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ADAPTIVE ACTIVE NOISE CONTROL SYSTEM FOR SECONDARY PATH FLUCTUATION PROBLEM

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Abstract. *An effective active noise control (ANC) system, based on identifying the so-called secondary path, can cancel the undesired noise. However, in some ANC applications, secondary path fluctuation will degrade the performance of noise cancellation. This paper proposes a novel average value algorithm with conventional filtered-X lease mean square (FXLMS) method to solve this problem. A method to tune the learning constant in FXLMS method is also presented to speed up the convergence of ANC system. Several computer simulations show that the proposed ANC method can cancel the noise effectively and still perform well with the fluctuation of secondary path.*
Keywords: Active noise control, Secondary path, FXLMS, Fluctuation

1. **Introduction.** Industrial noises often have significant power in the low frequency range (e.g., under 500 Hz). However, the conventional passive noise control (PNC) methods, suppressing acoustic noise using sound absorbing materials, generally do not work well at low frequency. The reason is that the thicknesses of the noise absorbers are not large enough, when compared with the acoustic wavelength at low frequency. Besides, it is also difficult to reduce low frequency sound being transmitted from one space to another space unless the intervening barrier is very heavy. For these reasons, passive schemes cannot control low frequency noise well. Besides, the back pressure, arousing by using absorber, will also deteriorate the performance of noise reduction.

Using artificial sound waves to cancel the unwanted noise, the theory of active noise control (ANC) was initially proposed by Lueg [12]. This method involves an electro acoustic system that cancels the primary (unwanted) noise; specifically, an anti-noise of equal amplitude and opposite phase is generated and combined with the primary noise, thus resulting in the cancellation of both noises. However, identification is important to most topics in ANC systems [2,7]. To implement an ANC system, one has to obtain the transfer functions of acoustic plant, microphone, and speakers for generating the artificial anti-noise signal. Many researchers employ the standard filtered-X least mean square (FXLMS) algorithm to develop ANC system verified the effectiveness by numerical simulations [1,5,9]. The so-called FXLMS algorithm is shown in Figure 1. Here, the yellow blocks are acoustic part and the blue ones show the artificial electrical part. The noise source is filtered through the primary acoustic path $P(z)$ and appears as a primary noise signal $d(n)$ at the error microphone. The objective of the adaptive filter $W(z)$ is to generate an appropriate anti-noise output $y(n)$ propagated by the secondary loudspeaker $S(z)$. This anti-noise signal $y'(n)$ cancels with the primary noise signal $d(n)$ to create a silent zone in the vicinity of the error microphone. The error microphone measures the residual noise $e(n)$, which is used by $W(z)$ for its adaptation to minimize the sound pressure at error microphone. Here $S'(z)$ accounts for the model of the secondary path $S(z)$ between the output $y(n)$ of the controller and the output $e(n)$ of the error microphone. The filtering of the reference signal $x(n)$ through $S'(z)$ is demanded by the fact that the output $y(n)$ of the adaptive filter is filtered through $S(z)$.

