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Acoustic Echo Cancellation by Adaptive Combination of Normalized Sub band Adaptive Filters by Using Stochastic Gradient Algorithm

Sunita, Parveen Bajaj

Electronics & Communication Engg. Deptt.
Maharishi Markandeshwar University, Sadopur
Ambala, India

Abstract: *Acoustic echo is a common occurrence in today's telecommunication systems. It occurs when an audio source and sink operate in full duplex mode; an example of this is a hands-free loudspeaker telephone. In this situation the received signal is output through the telephone loudspeaker (audio source), this audio signal is then reverberated through the physical environment and picked up by the systems microphone (audio sink). The effect is the return to the distant user of time delayed and attenuated images of their original speech signal. The signal interference caused by acoustic echo is distracting to both users and causes a reduction in the quality of the communication. Adaptive filtering techniques are used to reduce this unwanted echo thus increasing communication quality. The input signal of the adaptive filter is highly correlated and the impulse response of the echo path is very long. These characteristics will slow down the convergence rate of the adaptive filter if the well-known normalized least mean-square (NLMS) algorithm is used. The normalized subband adaptive filter (NSAF) offers a good solution to this problem because of its decorrelating property, Which requires a tradeoff between fast convergence rate and small steady state mean-square error (MSE). The proposed combination is carried out in sub band domain and the mixing parameter that controls the combination is adapted by means of a stochastic gradient algorithm which employs the sum of squared sub band errors as the cost function. For the adaptation of the component filters, in addition to the conventional decoupling update method, we also propose a coupling one, which can further improve the performance of the adaptive combination scheme.*

Keywords: *Acoustic echo canceller, Steady-state Mean-Square Error (MSE), Normalized least-Mean-Square (NLMS) algorithm, Sub band Adaptive Filters (SAFs), Normalized Sub band Adaptive Filters (NSAFs), hands-free telephone, teleconferencing system.*

1. INTRODUCTION

A common problem encountered in hands-free telephones and teleconferencing systems is the presence of echoes which are generated acoustically by the coupling between the loudspeaker and the microphone via the impulse response of a room. In recent years, there has been a great interest in the use of adaptive filters as acoustic echo cancellers to remove echoes. An adaptive filter can be characterized by its structure and adaptive filtering algorithm. The transversal filter with the well-known normalized least-mean-square algorithm is one of the most popular adaptive filters because of its simplicity and robust performance. In acoustic echo cancellation (AEC) applications, the speech input signal of the adaptive filter is highly correlated and the impulse response of the acoustic echo path is very long. These two characteristic will slow down the convergence rate of the acoustic echo canceller if the NLMS-based adaptive filter is used to remove echoes. One technique to solve the above problem is subband adaptive filtering. In conventional subband adaptive filters (SAFs), each subband uses an individual adaptive subfilter in its own adaptation loop, which decreases the convergence rate of SAFs because of the aliasing and band-edge effects. Lee and Gan presented a normalized SAF (NSAF) from the principle of minimum disturbance, with its complexity close to that of the NLMS-based adaptive filter.

Its central idea is to use the subband signals, normalized by their respective subband input Variances, to update the tap weights of a fullband adaptive filter. Although these SAFs can obtain fast convergence rate when applied to AEC applications, they all require a tradeoff between fast convergence rate and small steady-state mean-square error (MSE) because of the use of a fixed step size. They all need to estimate the system noise power in advance.

Recently, an adaptive combination of fullband adaptive filters has been proposed. The merit of this combination is that it can obtain both fast convergence rate and small steady-state MSE without estimate of the system noise power. More recently, a combination of subband adaptive filters for AEC has been proposed. In this paper, a new subband adaptive combination scheme to deal with the tradeoff problem encountered in acoustic echo cancellers which are implemented by NSAFs. The proposed combination is carried out in subband domain and the mixing parameter that controls the combination is adapted by means of a stochastic gradient algorithm which employs the sum of squared subband errors as the cost function.

