Subband and MSF Performance Comparison for AEC

O.P. Sahu[1], Sanjeev Kumar Dhull[2], and Sandeep K. Arya[2]

[1] Department of ECE, National Institute of Technology, Kurukshetra
opsahu_2011@yahoo.com
[2] Department of ECE Guru Jambheshwar University of Science and Technology
Hisarsanjeev_dhull_ap@yahoo.co.in, arya1sandeep@rediffmail.com

Abstract. We have designed and simulated two techniques for acoustic echo cancellation. These systems are based upon a least-mean-square (LMS) adaptive algorithm and uses multi sub and sub band technique. A comparative study of both methods has been carried out.

Keywords: LMS, Multiple subfilter, Sub band**,** AEC, ERLE.

1 Introduction

Acoustic echo cancellation (AEC) [1] is used in teleconferencing and its purpose is to provide high quality full-duplex communication. The main part of an AEC is an adaptive filter which estimates the impulse response of the loudspeaker-enclosure-microphone (LEM)[2] system. There are various adaptive algorithms for the AEC filter update, these are the least mean square, normalized least mean square (LMS, NLMS), affine projection (AP) [3][4]and recursive least squares (RLS) algorithms. As the echo cancellation environment is not stationary therefore echo reduction in rooms with long reverberation time is necessary. Hence, the signal processing methods are in demand in industry. The technique used in earlier stages was echo suppression .Due to some disadvantages of echo suppression echo cancellation came into picture and the process of Acoustic echo cancellation [15] is achieved with the help of adaptive filter which models the LEM system. The purpose of an acoustic echo-canceller is to reduce the amount of sound which a far-end teleconference transmits from returning to them. This paper is organized in four sections. Section two describes the simulation model of AEC in matlab using sub band and msf approach. Further, section three discusses the results. In the end section four concludes the paper.

2 Subband and Multiple Sub Filters

In order to obtain a full-duplex hands-free communication, in for instance a car, it is necessary to perform an acoustic echo cancellation of the far-end speaker. The echo cancellation must be adaptive and follow variations in the acoustic channel. The filter length of the acoustic canceller can be typically between 500-1500 FIR taps. Filter

V.V. Das, E. Ariwa, and S.B. Rahayu (Eds.): SPIT 2011, LNICST 62, pp. 236–242, 2012.

lengths of these sizes gives a large computational burden even with a simple adaptive Filter algorithm such as LMS[6]. These filters also suffer from long convergence time, especially if the reference signal spectrum has a large dynamic range i.e. a large eigen value spread in the corresponding signal covariance matrix. Sub-band techniques give a twofold advantage: the computational burden is essentially reduced by the number of sub-bands[7] and it is also possible to get a faster convergence because the spectral dynamic range in each sub-band will be smaller. In this paper we present an implementation of AEC using sub-band adaptive Filter and multiple sub filter methods.

2.1 Analysis of Sub-Band and MSF Adaptive Filters

The delay less attribute of this technique comes from the fact that the new adaptive weights are computed in sub-bands and then transformed to an equivalent full-band filter with means of an FFT. The filter works in real time on the loudspeaker signal. The coefficients are calculated separately in each band. They can be calculated either by employing the error signal $\epsilon(k)$ or the microphone input signal $d\ (k)$. If the signal $d\ (k)$ is used a local error signal in each band must be created and the calculations must not be performed in real time. This will however give somewhat lower suppression because the algorithm is blind towards the real error signal. The full band signal is divided into several sub-bands signals by using a polyphase FFT technique. The outputs from the sub-band filters are only down sampled by a factor D=M/2. This means that even sub-bands are centered at dc while odd sub-bands are centered at one half of the decimated sampling frequency, see in Fig.2. This fact must be considered in the poly phase filter bank. Since, we only consider full-band filters with real coefficients; it is enough to calculate M/2 complex sub-band signals. The rest can be found by utilizing the complex conjugate symmetry. If we have a N tap full-band filter, the filter length in each sub-band will be N/D. A N/D points FFT will be calculated on the adaptive weights in each sub-band [11][15]. These are then stacked to form a 0-(N/2-1) point array. The array is then completed by setting point N to zero and using the complex conjugate of points 1-(N/2-1) in reverse order. Finally, the N point array is transformed by N point inverse FFT to obtain the full-band filter weights.

