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Improvement of Active Noise Control Performance by Changing Reference Signal

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ABSTRACT

The LMS (Least-Mean-Square) algorithm, which can obtain the complex transfer function in real-time, allows the changes of noise characteristics and the environment of system, thereby maintaining the convergence performance. Therefore, many algorithms based on the LMS algorithm have been modified to solve for some practical considerations, and the FXLMS (Filtered-X LMS) algorithm has been widely investigated and applied as a feed-forward control scheme for an active control system. In 3-dimensional space system, measurement of the signal (reference and error) is more important than control algorithm. However, almost studies were focused on ANC (Active Noise Control) algorithm. In this paper, FXLMS algorithm is applied to the experiment on the ANC of the 3-dimensional enclosure system. And evaluate the ANC performance by changing the location of reference signal.

1. INTRODUCTION

The LMS (Least-Mean-Square) algorithm, which can obtain the complex transfer function in real-time, allows the changes of noise characteristics and the environment of system, thereby maintaining the convergence performance. Therefore, many algorithms based on the LMS algorithm have been modified to solve for some practical considerations, and the FXLMS (Filtered-X LMS) algorithm has been widely investigated and applied as a feed-forward control scheme for an active control system.

Meanwhile, established study of active noise control was limited 1D plane wave like duct system. However, passenger car, elevator etc. are 3D system. And case study is not unusual. Especially, elevator is gradually high rise and high speed. Therefore interior noise cause fear and uncomfortable. The interior noise of elevator is flow induced noise and wire noise. And it has low frequency range(below 1000Hz). The noise of 500~1000Hz range can control using passive noise reduction method but the effect of noise reduction is poor at low frequencies (below 500Hz). Therefore, Active noise control method is necessary.

However, as mentioned before, it is necessary to study application of active noise control method in that elevator is not guaranteed 1D plane wave and its characteristic of enclosure system. And it is necessary that active noise control is applied to 3D enclosure system (3D prototype) and evaluate its performance before applied to elevator system.

2. THEORY

2.1 FXLMS Algorthm

A block diagram of the FXLMS algorithm is shown in Figure 1.

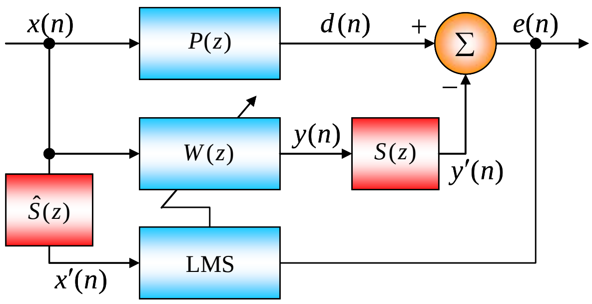


Figure 1. Block diagram of FXLMS algorithm

In the real system, it is necessary to compensate for the secondary-path transfer function $S(z)$ from $y(n)$ to $e(n)$, which includes the D/A converter, low-pass filter, power amplifier, loud speaker, acoustic path from the loud speaker to the error microphone, error microphone, power supply, low-pass filter, and A/D converter. The transfer function $S(z)$ is modeled by an off-line method, and its estimated value is written as $\hat{S}(z)$. In Figure. 1, the active noise control system using the FXLMS algorithm require a reference signal $x(n)$ for generating the control signal $y(n)$. To drive the control signal, the reference signal $x(n)$ has to pass through the adaptive filter $W(z)$ in order to minimize the error sensor signal $e(n)$.

Also, the reference signal $x(n)$ has to pass through the modeled transfer function of the secondary path $\hat{S}(z)$ in order to update the adaptive filter $W(z)$. Where the predicted value of the secondary path transfer function $\hat{S}(z)$ and the adaptive filter $W(z)$ are implemented for a finite impulse response. The error signal at time n is represented as follows:

$$\begin{aligned} e(n) &= d(n) - y'(n) \\ &= d(n) - s(n) * y(n) \\ &= d(n) - s(n) * \left[\mathbf{w}^T(n) * \mathbf{x}(n)\right] \end{aligned} \tag{1}$$

$s(n)$ is the impulse response of the secondary path transfer function $S(n)$ at time n, and $*$ is the convolution. $y(n)$ is generated after the reference signal $x(n)$ has passed through the adaptive filter $W(z)$. L is the filter order, Eq. (2) is the input vector of the reference signal $x(n)$ at time n, and Eq. (3) is the weight vector of the adaptive filter $W(z)$ at time n.

$$\mathbf{x}(n) = \left[x(n)\ \ x(n-1)\ \ x(n-2)\ \cdots\ x(n-L+1)\right] \tag{2}$$

$$\mathbf{w}(n) = \left[w_0(n)\ \ w_1(n)\ \ w_2(n)\ \cdots\ w_{L-1}(n)\right] \tag{3}$$

Our objective is to minimize the instantaneous square error $\hat{\xi}(n) = e^2(n)$. To satisfy this objective, steepest descent was used in linear programming and optimization problems to find a solution that minimizes an objective function. The concept of steepest descent can be implemented in the following algorithm.

