# A New Double Adaptation Algorithm for Acoustic Noise Control

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Abstract: Conventional adaptive algorithms such as LMS are deployed for modeling an unknown path in a system to solve a problem. But, in a lot of applications, there is more than one unknown path. In such cases conventional algorithms not aonly can not model the secondary path, but also any estimation of the secondary path bring a level of instability to the adaptation algorithm. The Filtered-X LMS algorithm (FXLMS) is suggested in literatures as an adaptive filter algorithm for using in the presence of secondary path. However, Filtered-X LMS algorithm suffers the need for an estimation of the secondary path. Here, SAR algorithms and double FXLMS algorithm are respectively proposed and introduced for simultaneously adaptive estimation of the secondary path besides the main path. The proposed approaches are numerically analyzed and evaluated through simulations results

Keywords— Adaptive Filtering, LMS Adaptation, Filtered-x LMS Adaptation, Smart Acoustic Room

# 1. Introduction

Nowadays, Acoustic Noise Cancellation (ANC) is widely used for different applications. One of them is cancellation of noise generated by fans, such as the ones found in air conditioning, computers, ventilation systems, etc.. Normally, fan noise is in the form of an unwanted audio signal. Here, by processing the fan noise record through the microphones, it is adaptively estimated and canceled.

Adaptive Filters (AF) perform the estimation in comparison to a desired signal. Similarly, ANC research topic in connection with AF has been deployed in hand free telephony or in teleconferencing systems and acoustic echo cancellers (AEC) [1]. Echo is an unwanted acoustic signal that is generated due to acoustic coupling between a loudspeaker and a microphone in a room. An ANC system [2] is another example for reducing the acoustic noise in a location of the room. Today, humans suffer from noise pollution besides the other types of pollutions. Flight cabin is a very common example of acoustic noise where a microphone and a channel for the same purpose is predicted. The flight cabin noise generated by the air flow and aircraft engine is very similar to the problem of fan noise cancellation. Similarly smart acoustic rooms wherein acoustic null points are generated by ANC systems are different faces of the same problem.

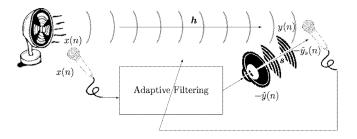


Figure 1. Active fan adaptive fan noise cancellation

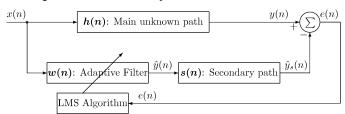


Figure 2. The block diagram of adaptive fan noise cancellation with consideration of the secondary path s(n).

### 2. Filtered-x LMS

A conventional adaptive algorithm such as the LMS algorithm is likely to be unstable in applications with phase shift (delay) introduced by the secondary path. The well-known Filtered-x LMS-algorithm (FXLMS) is, however, an adaptive filter algorithm which is suitable for adaptation at the presence of a secondary path. It is developed from the LMS algorithm, where a model of the dynamic system between the filter output and the estimate (i.e. the secondary path) is introduced between the input signal and the algorithm for the adaptation of the coefficient vector.

Figure 1 shows an example system with requirement of estimation of a secondary path. As it is seen in the figure 1, an adaptive filter is deployed to estimate the acoustic path between the fan source and the right-end microphone, the estimated path is convolved to the source fan signal x(n), and the revers of the result  $-\hat{y}(n)$  is played through the speakers to the right-end microphone. Indeed the right end microphone should receive the far-end signal y(n) together with the reverse of its estimation  $-\hat{y}(n)$  which implies the acous-

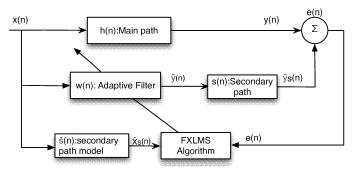


Figure 3. The block diagram of adaptive fan noise cancellation with consideration filtered-x LMS algorithm.

tic error of fan noise adaptive synthesizing. The error signal e(n) is required for adaptation of the filter. However, there is an acoustic path between the filter output speaker and error recorder microphone as called secondary path s(n) which affects whole adaptation performance and leads to instability of the LMS adaptation algorithm. Figure 2 shows the block diagram of whole adaptation system with consideration of the secondary path.

To overcome the problem of the secondary path between the speaker and the microphone, filtered-x LMS algorithm is used instead of LMS for the reason of resolving the instability caused by the secondary path. The FXLMS block diagram is shown in Figure 3.

