A method stably working feedback type active noise control system for preventive panel of sound leakage

Kensaku FUJII[1]; Mitsuji MUNEYASU[2]

[1] Kodaway Laboratory, Japan

[2] Kansai University, Japan

ABSTRACT

In this paper, we propose a method capable of providing the maximum noise reduction effect without adjusting the step size. The feedback type active noise control system estimates the coefficients of the noise control filter by applying the filtered-x algorithm to the steepest descent method, for example, the least mean square (LMS) algorithm. The steepest decent method, however, stipulates the upper limit of the step size guaranteeing the stable estimation according to the auto-correlation function of the reference noise. Our target is the reduction of non-stationary noise, such as speech signal and traffic noise, whose auto-correlation function is continuously changing. The filtered-x algorithm moreover processes the reference noise with the feedback control filter, which changes the auto-correlation function according to the impulse response of the feedback path. We then propose to apply the sub-recursive least square (sub-RLS) algorithm not requiring the inverse matrix operation to the estimation of the coefficients of the noise control filter. The Sub-RLS algorithm can estimate the coefficients independently of the auto-correlation function, and provides almost the same convergence speed as that of RLS algorithm. We finally verify using computer simulation that the sub-RLS algorithm successfully reduces the reference noise with a fixed step size.

Keywords: Sub-RLS algorithm, Stability condition, Step size, Linear prediction, Feedback path: 38.2

1. INTRODUCTION

The feedback type active noise control system (1) can be effectively applied to the preventive panel of sound leakage. One of typical examples is the road traffic noise barrier (2), whose target nose is radiated by vehicles moving rapidly. The characteristics of the target noise are therefore assumed to fluctuate according to the traffic volume and speed. The other example is the panel for partitioning meeting booths, whose sound to be reduced is mainly voice. The target noise should be supposed to be non-stationary in such cases.

The feedback type active noise control system is characterized by using only one microphone for both the detection of the target noise and the monitor of the reduction error. Practically, the target noise and the secondary noise simultaneously arrive at the microphone, which outputs the sum of them as the reduction error. The target noise is therefore required deriving by processing the reduction error.

The secondary noise is generated by the noise control filter, which is radiated from a loudspeaker and then arrives at the microphone through the feedback path. In practical use, the feedback path from the loudspeaker to the microphone is previously estimated by working the feedback control filter as an adaptive filter (3). The feedback control filter is connected in parallel to the feedback path so as to cancel the feedback path. The target noise can be thereby calculated as the difference between the two outputs of the microphone and the feedback control filter.

The system applies the calculated target noise to the noise control filter, and estimates the coefficients of the noise control filter so as to minimize the reduction error detected by the microphone. The target noise is consequently reduced. The estimation, however, generally requires the application of the filtered-x algorithm. The calculated target noise is then applied to the other feedback control filter with the same coefficients as those of the filter used for the cancelation of the

[1] kodawayfujii@gmail.com

[2] muneyasu@kansai-u.ac.jp

feedback path. The coefficients of the noise control filter are estimated by using the output of the other feedback control filter as the reference signal.

In practical systems, the adaptive algorithm used for the estimation is the steepest descent method, for example, the least mean square (LMS) algorithm. The reference signal provided by applying the calculated target noise to the other feedback control filter, which characterizes the filtered-x algorithm, causes the problem that the auto-correlation function of the reference signal is changed by the characteristics of the feedback control filter modeled on the feedback path. Actually, the condition for guaranteeing the stable estimation, concretely the range of the step size applied to the steepest descent method, depends on the auto-correlation function of the reference signal.

In this paper, we focus on the matter that the application of the steepest decent method requires the adjustment of the step size previous to controlling the target noise. In practical use, the characteristics of the feedback path are unknown until the system is installed. This means that the manual adjustment of the step size is required previous to controlling the target noise. The system is expected to work without any previous adjustment. We have thus presented a method for automatically estimating the coefficients of the feedback control filter (4). It is desirable that the previous adjustment of the step size also is unnecessary.

