International Journal of Recent Research and Review, Vol. VII, Issue 2, June 2014

ISSN 2277 – 8322

Subband Adaptive Filter Architecture with Low Convergence

Khet Singh

Electronics and communication Department, University College of Engineering,
Rajasthan Technical University, Kota (Rajasthan)

E-mail: ket.vcet@gmail.com

***Abstract*- Subband Adaptive Filter Architecture techniques have been recently used for subband adaptive filters, since some of the applications such as acoustic echo cancellation and wideband active noise control need adaptive filters with thousands of taps, which result in high computational complexity and low convergence rate. By the use of subband adaptive algorithms, the computational complexity may be reduced along with convergence rate; however, a delay is introduced in the signal path. To remove the delay, the delayless subband adaptive filter architecture open loop structures have been introduced. This paper presents a new open loop delayless subband adaptive filter structure and closed loop suband adaptive filter , where the performance concerning the mean square error of the subband adaptive algorithm, caused due to the aliasing existing in the subband structure, is superior to the results obtained up to now for open loop delayless structures.In applications like echo sound cancellation and speech enhancement, where there is need to track continuously, adaptive filtering is usually used. Long adaptive filters gives problems like low convergence and high complexity. Subband adaptive filtering has been introduced to overcome these problems. The filter banks used in subband adaptive filtering introduce large delays. In order to compensate for the delays, delayless subband adaptive filtering is introduced. Delayless subband adaptive filtering is used in both open loop and closed loop configuration, where the subband filters are transformed to a fullband filter using a weight.**

***Keywords* - convergence, least mean square, inter symbol interference, nlms, critically subband sampled.**

I. INTRODUCTION

Subband adaptive filtering is rapidly becoming one of the most effective techniques for reducing computational complexity and improving the convergence rate of algorithms in adaptive signal processing applications. Discuss the basic principles that underlie the design and implementation of subband adaptive filters. Comprehensive coverage of recent developments, such as multiband tap–weight adaptation, delayless architectures, and filter–bank design methods for reducing band–edge effects are included. Several analysis techniques and complexity evaluation are also introduced in this report to provide better understanding of subband adaptive filtering [1].

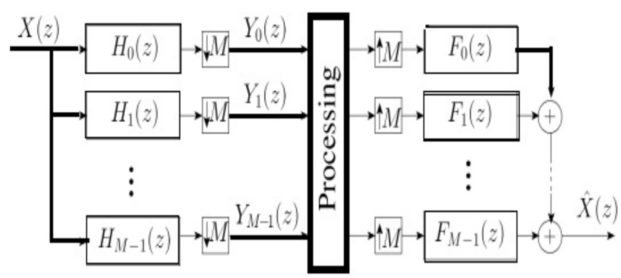

Fig.1. Subband adaptive filtering

Adaptive filtering is a widespread technique in many applications. For acoustic echo cancellation (AEC) hands-free telephony very large adaptive filters are used in system identification on text, whereas in digital communications, adaptive filters perform the channel distortion equalization [2]. The present need for increased throughput in new systems also results in an increase of the equalizer length. In these two areas, there is a demand for efficient and low complexity algorithms.

II. DELAYLESS SUBBAND ADAPTIVE FILTERING WITH SUBBAND TO FULLBAND TRANSFORM

As suband adaptive filters in general, delayless subband adaptive filters consist of two main parts, a filtering operation and a coefficient adaptation operation. The filtering operationis performed by a fullband filter in the time-domain, see fig. 2. In a practical implementation, a long fullband filter has high computational complexity. In some icases it may be preferable to implement the fullband filter partially in the frequency doman [3].

$$y(n) = f^T([n/D])x(n) \quad (1)$$

where $\mathbf{f}(k) = [f0(k), \ldots, f_{Lf-1}(k)]^T$ is a vector containing fullband filter coefficients at time instant k. The variable k denotes the subband signal time index which is related to the full rate time index n according to k=∟(n/D) where [.] denotes round-off towards the closest integer towards minus infinity. Vector $\mathbf{x}(n) = [x(n), \ldots, x(n - L_{f+1})]^T$ is the input signal vector. The fullband filter length is denoted by Lf . A fullband error signal $e(n)$ is obtained as[1]

$$e(n) = d(n) - y(n) \quad (2)$$

Consider a filter bank with M subbands, causal FIR analysis filters $h_m(n)$ of length L_h, and decimators with decimation rate D. Input signal $x(n)$ is decomposed into subband signal $x_m(k)$ according to

$$x_m(k) = h^T_m x(kD). \quad m= 0\ldots..M-1 \quad (3)$$

where, $hm = [h_{m,0}, \ldots, hm,L_{h-1}]^T$ and $x(kD)$ is an input signal vector of corresponding length.