We have been improving adaptive combination of NSAFs. In which large convergence rate & small steady state MSE can be greatly achieved by using less amount of less amount of adaptive filters as comparison to previous adaptive combination scheme.

2. SIGNAL MODEL AND NSAF

2.1Signal Model of Acoustic Echo Canceller

A block diagram for acoustic echo canceller is illustrated in Fig.1.When the far-end signal u(n) goes through the echo pathW_0 (z) of a room, the acoustic echo is produced. The acoustic echo is picked up by the microphone together with the near-end signal η(n), resulting in the microphone signal d(n).The near-end signal may contain the system noise v(n) and near-end speech s(n).The goal of the adaptive filter wˆ(k) is to produce a replica of the echo signal, y(n), by adaptively adjusting the tap-weights of Wˆ(z).Then y(n) can be used to cancel the echo by subtracting it from the microphone signal d(n).

2.2 NSAF

Fig.2 shows the structure of the NSAF, which is a subband representation of Fig. 1.subband structure, both the microphone signal d(n) and the far-end signal u(n) are partitioned into N subband signals via the analysis filter $H_i(z)$,i= 0,1,..,N-1.The subband signals $d_i(n)$ and $y_i(n)$ for i = 0,1,.., $N-1$ are all critically decimated to a lower sampling rate. The decimated subband error signals are then defined as $e_{i,D}(n)=d_{iD}(n)-y_{iD}(n)$,$i$ = 0,1,…,N-1.Here we use the variable n and k to index the original and decimated sequences, respectively. Then the update of the NSAF can be formulated by

$$\hat{w}(k+1) = \hat{w}(k) + \mu \sum_{i=0}^{N-1} \frac{\mathbf{u}_i(k)}{\delta + \|\mathbf{u}_i(k)\|^2} e_{i,D}(k) \qquad (1)$$

Where

$\mathbf{u}_i(k) = [u_i(kN), u_i(kN-1), \ldots, u_i(kN-M+1)]^T$. M is the length of the adaptive filter wˆ(k),
μ is the step-size, andδ is the regularization parameter.

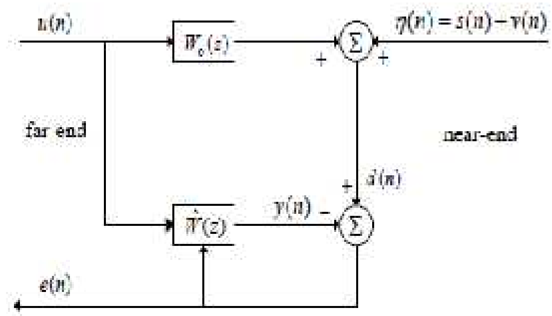

Fig. 1. Block diagram for an acoustic echo canceller.

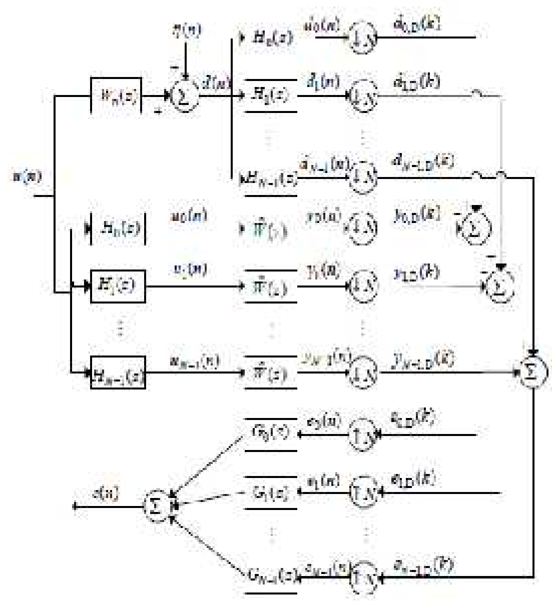

Fig. 2. Structure of the NSAF.

