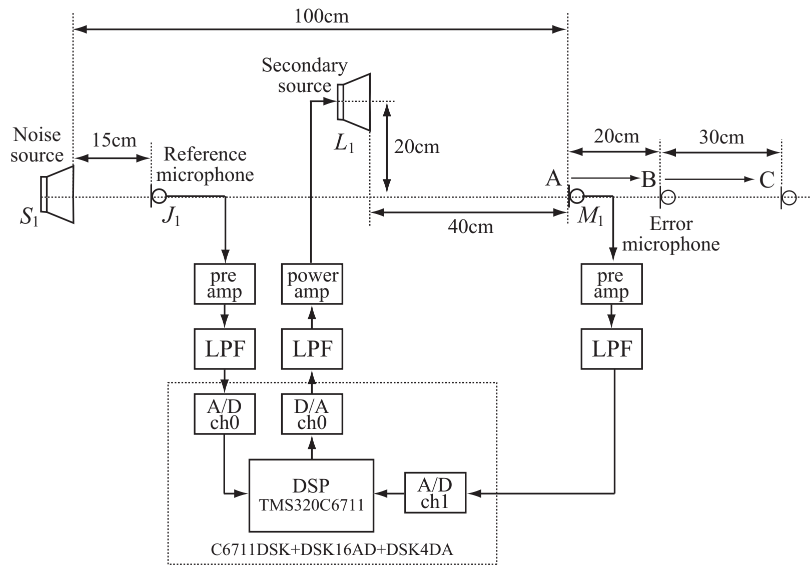


Fig. 4. Measurement model of the ANC system.

where $\mathbf{M}$ is a coefficient vector which represents the ratio of the amplitude spectrum at each frequency to the peak value of secondary path as shown in Figure. 2, and is smoothed to suppress the influence of the dip. This gradient equation is computed at each frequency in consideration of the attenuation of the perturbation components after propagating the secondary path by using the coefficient vector $\mathbf{M}$, and the cancellation performance of the noise at high frequencies can be consequently improved.

The coefficient vector $\mathbf{M}$ shown in Fig. 2 was computed using the frequency characteristic of the secondary path measured when the distance between the secondary source and the error microphone was set to 40cm in a room with reverberation. Although this characteristic changes with environment, if the distance of a secondary path is short, the variation is small as shown in Figure 3 and the characteristic computed in advance can be used. However, since the influence of reflective becomes large when the distance of a secondary path is long, the characteristic tends to change with environment. Therefore, it is desirable that the frequency characteristic of the secondary path in the environment is measured once, or step size is made small, in order to make the proposed method operate stably. Even if step size is made small for stability, the proposed method can converge faster than the conventional one.

4. EXPRIMENTAL RESULTS

4.1. Experiment Conditions

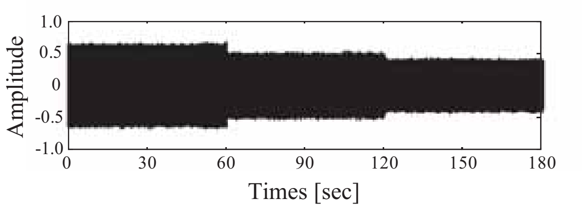
We demonstrate that the ANC system using the proposed method can effectively control noise in the environment with reverberation. Figure 4 shows a measurement model of ANC system used in the experiment. DSP used in this experiment is TMS320C6711 (Texas Instruments Co. product). In the experiment, the error microphone M_1 was first arranged at A point, moved from A point to B point after ANC system startup, and finally moved from B point to C point, in order to verify the effectiveness for the variation of the secondary path. Table 1 shows experiment conditions. Noise source is multi-sinusoidal wave (100, 300, 500, 700, 900[Hz]). Moreover, the proposed and the conventional method were set up with α=0.08 and μ=0.015 so that the convergence becomes fast and stable, and the noise reduction after convergence becomes almost equivalent. Furthermore, the coefficient vector M used in the proposed method was computed using the frequency characteristic of the secondary path measured at A point in Fig. 4.