The least square (LS) type algorithm can stably estimate the coefficients of the noise control filter independent of the auto-correlation function of the reference signal. The LS algorithm, however, requires the operation of the inverse matrix accompanied with high pressing cost. We propose to apply the sub-recursive least square (sub-RLS) algorithm (5, 6) to the estimation. The calculation cost of the sub-RLS algorithm is lower than that of the recursive least square (RLS) algorithm, and moreover the convergence speed is almost the same.

In this study, the target noise is supposed to be non-stationary; accordingly the forgetting factor corresponding to the step size is similarly applied to the sub-RLS algorithm. We first show using the speech signal recorded in (7) that the steepest descent method, the normalized least mean square (NLMS) algorithm suitable for the estimation system using the speech signal as the reference signal, requires adjusting the step size accordingly to the characteristics of the feedback path for guaranteeing the stable performance of the system. We next demonstrate that the sub-RLS algorithm can stably provide the maximum noise reduction effect with the step size of from 0.4 to 0.5 independent of the impulse response.

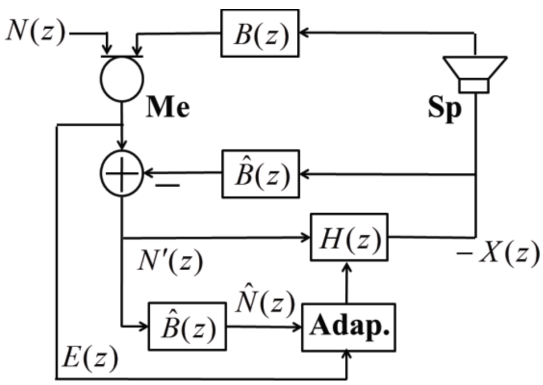

Figure 1 – configuration of feedback type active noise control system

2. CONFIGURATION OF FEEDBACK TYPE ACTIVE NOISE CONTROL SYSTEM

Figure 1 shows the configuration of the feedback type active noise control system reducing the target noise $N(z)$ arriving at the microphone Me, by adding the secondary noise $-X(z)$ radiated from the loudspeaker Sp. In this figure, the signals, the acoustic path and the filters are denoted by z-transform and transfer function, respectively. In addition, “Adap.” designates the estimation circuit of the coefficients of noise control filter $H(z)$ by applying an adaptive algorithm, for example, the least mean square (LMS) algorithm.

In this configuration, the system generally identifies the feedback path $B(z)$ from the loudspeaker Sp to the microphone Me, by working the feedback control filter $\hat{B}(z)$ as an adaptive filter previous to controlling the target noise $N(z)$. After the identification, the feedback control filter approximately corresponds to the feedback path. The feedback path is consequently cancelled by the feedback control filter parallel connected as shown in Fig. 1. The target noise $N(z)$ is thus derived as $N'(z)$, which is used as the reference signal necessary for estimating the coefficients of the noise control filter.

By using the derived $N'(z)$, the noise control filter generates the secondary noise $-X(z)$, which is radiated from the loudspeaker and then arrives at the microphone as $-X(z)B(z)$. The primary noise $N(z)$ is reduced by the addition of the secondary noise; consequently the reduction error,

$$E(z) = N(z) - X(z)B(z), \tag{1}$$

is detected by the microphone. The coefficients of the noise control filter $H(z)$ are updated using the adaptive algorithm so that the reduction error $E(z)$ is minimized. The target noise $N(z)$ is thus reduced.

In this study, we first explain that the feedback type active noise control system works as a linear prediction system. Figure 2 shows the configuration of the system simplified on the assumption that the feedback path is completely canceled by the feedback control filter. Here, we note that the feedback path $B(z)$ and the noise control filter $H(z)$ can be regarded as linear system. Then, we can reverse the order of them as shown in Fig.3.