3. ADAPTIVE COMBINATION OF TWO NSAFS

It is well known that for an adaptive filter, a large (small) step-size yields a fast (slow) convergence rate but a large (small) steady-state MSE. The fundamental idea of adaptive combination is that two adaptive filters with different step-sizes are adapted separately, and the output signals of the component filters are combined by a mixing parameter in such a manner that the advantage of both filters is kept, i.e., the fast convergence rate from the large step-size adaptive filter and the small steady-state MSE from the small step-size adaptive filter.

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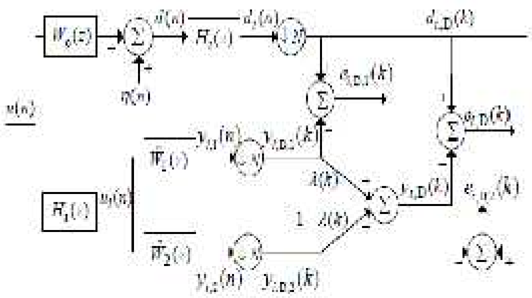


Fig. 3. Adaptive combination of two NSAFs for one subband structure.

The proposed subband adaptive combination scheme using two NSAFs (called the CNSAF). The update of the NSAFs in Fig. 3 can be written as.

$$\begin{cases}\hat{\mathbf{w}}_1(k+1)=\hat{\mathbf{w}}_1(k)+\mu_1\sum_{i=0}^{N-1}\frac{\mathbf{u}_i(k)}{\delta+\|\mathbf{u}_i(k)\|^2}e_{i,D,1}(k)\\ \hat{\mathbf{w}}_2(k+1)=\hat{\mathbf{w}}_2(k)+\mu_2\sum_{i=0}^{N-1}\frac{\mathbf{u}_i(k)}{\delta+\|\mathbf{u}_i(k)\|^2}e_{i,D,2}(k)\end{cases} \qquad (2)$$

where

$$\begin{cases}e_{i,D,1}(k)=d_{i,D}(k)-y_{i,D,1}(k)\\ e_{i,D,2}(k)=d_{i,D}(k)-y_{i,D,2}(k)\end{cases} \qquad (3)$$

and

$$\begin{cases}y_{i,D,1}(k)=\hat{\mathbf{w}}_1^T(k)\mathbf{u}_i(k)\\ y_{i,D,2}(k)=\hat{\mathbf{w}}_2^T(k)\mathbf{u}_i(k)\end{cases},\quad i=0,1,\ldots,N-1. \qquad (4)$$

we assume $\mu_1 > \mu_2$. Then $\hat{\mathbf{w}}_1(k)$ has a faster convergence rate but a larger steady-state MSE than $\hat{\mathbf{w}}_2(k)$ .The subband outputs of the overall filter are

$$y_{i,D}(k)=\lambda(k)y_{i,D,1}(k)+[1-\lambda(k)]y_{i,D,2}(k),\quad i=0,1,\ldots,N-1 \qquad (5)$$

Where $\lambda(k)$ is the mixing parameter that lies between zero and one. Using (4) in (5) yields.

$$\begin{aligned}y_{i,D}(k)&=\lambda(k)\hat{\mathbf{w}}_1^T(k)\mathbf{u}_i(k)+[1-\lambda(k)]\hat{\mathbf{w}}_2^T(k)\mathbf{u}_i(k)\\ &=\{\lambda(k)\hat{\mathbf{w}}_1(k)+[1-\lambda(k)]\hat{\mathbf{w}}_2(k)\}^T\mathbf{u}_i(k).\end{aligned} \qquad (6)$$

Instead of directly adjusting $\lambda(k)$, we will adapt a variable $\alpha(k)$ that defines $\lambda(k)$ as sigmoidal function.

$\lambda(k)= 1/1+e^{-\alpha(k)}$

The update equation for $\alpha(k)$ is given by

$$\alpha(k+1)=\alpha(k)+\mu_\alpha\lambda(k)[1-\lambda(k)]\sum_{i=0}^{N-1}e_{i,D}(k)[y_{i,D,1}(k)-y_{i,D,2}(k)]$$

Where μ_α is the step-size for adapting $\alpha(k)$.