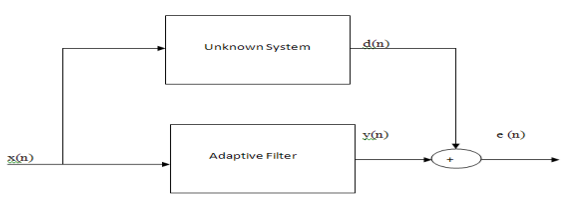

Fig. 1. Full Band version of Adaptive Filter

Fig.1 depicts the full-band version of an identification system, where *x (n)* represents the input data, which is common to both unknown system and adaptive filter. The desired signal *d (n)* contributes to the error minimization by subtracting from it the output of the adaptive filter *y (n)*. Adaptive identification is a procedure that learns more about the model as long as a new pair of measurements is received, updating the knowledge in order to incorporate the newly received information. The error signal *e (n)* ideally should be equal to the near-end signal *x(n)*. Classical full-band cancellers are unattractive for real time processing and their computational requirements exceed the capabilities of present day DSPs. Fig.2 depicts the sub-band adaptive filtering (SAF) for *M* sub-bands. Using analysis filter banks, the original signal isdecomposed into *M* signals (*x0(n), x1(n)…xM-1(n)*) bysubdividing its spectra. Adaptive filtering[19] is then performedin these sub-bands by a set of independent filters (*h0(n)* , *h1(n),…, hM-1(n)*). The outputs of these filters are subsequentlycombined using a synthesis filter bank to reconstruct the full bandoutput.

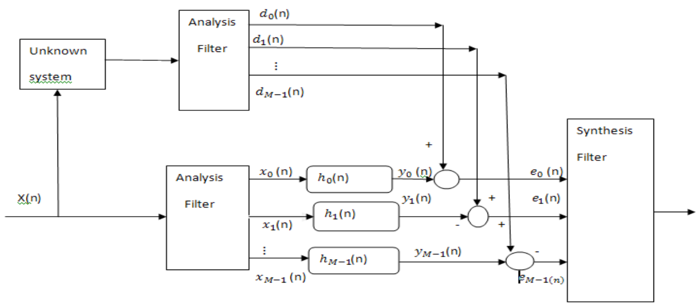

Fig. 2. Sub-band system

To ease the processing, down-sampling (L↓) and up-sampling (L↑) can be inserted between the analysis and synthesis filter banks[11]. The analysis(Fig.3) filter bank design problem reduces to the design of a single prototype non-recursive filter *P(z)*, the analysis filters being modulated versions of the prototype. Ideal analysis filters are band-pass filters with normalized centre frequencies $\omega m = 2\ m/M$, $m = 0\ldots M-1$, and with bandwidth 2 / M.

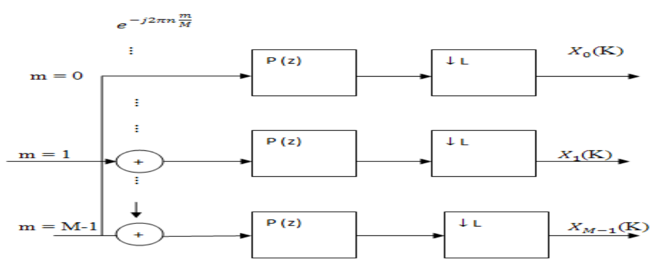

Fig. 3. Analysis filter

The ideal filters have unit magnitude and zero-phase in the pass-band while the stop-band magnitude is zero. While zero phase filters involve non causality, the requirements need to be relaxed by using linear phase filters. The choice is to use FIR filters that have linear phase, but not ideal magnitude requirements. This approximation leads to aliasing effects. The design of synthesis Fig.4 filter banks reduces also to the design of a single synthesis prototype filter *Q (z)*. When designing the synthesis filter bank, the focus is on the performance of the analysis-synthesis filter bank as a whole.

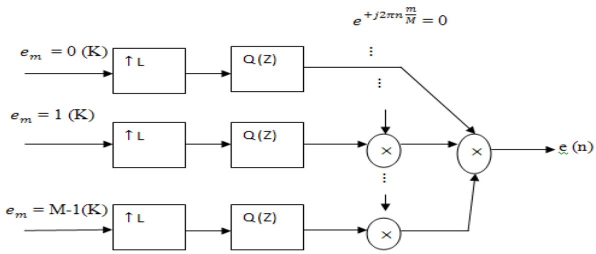

Fig. 4. Synthesis filter

Achieving zero residual error (no alias effect) requires the sub-band filters and sub-band models to have an infinite tap size. Since we always use FIR sub-band filters and sub-band models, residual errors are unavoidable. This implies that in the design of a sub-band identification system, there is a tradeoff between asymptotic residual error and computational cost.