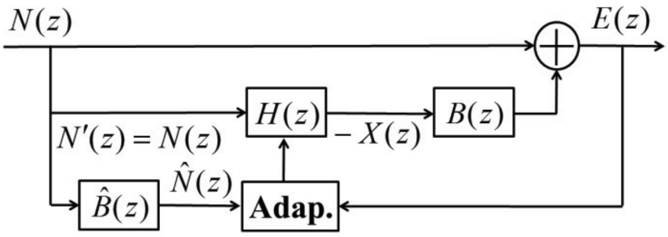

Figure 2 – Configuration simplified on the assumption that the feedback path is completely cancelled

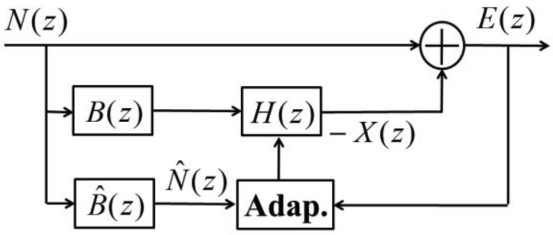

Figure 3 – Configuration reversed by regarding feedback path and noise control filter as linear system

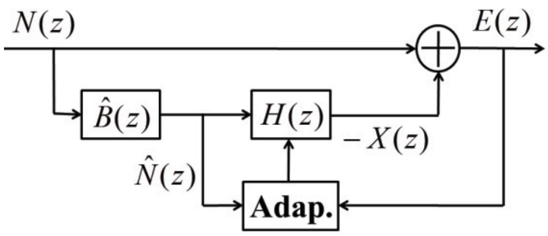

Figure 4 – Fundamental configuration of feedback type active noise control system shown in Fig.1

The reversed configuration can be moreover rearranged to the expression shown in Fig. 4, since the output of the feedback path corresponds to that of the feedback control filter on the same assumption. The rearranged expression clearly states that the feedback type active noise control system works as a linear prediction system to which the principle of the filtered-x algorithm is applied.

From the configuration shown in Fig.4, we can see that the reference signal applied to the noise control filter is converted from the target noise $N(z)\,[\approx N'(z)]$ to $\hat{N}(z)$. This conversion indicates that the auto-correlation function of the reference signal $\hat{N}(z)$ used for estimating the coefficients of the noise control filter is changed by the characteristics of the feedback path $B(z)$.

The auto-correlation function of the reference signal determines the range of the step size in the case of applying the steepest descent method, for example, the LMS algorithm to the estimation. Then, we require adjusting the step size previous to controlling the target noise to work the system stably. In practical use, the auto-correlation function cannot be settled until the feedback path is estimated even if the characteristics of the target noise is known.

Our purpose is the reduction of the non-stationary noise. A small step size then deteriorates the performance of the linear prediction. A large step size oppositely makes the error of the linear prediction diverge. The adjustment of the step size is essential to providing the high and stable performance to the feedback type active noise control system.

3. APPLICATION OF SUB-RLS ALGORITHM

A method capable of preventing the divergence is to apply the recursive least square (RLS) algorithm to the estimation of the coefficients of the noise control filter. The RLS algorithm, however, requires the complex computation, which increases the amount of calculation. In this paper, we propose to apply the sub-RLS algorithm (5, 6) to the estimation of the coefficients of the noise control filter. The sub-RLS algorithm is classified to one of variations of the least square (LS) algorithm, however, does not require the inverse matrix operation. Moreover, the convergence property of the sub-RLS algorithm is almost the same as that of the RLS algorithm (6).

The sub-RLS algorithm is expressed by

$$\boldsymbol{H}_{j+1} = \hat{\boldsymbol{R}}_j^{-1}\left[\boldsymbol{P}_j - \boldsymbol{S}_j\boldsymbol{H}_j\right], \tag{2}$$

where

$$\boldsymbol{H}_j = \left[H_j(0) \quad H_j(1) \quad \cdots \quad H_j(I-1)\right]^T, \tag{3}$$

is the coefficient vector given to the noise control filter working as an adaptive filter at j sample time index,