4. IMPROVED ADAPTIVE COMBINATION SCHEME

In improved adaptive combination scheme, large convergence rate and small steady state MSE can be greatly achieved by using less amount of adaptive filters as comparison to previous adaptive combination scheme. This can be achieved by using the idea of adaptive combination of speech input signal before going to the NSAFS. This improved CNSAF is called ICNSAF. The algorithm of Improved ACNSAFs such as:

- Partitions the full band input and desired signals into N sub bands.
- Decimates the sub band signals from the original sampling rate fs to the lower rate of fs /N. The long M-tap filter is now replaced by N shorter MS-tap FIR filters (where MS <M) operating in parallel at a lower rate. The analysis filter bank consists of N parallel band pass filters that are designed to partition the full band signal into N overlapped frequency bands.
- Before going into NSAF filter they are adaptively combined by using a mixing parameter lambda.
- A synthesis filter bank is used to combine the N sub band output signals, $y_{i,D}(n)$, i = 0, 1, . . .,N − 1, into a full band signal y(n). The synthesis filter bank consists of a bank of interpolators that up sample the sub band signals before filtering and adds these sub band signals. The main advantage of this proposed method is that, it has requires less no. of filters. As it requires less no. of filters also requires less memory.

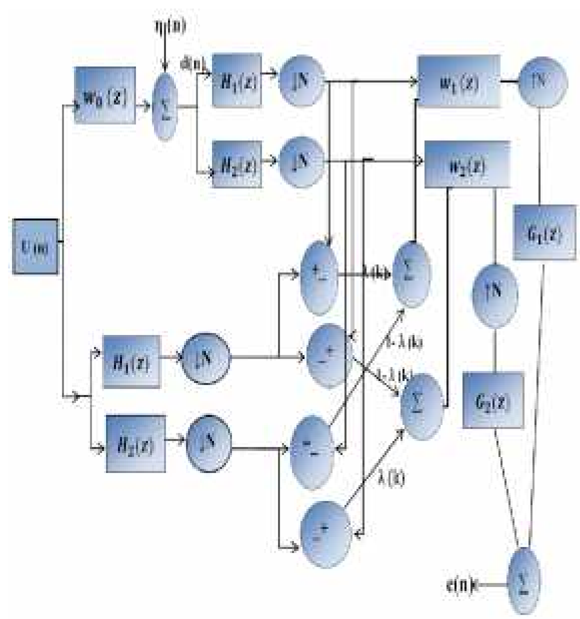


Fig. 4 Improved Adaptive combination of NSAFs

5. EXPERIMENTAL RESULTS

The full band and sub-band systems, adaptive combination of sub-band adaptive filters and its improvement were modeled in MATLAB Simulink and many simulations for different inputs and number of sub-bands were performed. For the adaptive algorithm several different algorithms can be used, but the most common one is the normalized least mean squares (NLMS). The order of the NLMS filters was chosen from N=64 to N=2 .The designs were made in MATLAB Simulink environment and the simulations were run for speech input. A reverberating effect was added to the input by an artificial Schroeder reverberate which contained four comb filters in parallel and two all-passes filters series connected. The first estimation of a system capability is represented by SNR (signal to noise ratio).The second estimation of a system capability is represented by the (output error-voice input) , but in order to measure its potential, Echo Return Loss Enhancement (ERLE) should be computed; it is defined as the ratio of the power of the desired signal over the power of the residual signal.