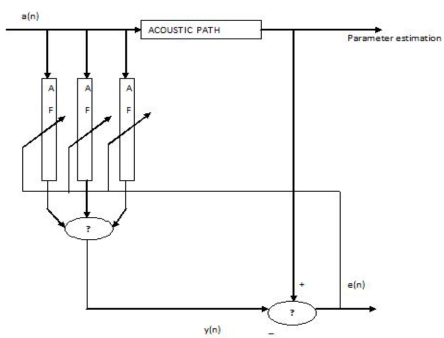

Fig. 5. Multiple sub filter echo cancellation

As shown in Figure.5 we can do echo cancellation by multiple sub filter structures also.The aim of the paper is to compare these two techniques. The system capability can be represented by the output error, but for accurate meaure Echo Return Loss Enhancement (ERLE) is the basic formula to compute the performance; it is defined as the ratio of the power of the desired signal over the power of the residual signal:

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

3 Simulation Results

The next work is to do simulations of multiple filter and sub band model. For simulations comparisons we are taking far end and near end speech signals. ERLE is the main comparison parameters for this approach. Simulations are carried by estimating correct values of time delay, gain and step size and order of filter. We have calculated true estimate of time delay using GAE algorithm and comparing the results as shown in different figures. We have plotted different ERLE graphs for full band.MSF and Sub band strucuture.The filter length required in case of single full band is 1075. Filter length required in MSF is 194 where as it has been reduced to 8 in case of sub band structure.

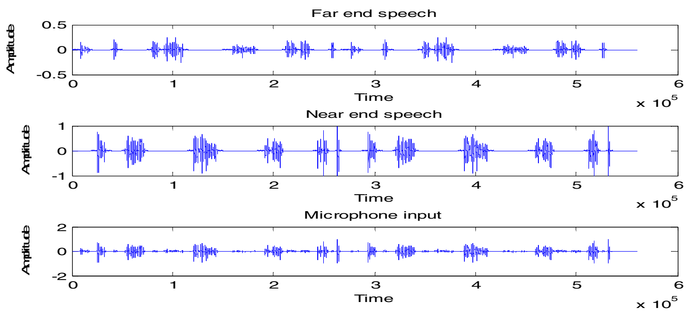

Fig. 6. Far end and Near End Speech signal

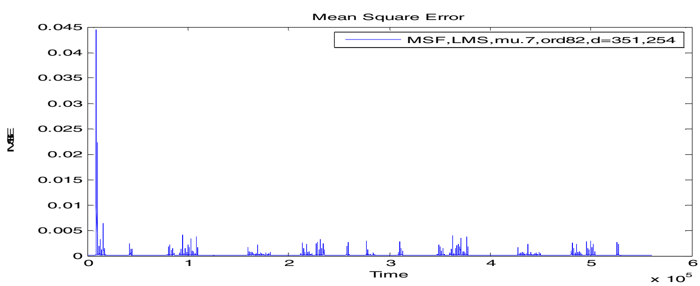

Fig. 7. MSE of Multiple filter approach (LMS)