$$\hat{\boldsymbol{R}}_j = \left[R_j(k,k)\right] \tag{4}$$

and

$$\boldsymbol{S}_j = \left[R_j(k,m)\right] \qquad (k \neq m) \tag{5}$$

are the diagonal and the non-diagonal matrix of the auto-correlation matrix of the reference signal $\hat{N}(z)$ provided by the other feedback control filter introduced on the principle of the filtered-x algorithm,

$$\boldsymbol{R}_j = \begin{bmatrix} R_j(0,0) & R_j(0,1) & \cdots & R_j(0,I-1) \\ R_j(1,0) & R_j(1,1) & \cdots & R_j(1,I-1) \\ \vdots & \vdots & \ddots & \vdots \\ R_j(I-1,0) & R_j(I-1,1) & \cdots & R_j(i-1,I-1) \end{bmatrix}, \tag{6}$$

respectively,

$$k, m = 0,1,2,\cdots,(I-1), \tag{7}$$

and

$$\boldsymbol{P}_j = \begin{bmatrix} P_j(0) & P_j(1) & \cdots & P_j(I-1) \end{bmatrix}^T, \tag{8}$$

is the cross-correlation vector between the derived target noise $N'(z)$ and the reference signal $\hat{N}(z)$. It should be noted here that $\hat{\boldsymbol{R}}_j$ is the diagonal matrix. The inverse matrix $\hat{\boldsymbol{R}}_j^{-1}$ can be therefore expressed by the simple form,

$$\hat{\boldsymbol{R}}_j^{-1} = \begin{bmatrix} 1/R_j(0,0) & 0 & \cdots & 0 \\ 0 & 1/R_j(1,1) & \ddots & \vdots \\ \vdots & \ddots & \ddots & 0 \\ 0 & \cdots & 0 & 1/R_j(I-1,I-1) \end{bmatrix}. \tag{9}$$

The sub-RLS algorithm moreover approximates the LS algorithm by repeatedly operating Eq. (2) as mentioned below:

$$\boldsymbol{H}_j^{(n+1)} = \hat{\boldsymbol{R}}_j^{-1}\left[\boldsymbol{P}_j - \boldsymbol{S}_j \boldsymbol{H}_j^{(n)}\right] \tag{10}$$

where

$$n = 0,1,2,\cdots,N,(N+1),\cdots,\infty, \tag{11}$$

$\boldsymbol{H}_j^{(0)}$ is equal to $\boldsymbol{H}_j$ given in Eq. (2), and $\boldsymbol{H}_j^{(\infty)}$ corresponds to the coefficient vector calculated by the LS algorithm:

$$\boldsymbol{H}_{j+1} = \boldsymbol{R}_j^{-1}\boldsymbol{P}_j. \tag{12}$$

On the other hand, our purpose is the reduction of non-stationary noise, which accordingly requires the application of the forgetting factor to the calculation of the elements of the auto-correlation matrix $\boldsymbol{R}_j$ and the cross-correlation vector $\boldsymbol{P}_j$. In this paper, we define the forgetting factor as follows:

$$\rho = 1 - \mu/I, \tag{13}$$

where μ is the step size applied to the normalized least mean square (NLMS) algorithm and I is the number of taps of the noise control filter. The elements of the auto-correlation matrix and the cross-correlation vector are calculated using the forgetting factor as shown below:

$$R_j(k,m) = \rho R_{j-1}(k,m) + \hat{N}_j(k)\hat{N}_j(m) \tag{14}$$

and

$$P_j(k) = \rho P_{j-1}(k) + N_j'(k)N_j(m), \tag{15}$$

respectively.

4. VERIFICATION BY SIMULATION

The typical target noise reduced by the preventive panel is speech signal. The power of the speech signal, however, largely fluctuates. In this simulation, we substitute the NLMS algorithm for the LMS algorithm, to prevent the power fluctuation from disturbing the derivation of the range of the step size guaranteeing the stable performance to the system. We verify using the speech signal (4) shown in Fig. 3 that the range of the step size stably reducing the target noise changes in the case of using the NLMS algorithm and oppositely the sub-RLS algorithm determines the range independent of the characteristics of the feedback path.

