International Journal of Electronics and Electrical Engineering
ISSN : 2277-7040 Volume 2 Issue 9 (September 2012)
http://www.ijecee.com/ https://sites.google.com/site/ijeceejournal/

ECHO CANCELLATION BY ADAPTIVE COMBINATION OF NORMALIZED SUB BAND ADAPTIVE FILTERS

Rekha Saroha[1], Nirpender Singh[2], Rohit Anand[3]

[1] Student at Electronics and Communication Engineering,NCCE ,Panipat,INDIA

[2] Assistant Professor,Electronics and Communication Engineering, KITM, Karnal,INDIA

[3] Assistant Professor,Electronics and Communication Engineering,NCCE,Panipat,INDIA

saroharekha@yahoo.com,nirpender@gmail.com, roh_anand@rediffmail.com

Abstract. In acoustic echo cancellation the highly correlated speech input signal and very large impulse response path of echo signal will slow down the convergence rate of adaptive filters if full band adaptive filter is used. To solve these problems sub band adaptive filters are used. Adaptive combination methods provide an interesting way to improve adaptive filter's performance. Adaptive combination of normalized sub band adaptive filters is used. In our proposed method, adaptive combination of sub band signals before going to the adaptive filters. Experimental results show that the combination method can obtain both fast convergence rate and small steady state MSE by using less number of adaptive filters by adjusting step size parameter and mixing parameter.

Keywords: Round Trip Delay (RTD), Acoustic Echo Cancellation (AEC), Normalized Least Mean Square Algorithm (NLMS),Mean Square Error (MSE), Sub band Adaptive Filters (SAF), Adaptive Combination Normalized Sub band Adaptive Filters (ACNSAF)

International Journal of Electronics and Electrical Engineering
ISSN : 2277-7040 Volume 2 Issue 9 (September 2012)
http://www.ijecee.com/ https://sites.google.com/site/ijeceejournal/

1 Introduction

The history of echo cancellation begins on 10th July 1962.In telephones and teleconferencing; a reflection can occur where there is an impedance mismatch. If the reflected signal reaches the far end subscriber with a RTD of a few milliseconds then it is perceived as reverberator. If RTD exceeds a few tens of milliseconds the reflection is known as distinct echo [1].

Echo suppressor was used to remove echo which introduces a very large transmission loss in return path. A new technique that did not interrupt the echo return path called echo cancellation. The AEC estimates the characteristics of the echo path and generates a replica of the echo. The echo is then subtracted from the received signal. Adaptive filters are used as echo canceller. The normalized least-mean-square (NLMS) algorithm is one of the most popular adaptive filters. Speech input signal of the adaptive filter[26] is highly correlated and the impulse response of the acoustic echo path is very long. These two characteristics will slow down the convergence rate of the acoustic echo canceller. So sub band adaptive filtering is used to solve these problems. More recently, a combination of SAFs for AEC has been proposed [10], which is based on a conventional sub band structure. In this paper we propose a new scheme for adaptive combination of sub band adaptive filters deal with the tradeoff problem encountered in AEC which are implemented by NSAFs.

Section II represents the Simulink model for full band adaptive filter acts as acoustic echo cancellation [2].Section III represents sub band adaptive filter and adaptive combination of normalized sub band adaptive filters. Section IV improved adaptive combination of normalized sub band adaptive filters. Section v represents Experiment results.

2.SIMULINK MODEL OF FULLBAND ADAPTIVE FILTER ACTS AS ACOUSTIC ECHO CANCELLATION

Speech signal originating from loudspeaker is received by microphone passing through acoustic echo path. The acoustic echo is removed by adaptive filters. The d

(n) signal contains the speech signal and noise signal. The goal of the adaptive filter **w** (n) is to produce a replica of the echo signal *y* (*n*). *y* (*n*) can be used to cancel the echo by subtracting it from the microphone signal *d*(*n*) resulting in error free signal e(n)[3].