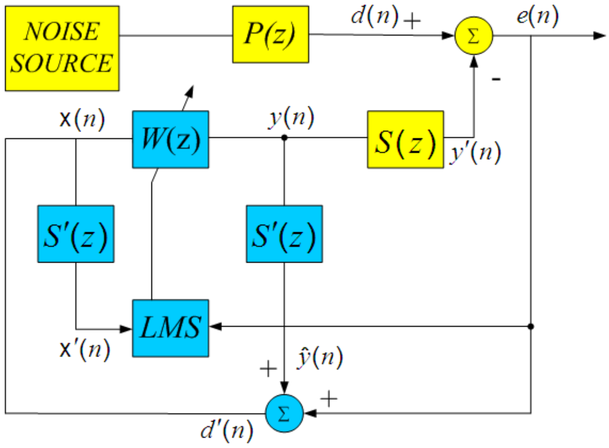


FIGURE 1. FXLMS algorithm based single-channel feedback ANC system

In general, FXLMS algorithm obtained the $S'(z)$ by offline identification method. It is computationally simple and fairly robust to the modeling errors between the secondary path and the modeling filter, but the convergence speed of FXLMS algorithm is slow. So, the noise reduction performance is inferior to that under the condition of secondary path variation. This situation especially happens when one wears an ANC headset. The secondary path of an ANC headset is the location from loudspeaker to the error microphone in the ear cup. In a practical case, one often adjusts the headset for comfort, changing the secondary path. Figure 2 shows the condition. Some researchers have developed ANC systems for headsets to reduce the environmental noise while listening to music or working in noisy environments [13,14]. However, these works did not study the problem of secondary path variation. In the past researches, Song [15] proposed a means of adding an analog feedback loop to a digital ANC system. This hybrid system could be used to reduce disturbances during the identification of the secondary path. Akhtar [8] used offline modeling method to consider the slow variation of secondary path in an FXLMS based ANC system. However, offline method could be failed due to the convergent speed of adaptive weights, when the secondary path suddenly changed [3,4]. The variable learning constant scheme and average value algorithm were also examined to decide the weighting vector $W(z)$ in digital filter in [6,10]. Accordingly, an effective way to adjust the weights of filters in an ANC must be considered.

This paper uses online identification of the secondary path characteristics to ensure the stability and maintain the noise reduction performance especially when the secondary path is time varying. An averaging algorithm is presented to average the weighting values, avoiding the abrupt change of coefficients. This method constrains the huge variation of

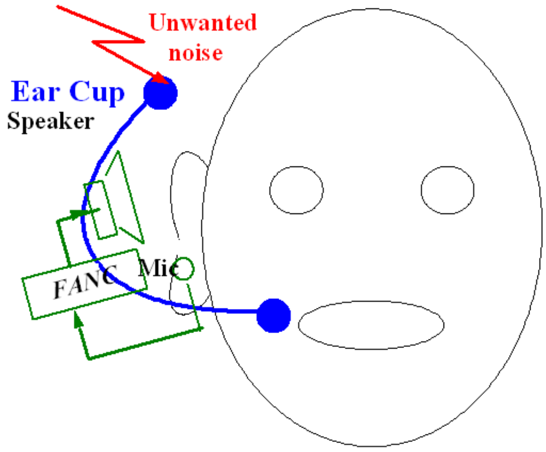


FIGURE 2. Practical case of secondary path variation

filter, and thus will not give too much change when the secondary path varied suddenly or temporarily in an ANC system. Meanwhile, an adaptive learning rate is also presented to tune the filter, helping to compensate for the practical case of secondary path variation.

2. **Proposed Method.** The proposed method is shown in Figure 3. In Figure 3, the $W(z)$ is an FIR filter of tap-weight length L, the output signal $y(n)$ is

$$y(n) = W^T(n)X(n) \tag{1}$$

where $W(n) = [w_0(n), w_1(n), \cdots, w_{L-1}(n)]^T$ is the time sample of the tap-weight vector $W(z)$ and the signal sequences $X(n) = [x(n), x(n-1), \cdots, x(n-L+1)]^T$ and $Y(n) = [y(n), y(n-1), \cdots, y(n-M+1)]^T$. Besides, $S(n)$ and $S'(n)$ are the respective time samples sequences of secondary path $S(z)$ and $S'(z)$, where $S(n) = [s_0(n), s_1(n), \cdots, s_{M-1}(n)]^T$ and $S'(n) = [s'_0(n), s'_1(n), \cdots, s'_{M-1}(n)]^T$. The signal $y'(n) = S(n)^TY(n)$ represents the cancelling signal. Thus, the residual error signal $e(n)$ is

$$e(n) = d(n) - y'(n) \tag{2}$$

Besides, we have

$$d'(n) = e(n) + \hat{y}(n) \tag{3}$$

where $\hat{y}(n) = S'(n)^TY(n)$. We can use $d'(n)$ to imitate the undesired noise $x(n)$; therefore, $x'(n)$ can be computed as $x'(n) = S(n) * X(n)$. By using the gradient estimation method, one can use the following equations to update the coefficients of adaptive filter $W(z)$,

$$w(n+1) = \bar{w}(n) + \bar{g}(n) \tag{4}$$

where

$$\bar{w}(n) = \frac{1}{n}[(n-1)\bar{w}(n-1) + w(n)] \tag{5}$$

and

$$\bar{g}(n) = \frac{1}{n^\gamma}[(n-1)^\gamma\bar{g}(n-1) + u_w(n)e(n)x'(n)] \tag{6}$$

with $0.5 < \gamma < 1$ is chosen [11]. In Equations (5) and (6), it shows the idea of average value, and the variable step size is described as $u_w(n)$. Besides, one defines the power of the residual error signal $P_x(n)$ and the power of the modelling error signal $P_e(n)$, so that the powers can be estimated from the following low-pass estimators, respectively:

$$P_e(n) = \lambda P_e(n-1) + (1-\lambda)e^2(n) \tag{7}$$

$$P_x(n) = \lambda P_x(n-1) + (1-\lambda)x^2(n) \tag{8}$$

where λ is the forgetting factor $(0.9 < \lambda < 1)$, and the ratio of the estimated powers is obtained.