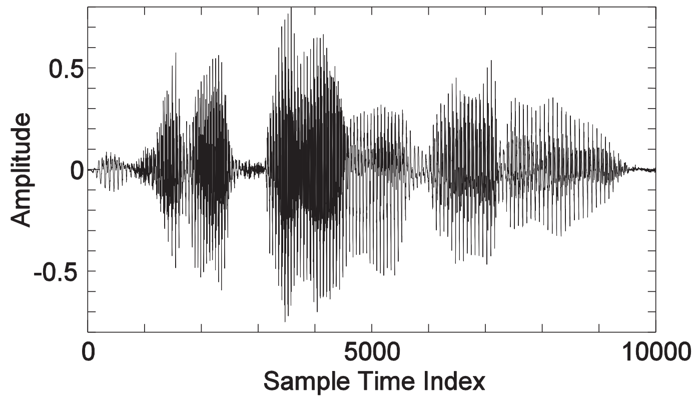

Figure 3 – Speech signal used for simulations

Table 1 – Equipment used for measuring feedback paths

Equipment	Product
Loudspeaker amplifier	CP600 (Catsystem)
Loudspeaker	ASW-2729D (Amon Ind.)
Loudspeaker	TD307 (FUJITSU TEN)
Microphone amplifier	AT-MA2 (Audio Technica)
Microphone	AT-805F (Audio Technica)
USB audio interface	UA-1G (Rorand)
Personal computer	PCG-6R1N (SONY)
Audio capture	FA-66(Rorand)

4.1 Simulation condition

In this study, we measured the impulse responses of the feedback paths, using the equipment shown in Table 1, where the distance from the loudspeaker to the microphone is 10 cm, and the sampling frequency is 8 kHz.

Figures 4 and 5 show the impulse responses measured by using the equipment listed in Table 1, whose power gains are 9.06 and 10.42, respectively. Moreover, Figure 6 shows the arithmetic mean of the impulse responses shown in Figs. 4 and 5, whose power gain is 4.19. The range of the step size stably working the linear prediction is calculated using the impulse responses, and the other conditions are as follows:

(i) Environmental noise: white noise
(ii) Power ratio of speech signal to environmental noise: 30 dB
(iii) Number of taps of noise control filter: 64
(iv) Adaptive algorithms: NLMS and Sub-RLS

We moreover assume that the feedback paths are completely estimated in order to remove the influence of the estimation error.

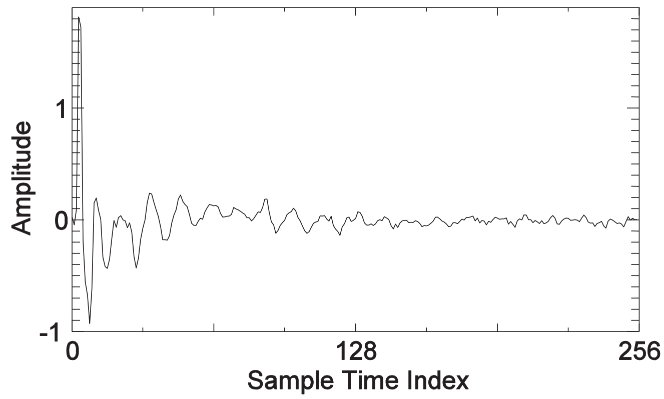

Figure 4 – Impulse response of feedback path measured by using ASW-2729D (Amon Ind.)