The filtered-x LMS algorithm is summarized as follows:

$$y(n) = \sum_{i=0}^{p-1} h(n)_i x(n-i) + v(n),$$
 (1)

i.e. fan noise and environment noise

$$\hat{y}(n) = \sum_{i=0}^{p-1} w(n)_i x(n-i), \qquad (2)$$

i.e. adaptive filter output

$$e(n) = y(n) - \sum_{i=0}^{s-1} s_i \, \hat{y}(n-i)$$
 (3)

$$\hat{\boldsymbol{w}}(n+1) = \hat{\boldsymbol{w}}(n) + \mu e(n) \mathbf{x}_{\hat{\mathbf{s}}}(\mathbf{n})$$
 (4)

where n is the index of the current input sample, p is the length of main path, s is the length of the secondary path,  $\mathbf{x}(n) = [x(n), x(n-1), \dots, x(n-p+1)]^T$  is the input signal segment (fan sound),  $\mathbf{h}(n) = [h_0(n), h_1(n), \dots, h_{p-1}(n)]^T$  is the main path from the fan to the error recorder microphone (right-end microphone),  $\mathbf{s}(n) = [s_0(n), s_1(n), \dots, s_{s-1}(n)]^T$  is the secondary path,  $\mathbf{w}(n) = [w_0(n), w_1(n), \dots, w_{p-1}(n)]^T$  is adaptive filter taps.

Although, FXLMS overcomes the problem of instability of LMS in presence of the secondary path, it still suffers from the need for availability of the secondary path estimation.

# 3. The proposed method

#### 3.1 Smart Acoustic Room (SAR)

In deployment of Filtered-X LMS, the convolution of the reference signal to the model of the previously obtained secondary route (secondary path) is used by an input to the LMS; to cancel the effect of the secondary path.

Therefore, the model of the secondary path is required. However, in the proposed method which SAR system [3] [4] is deployed, does not require the estimation of the secondary path. SAR adaptation algorithm has been originally invented

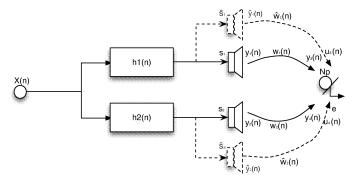


Figure 4. The smart Acoustic Room

for an smart acoustic room (SAR) wherein the acoustic response between two (or more) points could be controlled. It means to have a well estimation of the acoustic path between two points and then to make the appropriate signal to cancel an unwanted noise or to emphasis a desired signal. Figure 4 shows the proposed structure of smart acoustic room [5] [6]. An especial case of SAR is one of two filter to be 1 and just the other filter h to be under adaptation. This especial case is called here  $SAR_h$ . The  $SAR_h$  approach to the secondary path estimation is summarized as follows:

$$e(n) = x(n) * w_1(n) + x(n) * h(n) * w_2(n),$$
 (5)

where h(n) is the adaptive filter tap. Using the error value e(n), the two paths are updated simultaneously as follows:

$$\hat{w}_1(n+1) = \hat{w}_1(n) + 2\mu(e(n))x(n-i)$$
 (6)

$$\hat{w}_2(n+1) = \hat{w}_2(n) + 2\mu(-e(n))y(n-i)$$
 (7)

Where h(n) has the following relation with  $\hat{w}_1$  and  $\hat{w}_1$ :

$$h(Z) = -\frac{w_1(Z)}{w_2(Z)} \tag{8}$$

## 3.2 DFXLMS

Two FXLMS are employed in parallel as shown in the block diagram of Figure 5. The two filters are updated alternatively at each iteration one after the other.

The update equations of the two filters are as follows:

$$h_i(n+1) = h_i(n) + 2\mu e(n)x(n-i) * w_2(n)$$
 (9)

$$w_{2i}(n+1) = w_{2i}(n) + 2\mu(-e(n))x(n-i) * h(n)$$
 (10)

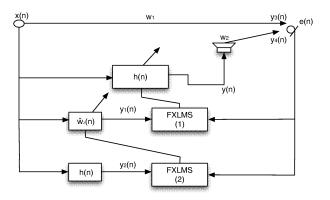


Figure 5. Double Filter-X LMS block diagram

Table 1. Simulation Conditions

Iterations Iterations	10,000
Monte Carlo Average	10
Tap Number	16
Main Path	rand*exp(-8*i/M)
Secondary Path	rand*exp(-8*i/M)
Step Size	0.1

### 4. Results and Simulations

The simulations were done based on the Monte-Carlo runs o 10000 samples per iteration. The filter length was 16 taps. The main path and secondary path were both exponential decays of random coefficients. The step size was 0.1.