$$q(n) = \frac{P_e}{P_x} \tag{9}$$

Therefore, we can compute the variable learning constant as follows,

$$u_w(n) = q(n)u_{w\,\min} + (1 - q(n))u_{w\,\max} \tag{10}$$

The constants λ and $u_{w\,\min}$, $u_{w\,\max}$ are determined experimentally [7].

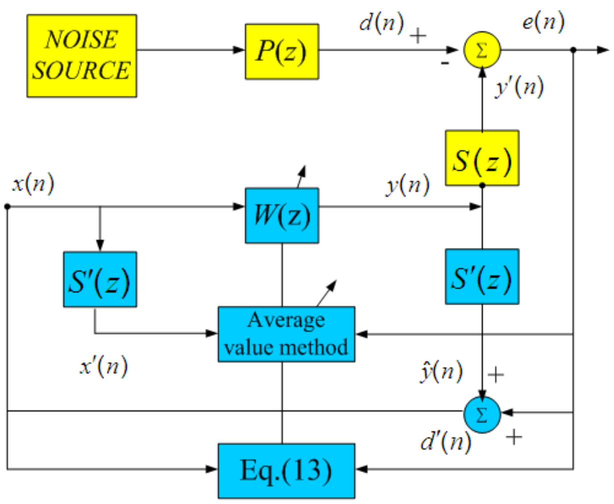


FIGURE 3. The proposed algorithm based single-channel feedback ANC system

Moreover, when $n \gg 1$ the (5) and (6) can be rewritten as follows,

$$\bar{w}(n) = \bar{w}(n-1) + \frac{1}{n}w(n) \tag{11}$$

$$\bar{g}(n) = \bar{g}(n-1) + \frac{u}{n^\gamma}e(n)x'(n) \tag{12}$$

by combining (4) and (11), (12), we have,

$$w(n+1) \approx \bar{w}(n-1)\left[I - \frac{u}{n^\gamma}R\right] + \bar{g}(n-1)\left[I - \frac{u}{n^\gamma}R\right] + \frac{u}{n^\gamma}h \tag{13}$$

Therefore, we can use average value algorithm and variable learning constant method to do the adaptation of the weights and fulfill the ANC system.

3. **Computer Simulations.** Several simulations are executed to verify the effectiveness of the proposed approach. The constants $u_{w\,\max}$ is 0.001, $u_{w\,\min}$ is 0.0007 and the sampling frequency is 8 KHz. One uses 32^{nd}-order and 32^{nd}-order FIR filters to model the secondary path and ANC filter, respectively, and uses conventional FXLMS method and the method in [8] to be the contrast. The primary acoustical path $P(z)$ and the secondary path $S(z)$ are chosen to be the same with [8], shown in Figure 4.

The first experiment uses a 300 Hz periodic signal as the undesired noise to illustrate the effectiveness of the proposed algorithm. Figure 5 shows the result, the blue lines show the original periodic noise. The results of FXLMS, method in [8] and proposed algorithm are shown by red lines in Figures 5(a)-(c), respectively. One can find that all the methods cancel the unwanted noise well. The next experiment shows the performance under the fluctuation of secondary path. One changes the frequency response of secondary path at the moment of weight adaptation 5000 to imitate the fluctuation of secondary path.