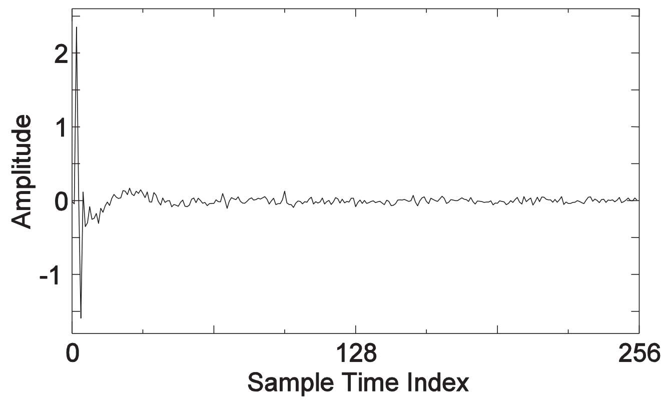

Figure 5 – Impulse response of feedback path measured by using TD307 (FUJITSU TEN)

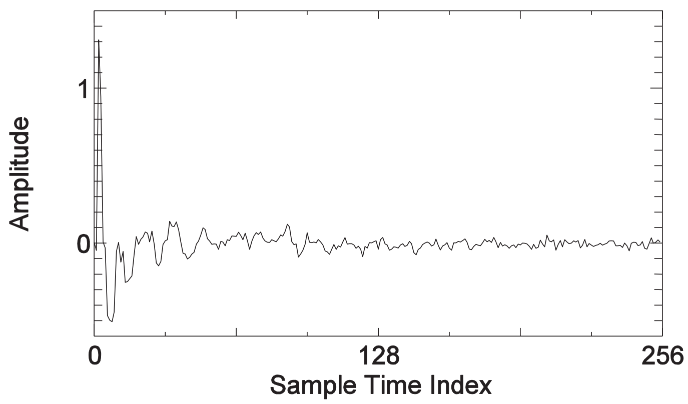

Figure 6 – Arithmetic mean of two impulse responses shown in Figs. 5 and 6

4.2 Results by NLMS algorithm

Figure 7 shows the noise reduction effect curves calculated by applying the NLMS algorithm to the estimation of the noise control filter, where "Amon", "TEN" and "Ave." denote the results calculated by using the impulse responses shown in Figs. 4, 5 and 6, respectively. In this simulation result, the ranges of the step size guaranteeing the stable performance are different as expected. Especially, we can see that the fine adjustment is required in the example that the loudspeaker TD307 (FUJITSU TEN) is applied. It follows from the result that the adjustment of the step size is essential for stably working the system when the NLMS algorithm is applied to the estimation of the coefficients of the noise control filter.

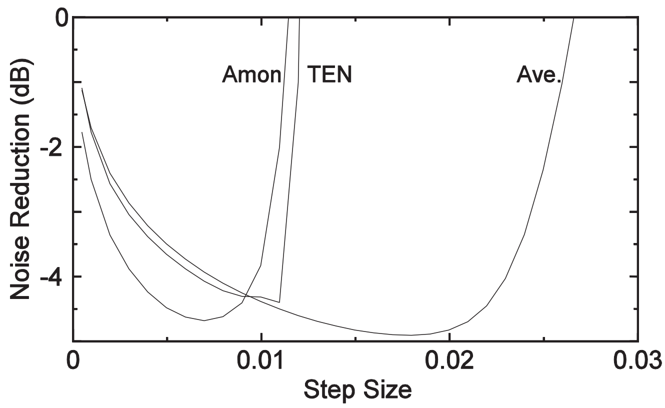

Figure 7 – Noise reduction effect curves calculated by applying NLMS algorithm

It should be noted here that the feedback paths used for the simulation are different in not only the shape but also the power gain. In practical use, the power gain also is indeterminate until the system is installed. In this simulation, we applied the NLMS algorithm whose stable condition is expected to be independent of the power of the reference signal. It is accordingly expected that the same shape of the impulse response gives the same curve independent of the power gain. The result shown in Fig. 7, however, indicates the possibility that the power gain relates to the step size providing the maximum effect. Actually, the three noise reduction effect curves are placed in order of the power gain, and moreover the power gain ratio of the impulse response measured by the loudspeaker of Amon Ind. (ASW-2729D) to that of the arithmetic mean is 2.49, which is nearly equal to the proportion of the step sizes providing the maximum effects.