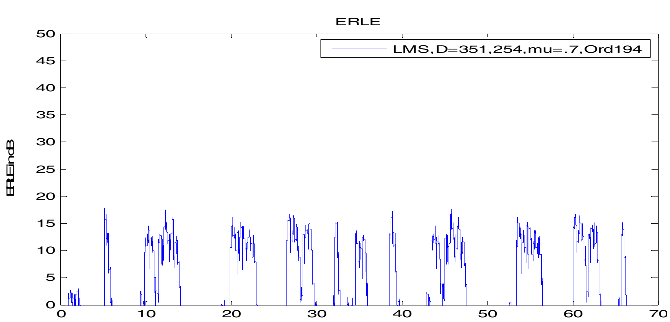

Fig. 8. ERLE for MSF using LMS

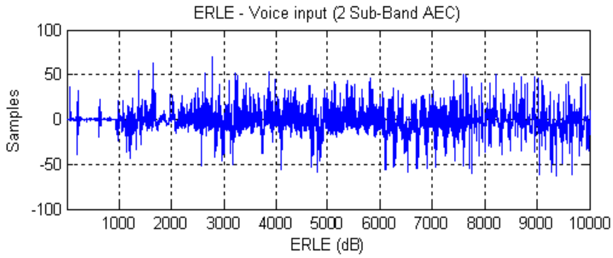

Fig. 9. ERLE for two subband

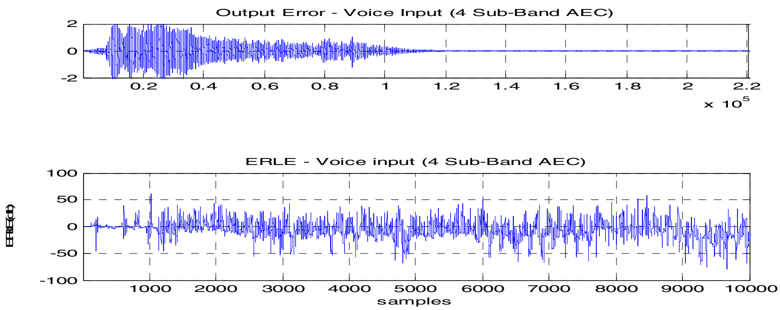

Fig. 10. ERLE for 4 Subband

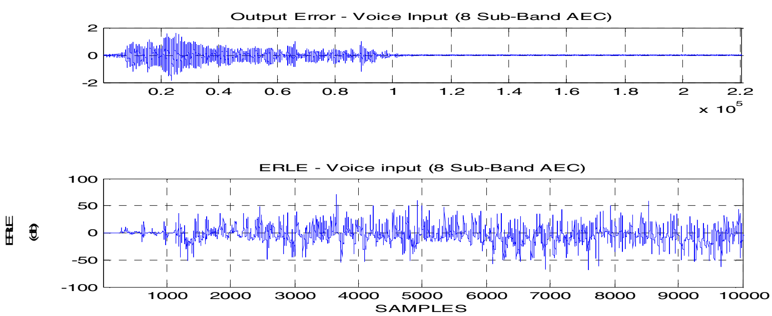

Fig. 11. ERLE for 8 subband

4 Conclusion

No doubt multiple sub filter converge faster than full band but Suband design of echo canceller converge faster than multiple sub filter design and filter length required in subband is also less as compared to multiple subfilter design.

References

1. Bershad, N.J., Bermudez, J.C.M., Tourneret, J.-Y.: An affine combination of two LMS adaptive filters —Transient mean-square analysis. IEEE Trans. Signal Process. 56(5), 1853–1864 (2008)
2. Petraglia, M.R., Batalheiro, P.B.: Non uniform sub band adaptive filtering with critical sampling. IEEE Trans. Signal Process. 56(2), 565–575 (2008)
3. Sayed, A.H.: Adaptive Filters. Wiley, New York (2008)
4. Ding, H.: Fast affine projection adaptation algorithms with stable and robust symmetric linear system solvers. IEEE Trans. Signal Process. 55(5), 1730–1740 (2007)
5. Ni, J., Li, F.: A variable regularization matrix normalized sub band adaptive filter. IEEE Signal Process. 16(2), 105–108 (2009)
6. Lee, K.A., Gan, W.S., Kuo, S.M.: Sub band Adaptive Filtering: Theory and Implementation. Wiley, Hoboken (2009)
7. Lee, K.A., Gan, W.S.: Inherent décor-relating and leastperturbation properties of the normalized sub band adaptive filter. IEEE Trans. Signal Process. 54(11), 4475–4480 (2006)
8. Arenas-Garcia, J., Figueira-Vidal, A.R., Sayed, A.H.: Mean-square performance of a convex combination of two adaptive filters. IEEE Trans. Signal Process. 54(3), 1078–1090 (2006)
9. Sondhi, M.M.: The history of echo cancellation. IEEE Signal Process. Mag. 23(5), 95–98 (2008)
10. Kao, C.C.: Design of echo cancellation and noise elimination for speech enhancement. IEEE Trans. Consumer Electronics 49(4), 1468–1473 (2003)
11. Courville, M.D., Duhamel, P.: Adaptive filtering in sub bands using a weighted criterion. IEEE Trans. Signal Process. 46(9), 2359–2371 (1998)
12. Pradhan, S.S., Reddy, V.U.: A new approach to sub band adaptive filtering. IEEE Trans. Signal Process. 47(3), 655–664 (1999)
13. Lee, K.A., Gan, W.S.: Improving convergence of the NLMS algorithm using constrained sub band updates. IEEE Signal Process. Lett. 11(9), 736–739 (2004)
14. Azpicueta-Ruiz, L.A., Figueiras-Vidal, A.R., Arenas-García, J.: A.:Acoustic echo cancellation in frequency domain using combinations of filters. In: 19th Int. Congress on Acoustics (ICA), Madrid (September 2007)
15. Ohno, S., Sakai, H.: Convergence Behavior of the LMS algorithm in sub band adaptive filtering. Signal Processing 81, 1053–1059 (2001) ISBN 0165-1684
16. Brennan, R., Schneiderm: A flexible filter bank Structure for extensive signal manipulation in digital Hearing aids. In: Proc. IEEE Int. Symp. Circuits and Systems, pp. 569–572 (1998)
17. Chao, J., Kawabe, S., Tsujii, S.: A new IIR adaptive echo canceller. IEEE Journal on Selc. Areas in Corn. 12, 1530–1539 (1994)
18. Widrow, B., Stearns, S.: Adaptive Signal Processing. Prentice-Hall, Englewood Cliffs (1985)
19. Kellermann, W.: Analysis and design of multirate systems for cancellation of acoustical echoes. In: Proc. Int. Conf. Acoust., Speech, Signal Proc., New York, pp. 2570–2573 (1988)
20. Gilloire, A., Vetterli, M.: Adaptive filtering in sub-bands. In: Proc. Int. Conf. Acoustic. Speech, Signal Proc., New York, NY (April 1988)
21. Irina, D.: Marina.:Subabd Adaptive Filtering, pp. 810–813. IEEE (2009)