$$ERLE = 10\log_{10}\frac{E(d^2(n))}{E(e^2(n))}$$

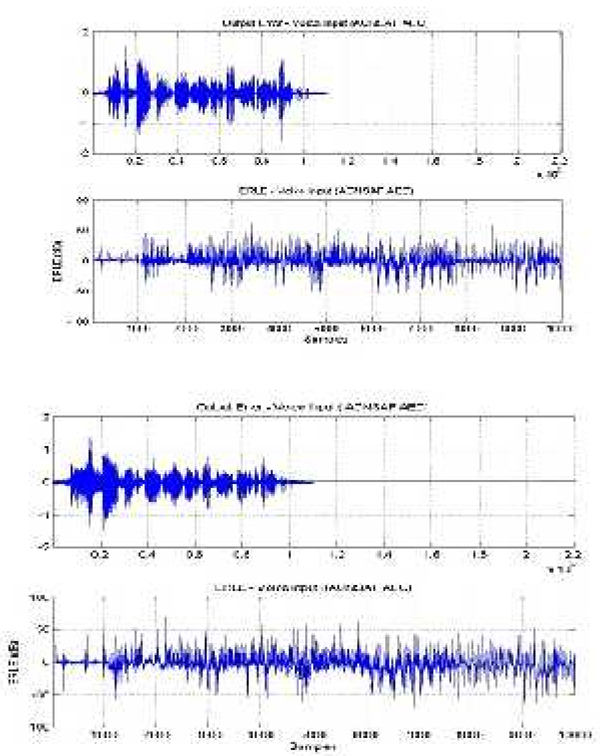


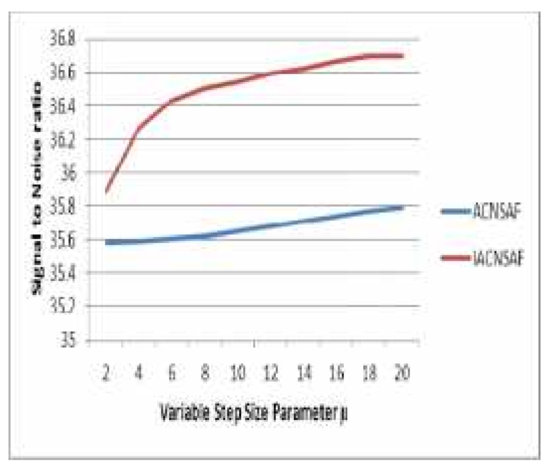
Figure 5.1: Performance Analysis of ERLE and Output Error-Voice Input of ACNSAF and IACNSAF for Variable Step Size μ=20.

Figure 5.1 shows the MATLAB Simulink results of adaptive combination of NSAFs, and Improved ACNSAFs. The MATLAB Simulink results mainly include waveform of original speech, error speech, echoed speech and output error-voice input and finally ERLE performances.

Now comparison based on SNR (signal to noise ratio) is shown in table 5.1.which shows that IACNSAF has greater SNR than ACNSAF. Graph 5.1 shows graphical representation of MATLAB observation as shown in table 5.1.

Table 5.1: Comparison Analysis of SNR of ACNSAF and IACNSAF Keeping α=4,=0.5 and Varying Step Size Parameter μ

Step size parameter	ACNSAF	IACNSAF
2	35.5819	35.8952
4	35.5923	36.2693
6	35.6089	36.4327
8	35.6227	36.505
10	35.6478	36.5452
12	35.6768	36.5954
14	35.7088	36.6242
16	35.7417	36.6645
18	35.7711	36.6979
20	35.792	36.6964



Graph 5.1: Graphical Representation of Table 5.1

6. CONCLUSION

The NSAF & CNSAF are good methods for implementing acoustic echo cancellers because of its fast convergence rate. However, it requires a tradeoff between fast convergence rate and small steady-state MSE. This paper presented an improved adaptive

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combination of two NSAFs to solve this problem. For adaptation of component filters, in addition to the conventional decoupling update method, we also propose a coupling one, which can further improve the performance of the adaptive combination scheme. To verify the effectiveness of the proposed scheme, simulations using different input signals as well as system noises with different SNRs were performed. The experimental results demonstrated that the proposed scheme can obtain improved performance as compared to the CNSAF.

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