Figure 8 shows the impulse response composed by using the exponential decade regular random number, where the delay of two sample periods, nearly equivalent to the distance of 10 cm as mentioned above, is inserted to that. We next confirm using the impulse response that the power gain also is one of the factors determining the range of the step size guaranteeing the stable performance to the system. Figure 9 shows the noise reduction effect curves calculated adjusting the power gain to those of the impulse responses shown in Figs. 4, 5 and 6. The result shown in Fig. 9 indicates that the power gain changes the range of the step size guaranteeing the stable performance, although the same shape impulse responses are applied. The result shown in Fig. 9 moreover demonstrates that the upper limit of the available range of the step size lowers in inverse proportion to the power gain.

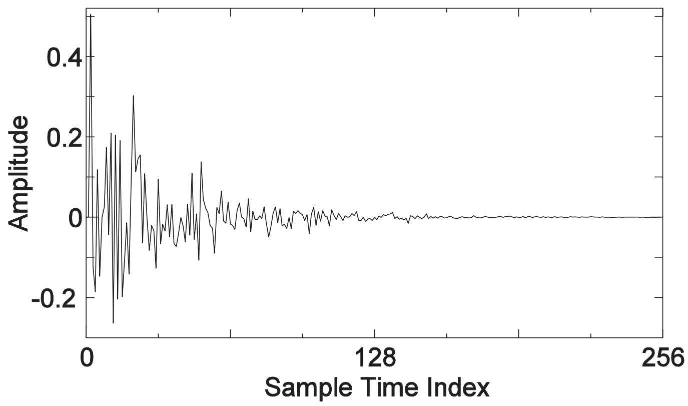

Figure 8 – Impulse response composed by using exponential decade regular random number

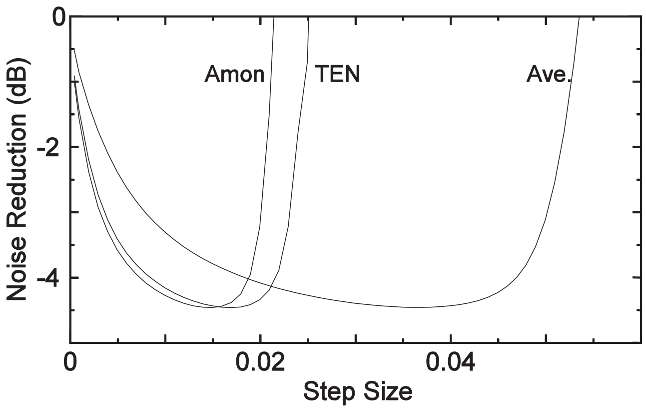

Figure 9 – Noise reduction effect curves calculated by applying NLMS algorithm to impulse response calculated using exponential decade regular random number

The comparison of the results shown in Figs. 7 and 9 also indicates that the shape of the impulse response changes the range of the step size guaranteeing the stable performance. The two results state that the adjustment of the step size is essential in practical use.

4.3 Results by Sub-RLS algorithm

As mentioned above, the sub-RLS algorithm is expected not to require the adjustment of step size. Figure 10 shows the noise reduction effect curves calculated by applying the sub-RLS algorithm to the estimation of the coefficients of the noise control filter. In this result, we can see that the step size providing the maximum effect to all the impulse responses is between 0.4 and 0.5, although the impulse responses used for the simulation are different in the power gain and the shape. The maximum effect is also improved in comparison to the results shown in Figs. 7 and 9.