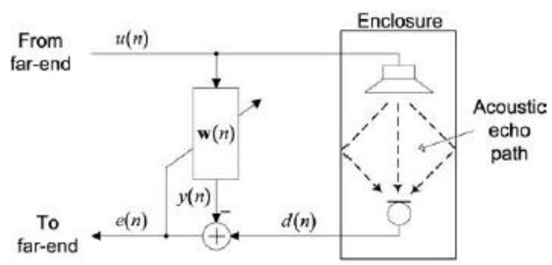

Fig. 1 : Block diagram of full band adaptive filter acts as AEC

Algorithm for AEC is as follows [2]:

Adjustable tap weights and input signal respectively can be expressed as:

$\mathrm{w(n)} = [\mathrm{w_0\ (n), w_1\ (n), \ldots . w}_{M-1}\ (n)]^T = \mathrm{w}^T\ (n)\mathrm{u(n)}$ ..(1)

U (n) = $[\mathrm{u(n), u(n-1), \ldots u(n-M+1)}]^T$ ….(2)

The output signal y (n) of adaptive filter is the multiplication of w(n) and u(n).

Y (n) = $\sum_{\mathrm{m=0}}^{M-1} w_m\ \mathrm{(n)u(n-m)} = \mathrm{w}^T\ \mathrm{(n)u(n)}$….(3)

e (n)=d(n)- $\mathrm{w}^T\ (n)$u(n)……..(4)

3. SIMULINK MODEL OF NSAF AND ITS ADAPTIVE COMBINATION

3.1 Simulink Model of NSAF

The input speech signal u (n) and desired output d(n) are decomposed into N spectral bands using analysis filters. Analysis filtering is then performed in these sub-bands by a set of independent filters (h0(n), h1(n),…, hM-1(n))[2].The sub band signals are further processed by individual adaptive sub filters Wi(z).Each sub band is computing error signal e(n). By updating the tap weights, minimizes the sub band error signal. The full band error signal *e (n)* is finally obtained by interpolating and recombining all the sub band error signals using a synthesis filter bank.

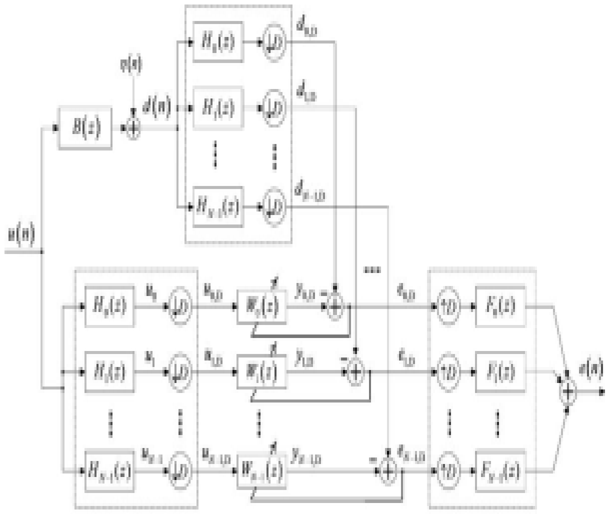

Fig. 2 : Block diagram of NSAF

3.2 Simulink Model of Adaptive Combination of NSAFs

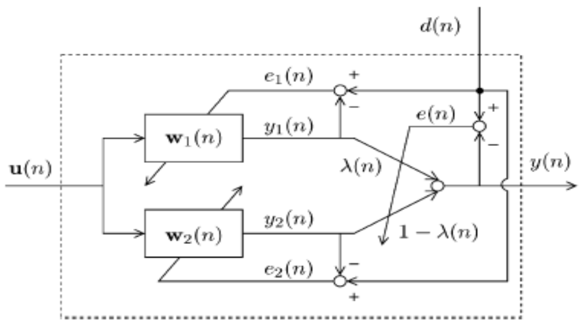

Fig. 3 : Adaptive combination of NSAFs (ACNSAF)

A large step size yields a fast convergence rate but also a large steady state MSE [7].To achieve fast convergence rate and small steady state MSE[19-21],adaptive combination of sub band adaptive filters is done .So that large step sizes adaptive filters give fast convergence rate and small step sizes adaptive filters give small steady state MSE. So idea becomes to adapt different step sizes filters independently and combination is carried out by using a mixing parameter lambda.