Table 1 summarizes the simulation conditions. The evaluation deployed criteria is mean squared error ratio in dB:

$$Errorratio[db] = 10log_{10} \sum_{n=1}^{N} e^{2}(n)$$
 (11)

Figures 6 - 13 depict the simulation results of deployment of LMS, FXLMS, general SAR, SAR $_h$  and the newly introduced DFXLMS algorithms in connection with the presence of a secondary path. Instability of LMS is clearly observed (Figure 6), while FXLMS performs well (Figure 6). SAR methods without any prior knowledge of secondary path performs well (Figures 7-10). As well, DFXLMS performs acceptable (Figures 11-13).

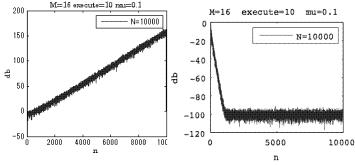


Figure 6. Learning curves of the LMS (right) and FXLMS (left) algorithms in the presence of the secondary path.

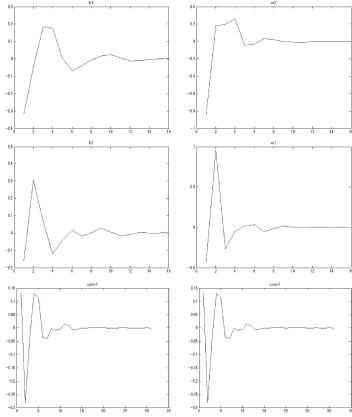


Figure 7. General SAR results (two filters alternative adaptation), left-up:  $\mathbf{h_1}$  and right-up:  $\mathbf{w_2}$ , left-middle:  $\mathbf{h_2}$ , and right-middle:  $\mathbf{w_1}$ , bottom row:  $h_1*w_1$  and  $h_2*w_2$ : the convergence of the convolutions results to each others indicates the convergence of the adaptation.

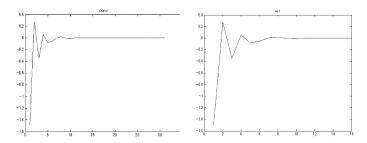


Figure 8. SAR<sub>h</sub> results for convolutions  $h * w_2$  and  $w_1$ .

# 5. Conclusion

In the case of availability of an estimation of the secondary path, filter-x LMS adaptation as a common solution is studied and its dominant performance with respect to LMS adaptation is approved through simulation results. SAR algorithms are proposed for the same purpose, while they benefit us by being needless to any estimation of the secondary path. SAR adaptively synthesizes both the main path and the secondary path. In addition, double-FXLMS as a combination of two interlinked FXLMS with alternative iteration steps is introduced to overcome the the FXLMS need for prior knowledge of the secondary path. The newly introduced DFXLMS has acceptable performance.

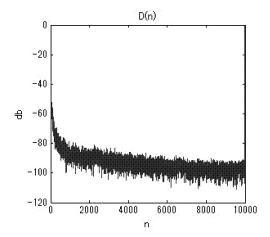


Figure 9. Learning curves for general SAR and SAR<sub>h</sub>.

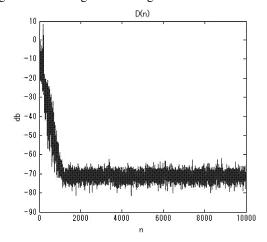


Figure 10. Learning curves for general SAR and  $SAR_h$ .

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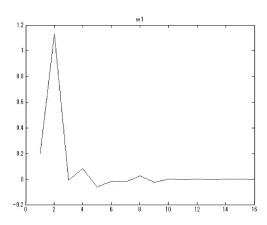


Figure 11. DFXLMS result for  $w_1$ .

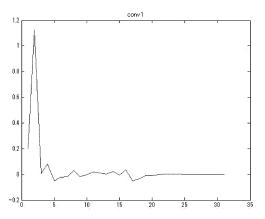


Figure 12. DFXLMS result for  $h * w_2$ .

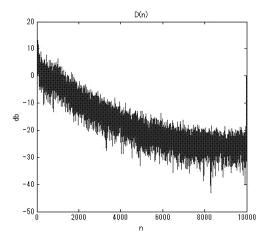


Figure 13. Learning curve of DFXLMS adaptation.