Table 1. Measurement conditions.

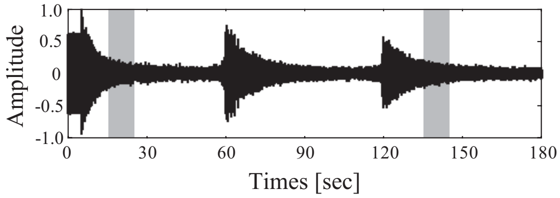
Tap length of noise control filter	128
Cut-off frequency of low-pass filter	1562.5[Hz]
Sampling frequency	5000[Hz]
Temperature	25°C

4.2. Cancellation Performance

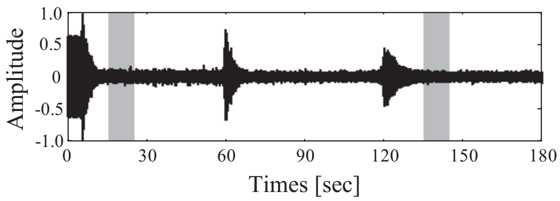
Figure 5 shows error signals of the ANC system at the error microphone M_1. In Fig. 5, (a), (b), and (c) are the cases before control, using the conventional and proposed methods, respectively. In (b) and (c), ANC was started in 5 seconds. In addition, the error microphone M_1 was moved from A point to B point and B point to C point in Fig. 4 in 60 and 120 seconds, respectively. Figure 6 shows error spectra. In Fig. 6 ,(a) and (b) are from 15 to 25 seconds after ANC system startup



(a)ANC system OFF

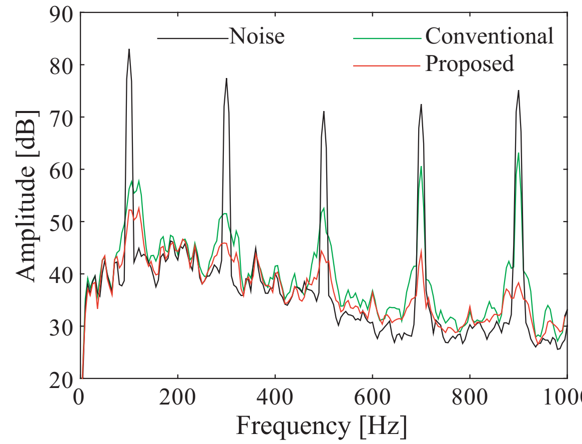


(b)Conventional

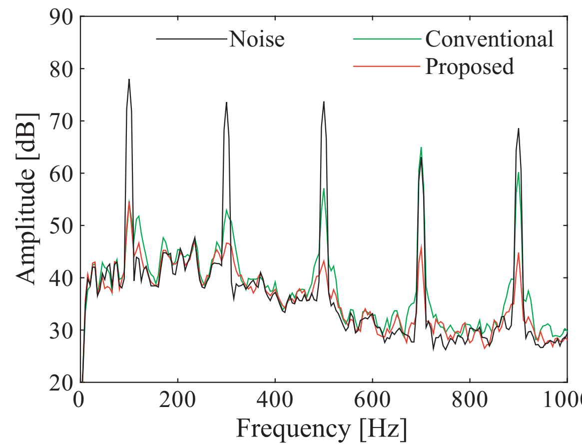


(c)Proposed

Fig. 5. Error signals.



(a)15∼25sec



(b)135∼145sec

Fig. 6. Error spectra.

and from 135 to 145 seconds after path change, respectively.

It can be seen from Fig. 5 that the ANC system using the proposed method can converge faster than that using the conventional method. In spite of using the coefficient vector M at A point, the convergence speed can be also improved greatly after the path change. It can be also seen from Fig. 6 that the proposed method is very effective at high frequencies compared with the conventional method.

5. CONCLUSIONS

In this paper, we have proposed the updating algorithm considering the frequency characteristic of the secondary path, and verified the cancellation performance. As a result, both the convergence speed and the effectiveness of noise reduction in higher frequency band have been improved.

6. REFERENCES

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