$y_1(n) = w_1^T(n)u_n$,

$y_2(n) = w_2^T(n)u_n$.

Consider $\mu_1 > \mu_2$, then $w_1(n)$ adaptive filter has faster convergence rate and large steady state MSE whereas $w_2(n)$ has slower faster convergence rate but small steady state MSE. So our purpose is to get large convergence rate and small steady state MSE, So combine both adaptive filters. The output of overall filter is:

$Y(n)=\lambda(n)\, y_1(n)+[1-\lambda(n)]\, y_2(n)$

where **λ** is mixing parameter.

The overall filter with tap weight factor of the form is:

$w(n)=\lambda(n) w_1(n)+ [1-\lambda(n)]\, w_2(n)$.

For adaptation of mixing parameter $\lambda(n)$,use stocastic gradient method to minimize error of overall filter $e^2(n) = [d(n) - y(n)]^2$.

However instead of directly adjusting (n),we will adapt a variable $\alpha(n)$ that defines **λ**(n) as a sigmoidal function

$(n)=sgm[\alpha(n)]=\frac{1}{1+e^{-\alpha(n)}}$

The update eq. for α(n) is given as:

$\alpha(n+1)=\alpha(n)+\mu e(n)[y_1(n)- y_2(n)]\boldsymbol{\lambda}(n)[1-\boldsymbol{\lambda}(n)]$

4. IMPROVED ADAPTIVE COMBINATION OF NORMALIZED SUBBAND ADAPTIVE FILTERS

A large convergence rate and small steady state MSE can be greatly achieved by using less amount of adaptive filters [25] as comparison to the previous adaptive combination method. This can be achieved by using the idea of adaptive combination of speech input signal before going to the NSAF.

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 5000 samples for Gaussian noise and sine wave input, respective 12*104 samples in the case of speech input. A reverberating effect was added to the input by an artificial Schroeder I reverberator which contained four comb filters in parallel and two all-passes filters series connected. The first 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 no.5(a)- 5(d) shows the Matlab simulink results of various conventional methods based on ERLE performances as shown below.