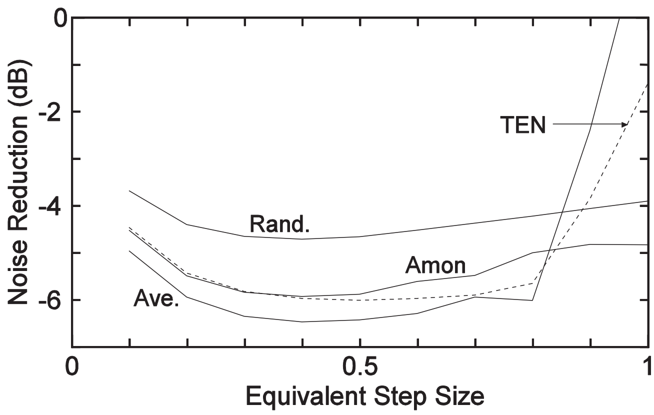

Figure 10 – Noise reduction effect curve provided by Sub-RLS algorithm

The result shown in Fig. 10, however, indicates that two impulse responses designated by “Ave.” and “FUJITSU TEN” degrade the noise reduction effect in larger than 0.8. Especially, the noise reduction effect expressed by “Ave.” diverges in the range of more than 0.9. The degradation and the divergence are caused by the approximate operation of the inverse matrix applied to the sub-RLS algorithm. The degree of the approximation, however, can be improved by repeating the estimation of the coefficients of the noise control filter as shown in Eq. (10).

Figure 11 shows the noise reduction effect curves calculated by repeating twice the operation shown in Eq. (10). We can see that the system can provide almost the same noise reduction effect independent of the shape and the power gain of the impulse responses. Especially, even in the results designated by “TEN” and “Ave.”, the system stably works. We can fix the step size at 0.4 or 0.5 without the previous measurement of the feedback path if applying the sub-RLS algorithm.

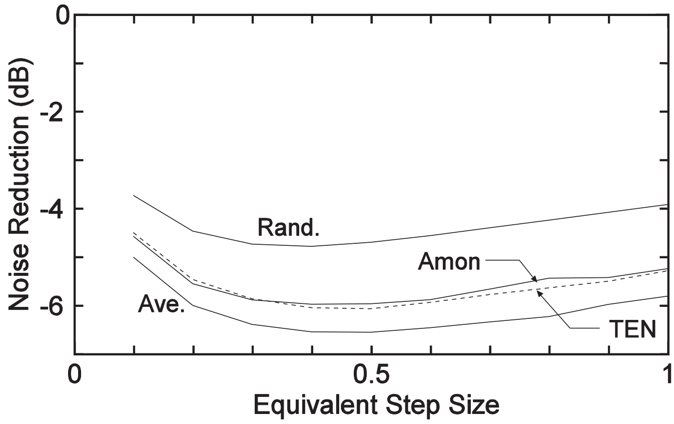

Figure 11 – Noise reduction effect provided by updating the estimation twice

5. SUMMARY

In this paper, we have indicated that the preliminary adjustment of the step size is required when applying the steepest descent method to the estimation of the coefficients of the noise control filter, and then have demonstrated that the preliminary adjustment are unnecessary when using sub-RLS algorithm. Moreover, the noise reduction effect can be thereby improved. In the near future, we increase the noise reduction effect to more than 10 dB.

ACKNOWLEDGEMENTS

Part of this research was financially supported by MEXT-Supported Program for the Strategic Research Foundation at Private Universities, 2013-2017.

REFERENCES

1. Elliott SJ., Nelson PA. Active noise control. IEEE Sig Process Mag. 1993;10(4):12-35.
2. Ohnishi K. Active soft edge noise barrier for road noise. J Acoust Soc Jpn. 2003; 59(7):420-421.
3. Hoshino T, Yamaguchi A, Ohashi T, Fujii K, Ohga J. On the application of active noise control to an electronic instrument. Proc 1993 Spring Meet Acoust Soc Jpn, p.413-414
4. Fujii K, Muneyasu M. A study on feedback type active noise control system for preventive panel of sound leakage. Proc 2015 Spring Meet Acoust Soc Jpn, p.595-596
5. Fujii K, Ohga J. A fast adaptive algorithm suitable for acoustic echo vanceller. IEICE Trans. Fundamentals, 1992;E75-A(11):1508-1515.
6. Fujii K, Muneyasu M, Hinamoto T, Tanaka Y. Analysis on the convergence property of the sub-RLS algorithm. IEICE Trans. Fundamentals, 2001;E84-A(10):2591-2594.
7. Audio/Acoustic Technical CD, Acoustics Consultant Association of Japan