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \frac{\mu}{2}\nabla\hat{\xi}(n) \tag{4}$$

Where $\nabla\hat{\xi}(n)$ is an instantaneous estimate value of the mean square error at time n and can be expressed as:

$$\begin{aligned} \nabla\hat{\xi}(n) &= \nabla e^2(n) = 2\left[\nabla e(n)\right]e(n) \\ &= 2\left[-s(n) * \mathbf{x}(n)\right]e(n) = -2\mathbf{x}'(n)e(n) \end{aligned} \tag{5}$$

Substituting Eq. (5) into Eq. (4), we have the FXLMS algorithm.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\,\mathbf{x}'(n)e(n) \tag{6}$$

In Eq. (6), μ is the step size that must satisfy Eq. (7).

$$0 < \mu < \frac{2}{LP_x} \tag{7}$$

Where P_x is the power of the reference signal.

2.2 Co-FXLMS Algorithm

As shown in Eq. (6), the stability, convergence time, and fluctuation of the FXLMS algorithm is governed by the step size μ and the filtered input signal $x'(n)$. In Eq. (7), the FXLMS algorithm uses the constant μ, and the upper bound on μ is made inversely proportional to the input signal power. Thus, weaker signals can use a larger μ, and stronger signals have to use a smaller μ. Therefore, when the input signal power changes excessively according to time, the FXLMS algorithm cannot converge. One useful approach is to normalize μ with respect to the power of the filtered input signal $x'(n)$.

$$\mu(n) = \frac{\alpha}{L\hat{P}_x'}, \qquad (0 < \alpha < 2) \tag{8}$$

In equation (8), α is the normalized step size and $\hat{P}_x'$ is the running estimate of the average power of the filtered input signal $x'(n)$. The simplest way to estimate the average power of the filtered input $\hat{P}_x'$ is to use a running-average filter. The following is an M^{th}-order running-average filter with the filtered input $x'(n)$.

$$\hat{P}_x'(n) = \frac{1}{M}\sum_{i=0}^{M-1} x'^2(n-i) \tag{9}$$

If the order of the running-average filter in Eq. (9) is equal to the number of adaptive filter coefficients L, then

$$\hat{P}_x'(n) = \frac{\mathbf{x}'^T(n)\mathbf{x}'(n)}{L} \tag{10}$$

Substituting Eq. (10) into Eq. (8), we obtain

$$\mu(n) = \frac{\alpha}{\mathbf{x}'^T(n)\mathbf{x}'(n)} \tag{11}$$

There is one additional practical difficulty that can arise when a nonstationary input is used. If the input $x(n)$ is zero for L consecutive samples, then $\mathbf{x}'(n) = 0$ and the step size in Eq. (11) becomes unbounded. This problem also occurs when the algorithm starts at $\mathbf{x}'(0) = 0$. To avoid this numerical difficulty, we let δ be a small positive value.

$$\mu(n) = \frac{\alpha}{\delta + \mathbf{x}'^T(n)\mathbf{x}'(n)} \tag{12}$$

The Co-FXLMS algorithm is realized using Eq. (12), an estimate of the correlation between the filtered input signal $x'(n)$, and the error signal $e(n)$ to adjust the step size of the adaptive algorithm. After $\mathbf{w}(n)$ has converged to the optimum weight $\mathbf{w}^o$, the correlation between the filtered input signal $x'(n)$ and the error signal $e(n)$ would be zero.

$$R(n) = E\left[e(n)x'(n)\right] = 0 \tag{13}$$

In Eq. (13), the estimated correlation $R(n)$ is an expected value of $e(n)x'(n)$, and $R(n)$ at time n is represented as follows.

$$\begin{aligned} R(n) &= E\left[e(n)x'(n)\right] \\ &= E\left[\left(d(n) - y'(n)\right)x'(n)\right] \\ &= E\left[\left(d(n) - \mathbf{w}^T(n)\mathbf{x}'(n)\right)x'(n)\right] \\ &= E\left[\left\{d(n) - \mathbf{w}^T(n)\left(\hat{s}(n) * \mathbf{x}(n)\right)\right\}x'(n)\right] \end{aligned} \tag{14}$$