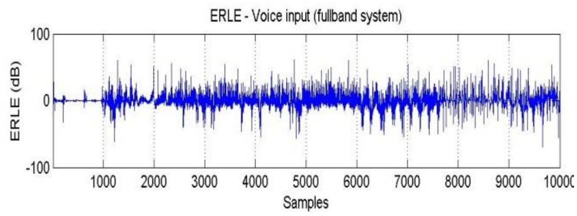

Fig no. 5 (a): Measured performance ERLE for full band system

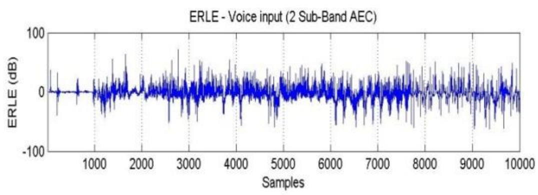

Fig no.5(b) : Measured performance ERLE for subband system

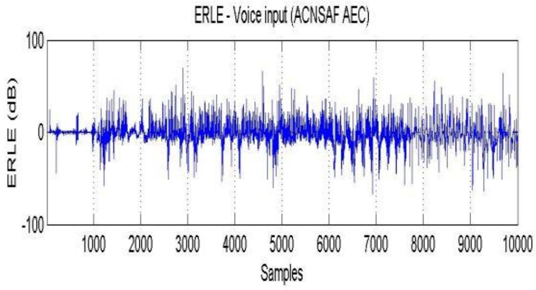

Fig no. 5(c) : Measured performance ERLE for ACNSAF

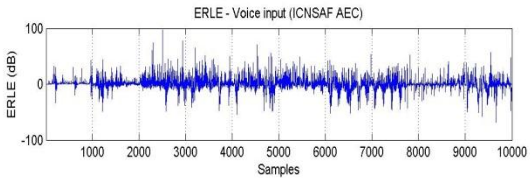

Fig. no. 5(d) : Measured performance ERLE for IACNSAF

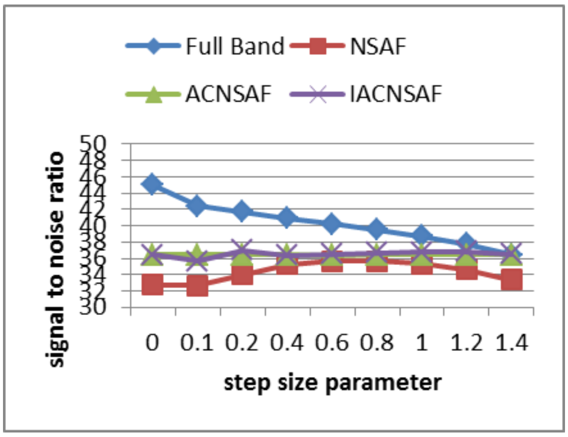

Fig no. 6 : Conventional Full band, sub band, ACNSAF,IACNSAF by varying Step size parameter

Figure no. 6 shows the graphical results of conventional full band, sub band, adaptive combination of NSAF and improved ACNSAF. It is clear from the figure that the conventional NSAF must carry out a tradeoff between fast convergence rate and small steady-state MSE by selection of the value of the step-size, whereas the

ACNSAF and IACNSAF exhibit both fast convergence rate and small steady-state MSE. In addition, one can see that the IACNSAF has a better behavior in the initial convergence phase than the ACNSAF. The performance of the ACNSAF is still better that that of the conventional NSAF, and is still not as good as that of the IACNSAF.

6. CONCLUSION

The NSAF is a good candidate 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 adaptive convex combination of two NSAFs to solve this problem. In addition to the conventional coupling update method for component filters, we also proposed a coupling update mechanism which requires less number of adaptive filters as than used in conventional method. 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 conventional ACNSAF.

7. REFERENCES

[1]M. M. Sondhi, “The history of echo cancellation,” IEEE Signal Process.Mag., vol. 23, no. 5, pp. 95–98, Sep. 2006.

[2] K. A. Lee, W. S. Gan, and S. M. Kuo, Subband Adaptive Filtering: Theory and Implementation. Hoboken, NJ: Wiley, 2009.

[3] Jingen Ni and Feng Li ,”adaptive combination of subband adaptive filters for acoustic echo cancellation.” IEEE Transactions on Consumer Electronics, Vol. 56, No. 3, August 2010

[4] Irina Dornean, Marina Topa, Botond Sandor Kirei, Marius Neag, “Sub-band adaptive filtering for acoustic echo cancellation.”2009.

[5] B. Farhang-Boroujeny, Adaptive Filters: Theory and Applications, New York: John Wiley & Sons, Inc., 1998.

[6] J. J. Shynk, ‘Frequency domain and multirate adaptive filtering filtering’, IEEE Signal Processing Mag., 9, January 1992, 14–37.

[7] N. J. Bershad, J. C. M. Bermudez and J.-Y. Tourneret, “An affine combination of two LMS adaptive filtersTransient mean-square analysis,” IEEE Trans. Signal Process., vol. 56, no. 5, pp. 1853–1864, May 2008.

[8] J. Arenas-García, M. Martínez-Ramón, A. Navia-Vázquez, and A. R. Figueiras-Vidal, “Plant identification via adaptive combination of transversal filters,” Signal Process., vol. 86, pp. 2430–2438, 2006.