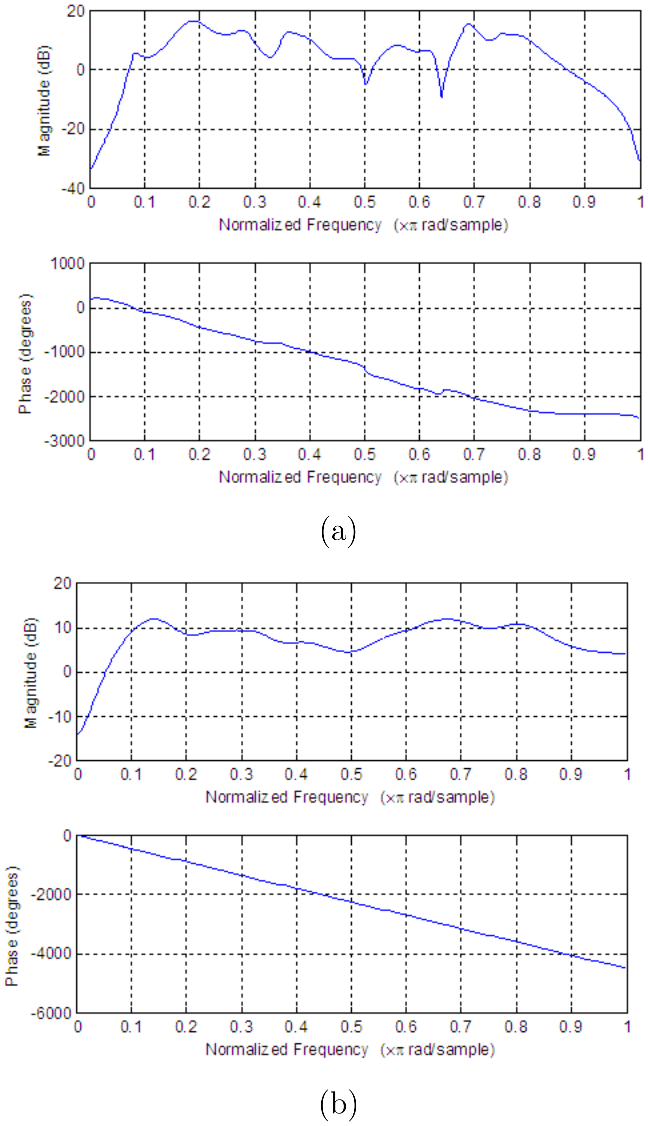
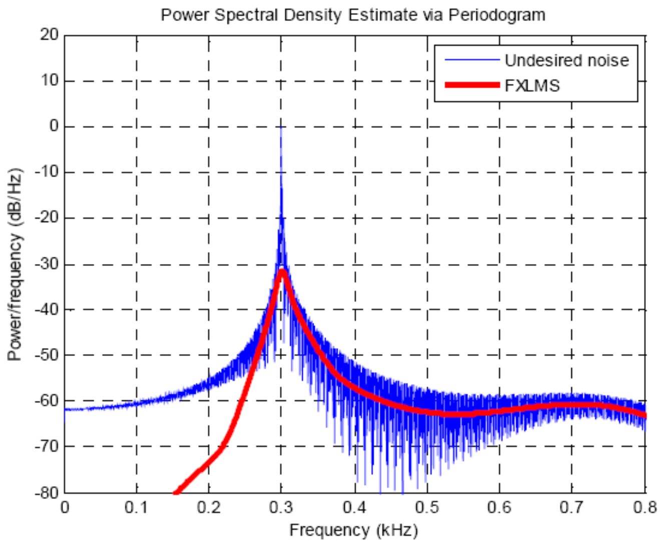
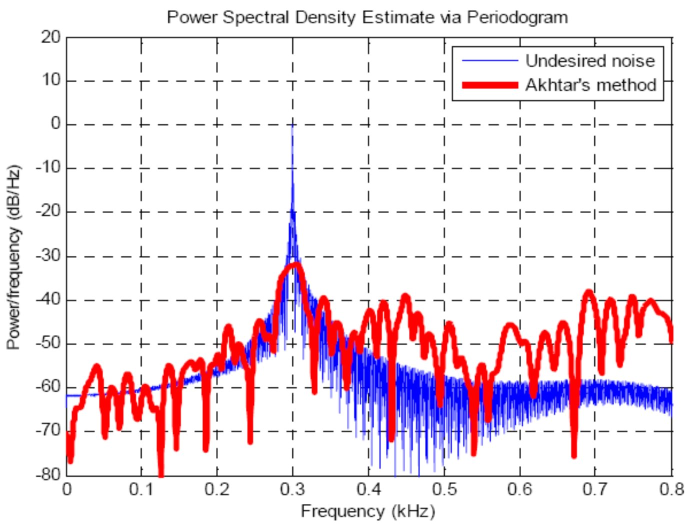