Where $\hat{s}(n)$ is the estimated impulse response of the secondary-path filter, $\hat{S}(z)$. When the current weight vector $\mathbf{w}(n)$ is far away from $\mathbf{w}^o$, the correlation is high and the adaptive algorithm is in an active state with a relatively large step size. After convergence of the adaptive filter, the filtered input $x'(n)$ and the error signal $e(n)$ are nearly uncorrelated, and the step size is set too close to zero, thus putting the adaptation algorithm into a sleep state for stability. Therefore, it is logical to let the step size $\mu(n)$ in Eq. (12) be proportional to the estimated correlation $R(n)$. Then, the step size $\mu(n)$ in Eq. (12) is represented as follows.

$$\mu(n) = \frac{C}{\delta + \mathbf{x}'^T(n)\mathbf{x}'(n)}R(n) \tag{15}$$

$$R(n) = \lambda R(n-1) + (1-\lambda)x'(n)e(n) \tag{16}$$

Where C is a weight factor and λ is a smoothing factor with $0 < \lambda < 1$. Substituting Eq. (15) into Eq. (6), we obtain the Co-FXLMS algorithm.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{CR(n)}{\delta + \mathbf{x}'^T(n)\mathbf{x}'(n)}\mathbf{x}'(n)e(n) \tag{17}$$

3. ACTIVE NOISE CONTROL OF 3D ENCLOSURE SYSTEM

3.1 Experimental Setup

The block diagram of the TMS320C32-based active noise control system is shown in Figure. 2.

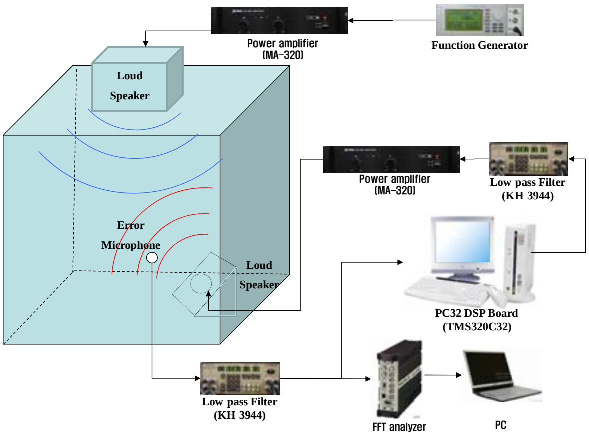


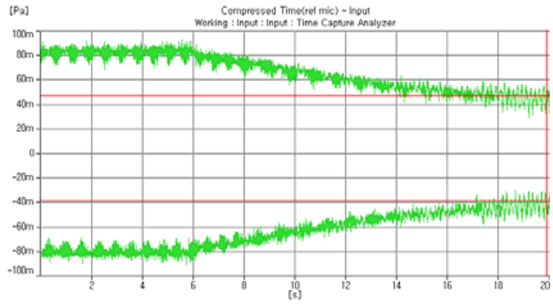
Figure 2. Experimental setup of 3D active noise control syetem

The active noise control system is designed to accept two input signals, one from the input microphone (B&K 4130) and one from the error microphone (B&K 4130). The output signal is sent to a band-pass filter (KH-3944) and converted to an analog form to drive a control speaker using a power amplifier (Inkel MA-320). The reference signal and error signal are detected by the FFT analyzer (Pulse 3560-B-040) for spectral analysis and displayed by the monitoring PC (HP notebook). The PC32 DSP board is comprised of an A/D converter and D/A converter, each with four channels. The C language ANC program (Co-FXLMS algorithm) consists of the main program, linker command file, and batch file. Only these three files are needed to build the COFF file, which is an executable file. To execute the ANC program, a COFF file has to be downloaded to the PC32 DSP board.

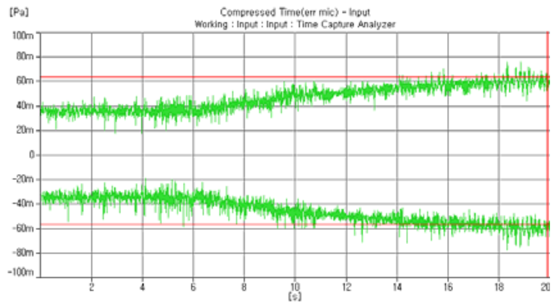
We select the location of error microphone and speaker based on result of acoustic field analysis. In the past study filter is usually used Low Pass Filter (0~500Hz). However, in this study we use band pass filter (target frequency range) for improvement of active control performance. The band pass filter performed to decrease local minimum of steepest decent method.

3.2 Experiment Result

After the application of 3-D active noise control, the noise measured from the error microphone was reduced at the beginning of control. But controlled noise again increased afterwards. The results of active noise control are shown below.