[9] J. Arenas-Garcia, A. R. Figueiras-Vidal, and A. H. Sayed, “Mean-square performance of a convex combination of two adaptive filters,” IEEE Trans. Signal Process., vol. 54, no. 3, pp. 1078–1090, Mar. 2006.

[10] L. A. Azpicueta-Ruiz, A. R. Figueiras-Vidal, and J. Arenas-García, “Acoustic echo cancellation in frequency domain using combinations of filters,” in 19th Int. Congress on Acoustics (ICA), Madrid, Sep. 2007.

[11] J. Arenas-García, V. Gómez-Verdejo, M. Martínez-Ramón, and A. R. Figueiras-Vidal, “Separate-variable adaptive combination of LMS adaptive filters for plant identification,” in Proc. 13th IEEE Int. Workshop Neural Networks Signal Processing, Toulouse, France, 2003, pp. 239–248.

[12] J. Arenas-García, M. Martínez-Ramón, V. Gómez-Verdejo, and A. R. Figueiras-Vidal, “Multiple plant identifier via adaptive LMS convex combination,” in Proc. IEEE Int. Symp. Intel. Signal Processing, Budapest, Hungary, 2003, pp. 137–142.

[13] A. H. Sayed, Fundamentals of Adaptive Filtering. New York: Wiley,2003.

[14] N. J. Bershad, J. C. M. Bermudez and J.-Y. Tourneret, “An affine combination of two LMS adaptive filters —Transient mean-square analysis,” IEEE Trans. Signal Process., vol. 56, no. 5, pp. 1853–1864, May 2008.

[15] K. A. Lee and W. S. Gan, “Improving convergence of the NLMS algorithm using constrained subband updates,” IEEE Signal Process. Lett., vol. 11, no. 9, pp. 736–739 , Sep. 2004.

[16] P. P. Vaidyanathan, Multirate Systems and Filter Banks. Englewood Cliffs, NJ: Prentice-Hall, 1993.

[17] S. Haykin, Adaptive Filter Theory. Upper Saddle River, NJ: Prentice- Hall, 2002.

[18] Sang-Wook Sohn; Jae-Jun Yun; Jeongkyu Lee; Hyeon-Deok Bae; Hun Choi "convex combination of subband adaptive filters for sparse impulse response system" Circuit and systems(MWSCAS),2011,IEEE 54th international Midwest symposium ,2011.

[19] M. S. E. Abadi and J. H. Husøy, "Selective partial update and setmembershi subband adaptive filters," *Signal Process*ing, vol. 88, no. 10, pp. 2463–2471, Oct. 2008.

[20] J. Ni and F. Li, "A variable regularization matrix normalized subband filter," *IEEE Signal Process. Lett*., vol. 16, no. 2, pp. 105–108 Feb. 2009.

[21] J. Ni and F. Li, "A variable step-size matrix normalized subband adaptive filter," *IEEE Trans. Audio, Speech, Lang. Process*., to be published, DOI: 10.1109/TASL.2009.2032948

[22] I. Dornean, M. Topa, B.S. Kirei, G. Oltean, HDL Implementation of the Variable Step Size N-LMS Adaptive Algorithm", Proc. of 2008 IEEE International Conference on Automation, Quality and Testing,Robotics, Cluj-Napoca, Romania, May 22-25, 2008, Vol III, pp. 243- 246.

[23] Trump T."A combination of two NLMS filters in an adaptive line enhancer"digital signal processing,2011 17th international conference on Digital Object Identifier.

[24] Wige, E.; Yammine, G.; Amon, P.; Hutter, A.; Kaup, A."Adaptive in-loop noise-filtered prediction for high efficiency video coding "2011 IEEE.

[25] Kozat, S.S.; Erdogan, A.T.; Singer, A.C.; Sayed, A.H."Transient Analysis of Adaptive Affine combinations"signal processing ,IEEE Transection,2011.Volume: 59 , issue 12.

[26] Elangovan,K."Comparative study on the channel estimation for OFDM system using LMS,NLMS, and RLS algorithm" PRIME,2012.