FIGURE 4. (a) The primary acoustical path $P(z)$, (b) secondary path $S(z)$

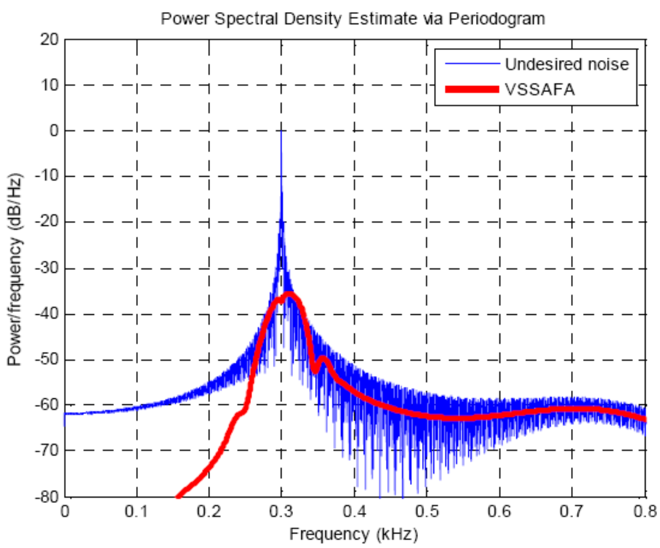
Figure 6 shows the before and after results of the frequency response in blue and red lines, respectively. The results are shown in Figure 7 and Figure 8. In order to see the fluctuation, the result of cancelling performance is shown in time domain. Figures 7(a)-(c) show the results of cancelling 300 Hz noise with fluctuation by FXLMS, the method in [8] and proposed algorithm, respectively. It is clear that the proposed algorithm also performs excellently in cancelling the signal with secondary path fluctuation. Nevertheless, the conventional FXLMS algorithm cannot cancel the noise satisfactorily. The method in [8] can not give the same performance as in previous experiment, either. The last experiment shows the enhancement in cancelling broadband noise. Figure 8(a) shows the performance of FXLMS algorithm with 200∼300 Hz nonlinear broadband noise. One can find the linear FXLMS based method can not control nonlinear noise any more. Figure 8(b) shows the result by method in [8] and Figure 8(c) shows the result by proposed method. By using the variable learning constant method, one can find the proposed method can converge faster than the method in [8]. In the meanwhile, the proposed method also provides the better performance than conventional ones.



(a)



(b)



(c)

FIGURE 5. Narrowband noise canceling performance 300 Hz, blue: ANC OFF, red: ANC ON. (a) FXLMS (b) method in [8] (c) proposed method.

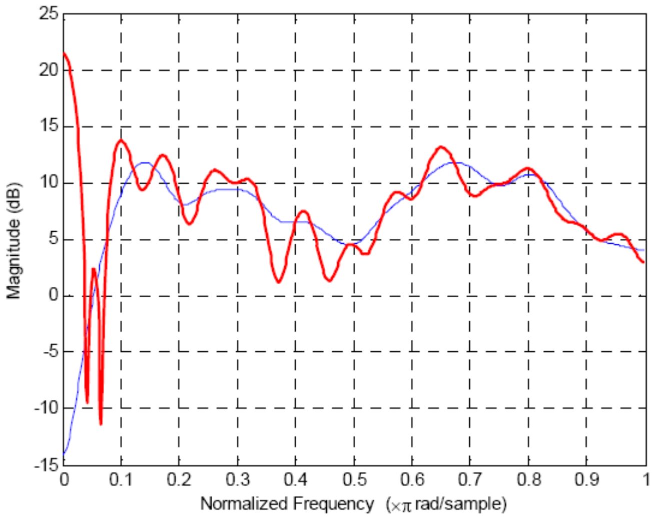


Figure 6. The fluctuation of secondary path, blue: BEFORE, red: AFTER

These results illustrate that the conventional FXLMS algorithm can only control the narrowband noise yet the proposed algorithm can cancel both the periodic and broadband noises well. The tuning method of the learning rate also helps to achieve faster convergence. Besides, the comparison of computing effort is also given in Table 1.