(a) Reference signal



(b) Error signal

Figure 3. The active noise control result of 3D enclosure system (Time domin/internal reference signal)

This happens when applying active noise control to 3-D enclosure system. There is insufficient information of noise to be used as a reference signal because the reference signal is dwindling away as a result of active noise control,

4. ACTIVE NOISE CONTROL OF MODIFIED 3D ENCLOSURE SYSTEM

4.1 Experimental Setup

Since the reference microphone is positioned inside the space where control speaker and error microphone are also placed, the reference signal affected by the control signal begin to lessen in the application of active noise control to 3-D enclosure system. Therefore, control of noise becomes unstable and noise shoots up later in controlling.

The system of active noise control is modified in order to improve the performance of the control, as described in Figure 4.

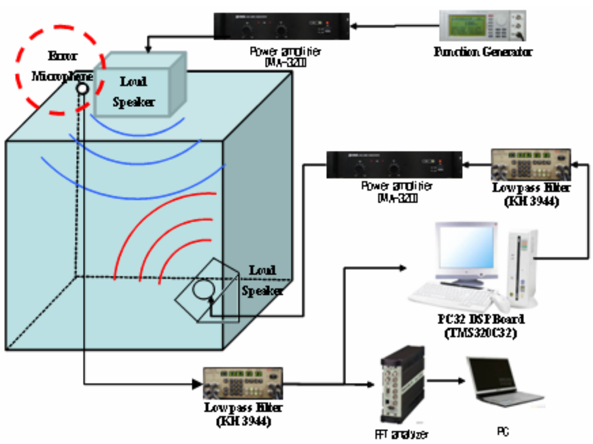
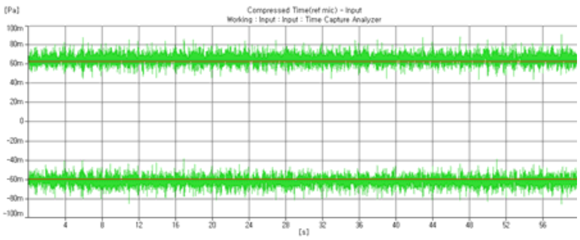


Figure 4. Experimental setup of modified 3D active noise control system (outer reference signal)

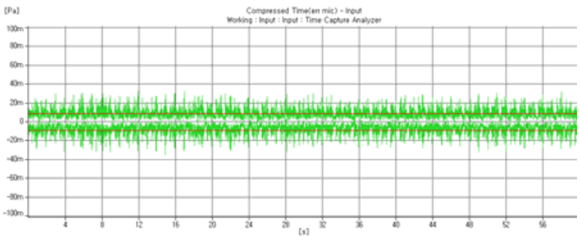
In the redesigned model of active noise control for 3-D enclosure system, a reference microphone is positioned at the outside of the system, so that it can receive a reference signal without affecting the stability of control. The performance of control is also ensured by confirming high correlation of the reference signal with the noise signal inside the system.

4.2 Experiment Result

The results of active noise control using the reference microphone placed outside are shown below.



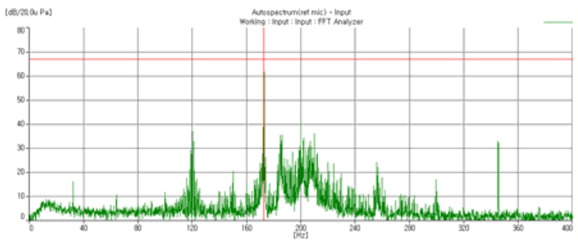
(a) Reference signal



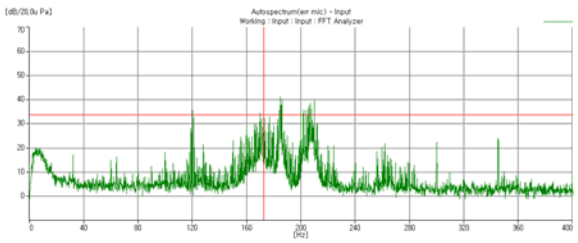
(b) Error signal

Figure 5. The active noise control result of 3D enclosure system (Time domin/outer reference signal)

It is confirmed that the reference signal obtained remains unaffected by active noise control inside the 3-D enclosure system. For that reason, the performance of control is improved with consistency. The results are given below.



(a) Before control



(b) After control

Figure 6. The active noise control result of 3D enclosure system (Frequency domin/outer reference signal)

In frequency domain, it is verified that the target frequency for control, 172Hz, is being gained as a reference signal shown below.

Table 1 The active noise control result of 3D enclosure system (Frequency domin/outer reference signal)

	Before Control (dB)	**After Control (dB)**
172Hz	**67.7**	**33.3**

5. CONCLUSION

Active noise control for 3-D enclosure system is carried out and a reference signal is obtained at the outside of the system to ensure the stability of control. Therefore, 34.4(dB) reduction of noise at the target frequency 172Hz is made from 67.7(dB) to 33.3(dB) after the application of active noise control. The consistent stability of control is also ensured later in controlling.

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