Table 1. Computing effort of the Akhtar's method and proposed method

method	**Number of Computations per Iteration**		
	Multiplications	Additions	Total operations (if M=L)
Method in [8]	3L+4M+10	3L+4M+5	14L+15
Proposed method	6L+2M+10	4L+2M+5	14L+15

Remark 3.1. *As is well known, FXLMS has a low convergent speed. However, in our ANC design, the coefficients of filter can be searched for by using the proposed method effectively. In a practical case of ANC headset, the proposed scheme can constrain the change of filter especially when the user adjusts the ear cup suddenly or temporarily, producing a robust system. The authors also discuss and analyze the performance of different acoustic plants and noise types to verify the effectiveness of variations in the environment/measured signal. Experiment of suddenly variation of secondary path is also examined to verify the performance of proposed system.*

4. **Conclusions.** One derives the average value method for ANC system with the fluctuation of the secondary path. The nonlinear processing ability of the proposed method improves the noise cancellation performance in an ANC system. One also designs the adjustable learning rate algorithm to accelerate the convergence. Computer simulations have been carried out to demonstrate the performance of the proposed method as a useful method for nonlinear ANC system. Compared to conventional methods, it is obvious that the proposed algorithm shows the enhancement in nonlinear broadband noise cancellation. The proposed algorithm is also versatile and can be used in other applications.

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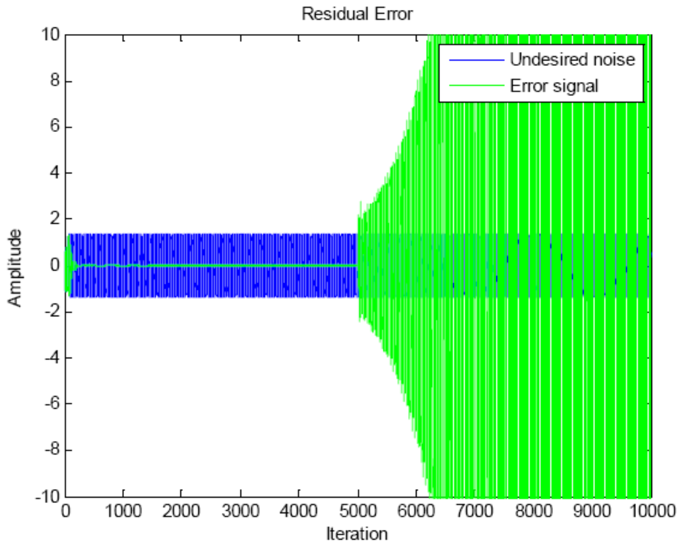
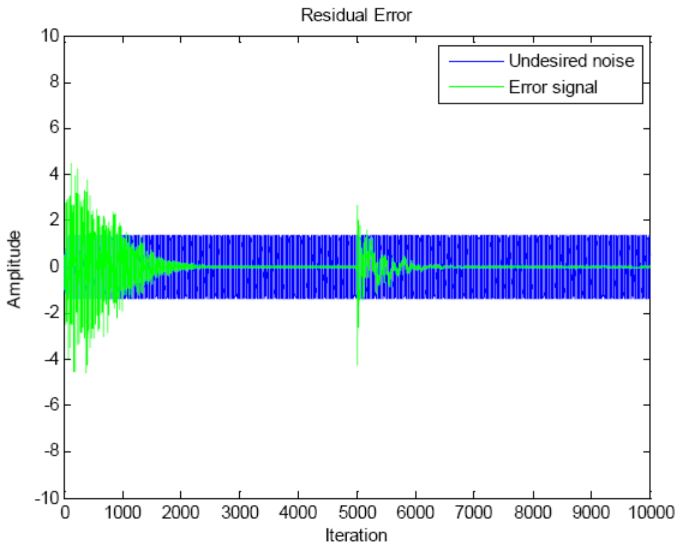
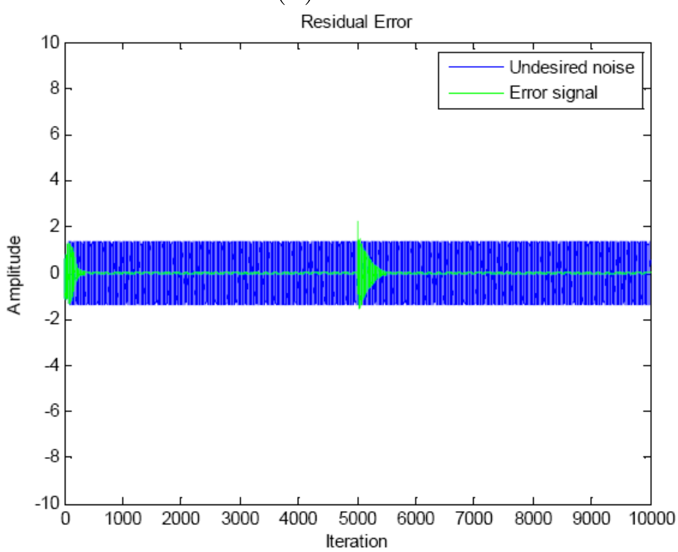


(a)

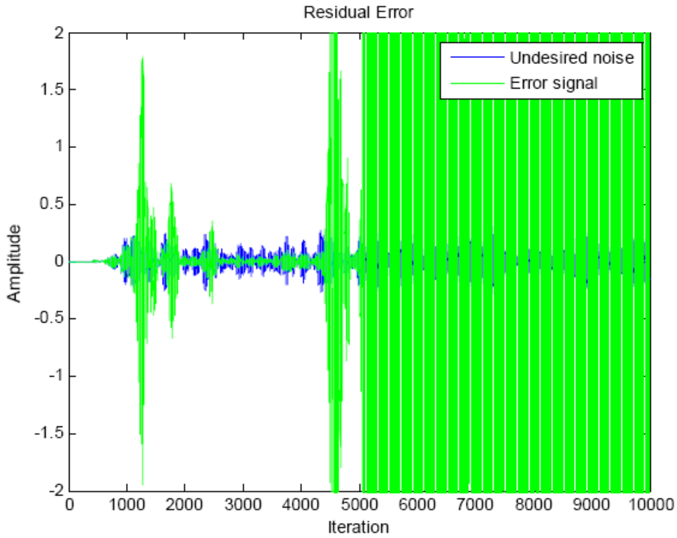


(b)

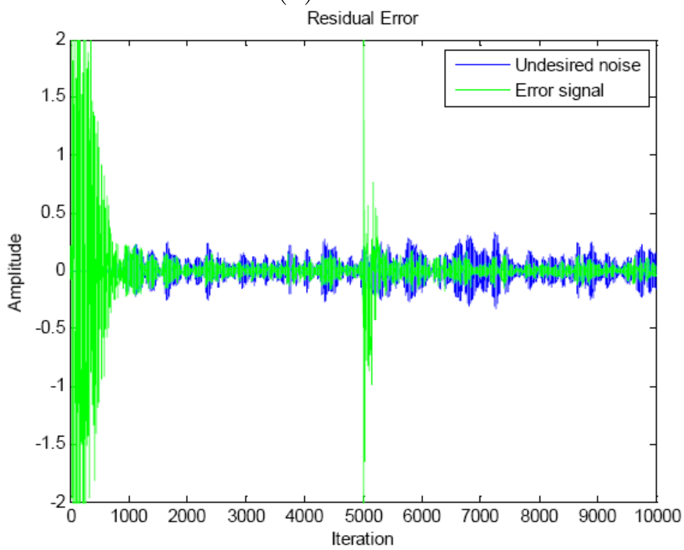


(c)

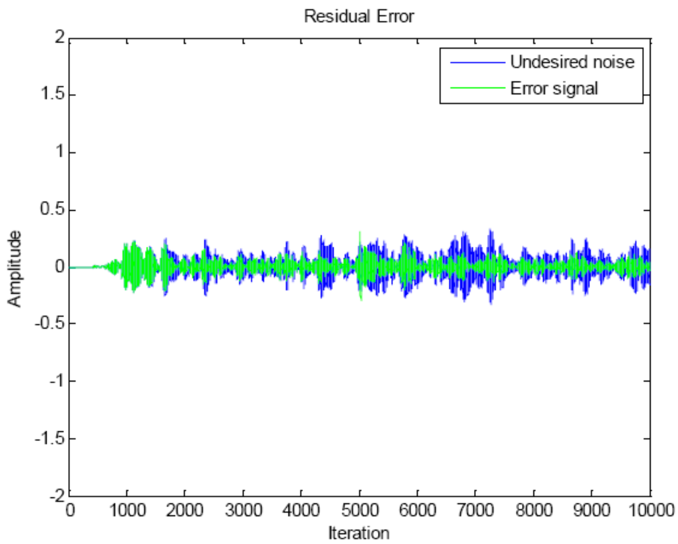
FIGURE 7. Narrowband noise canceling performance 300 Hz with fluctuation of secondary path, blue: ANC OFF, green: ANC ON. (a) FXLMS (b) method in [8] (c) proposed method.



(a)



(b)



(c)

FIGURE 8. Broadband noise canceling performance 200∼300 Hz, blue: ANC OFF, green: ANC ON. (a) FXLMS (b) method in [8] (c) proposed method.

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