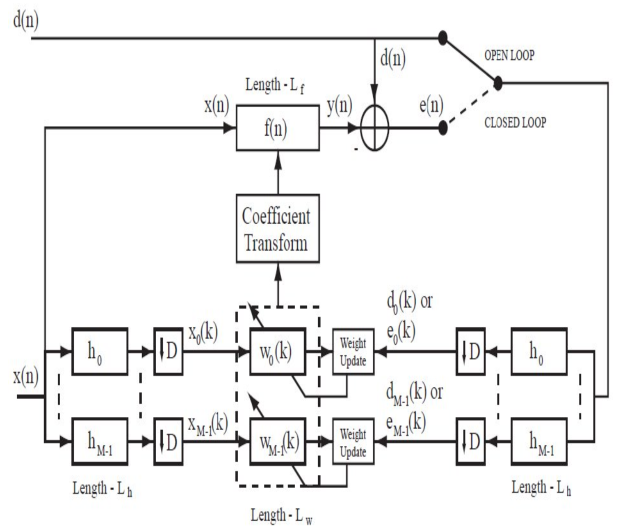

Fig.2 Delayless subband adaptive filters with open loop and closed loop configurations

The delayless subband adaptive filter has two operation configurations, open loop and closed loop. In the open loop configuration, the desired signal is decomposed into subband signals as [3]

$$d_m(k) = h^T_m d(kD) . \quad (4)$$

and subband error signals em(k) are obtained as

$$e(n) = d(n) - y(n) = d_m(k) - w^T_m(k)x_m(k) \quad (5)$$

where $wm(k) = [w_{m,0}(k), \ldots, w_{m,L_{w-1}}(k)]^T$ is a vector containing the adaptive filter coefficients at time instant k, and $x_m(k) = [x_m(k), \ldots, x_m(k - Lw + 1)]^T$ is an input subband signal vector. The length of the adaptive filters Lw is related to the full band filter length Lf and the decimation rate D as $L_w = Lf/D$. In the closed loop configuration, the subband error signals are obtained by decomposing the full band error signal into subband error signals [4].

$$e_m(k) = h^T_m e(kD) \quad (6)$$

In both configurations, a Normalized Least Mean Square (NLMS) algorithm is used in the subbands. Other adaptive algorithms can also be used, such as Recursive Least

$$w_m(k + 1) = w_m(k) + \mu_m(k)e_m(k)x^*_m(k). \quad (7)$$

The time-varying subband step size μ_m is calculated

$\mu_m(k) = \mu / P_m(k)$ (8)

where μ is a global step size and $Pm(k)$ is the short time subband power estimate of the mth subband signal $x_m(k)$

$$Pm(k) = \frac{x^H_m(k)x_m(k)}{Lw} \quad (9)$$

The fullband coefficients are obtained from the subband coefficients by means of a linear coefficient transform

$f(k) = Tw(k)$ (10)

where, $w(k) = [w^T_0(k), \ldots, w^T_{M-1}(k)]^T$ is a adaptive filter coefficient vector at time instant k, and matrix T is the subband-to-fullband transformThe open loop configuration is summarized by the equations for all n.

$y(n)=f^T([n/D])x(n)$ (11)
$x_m(k) = h^T_m x(kD)$ (12)
$d_m(k) = h^T_m d(kD)$ (13)
$y_m(k) = w^T_m(k)x_m(k)$ (14)
$e(n) = d(n) - y(n)$ (15)
$Pm(k) = \frac{x^H_m(k)x_m(k)}{Lw}$
$\mu_m(k) = \mu / P_m(k)$ (16)
$w_m(k+1) = w_m(k) + \mu_m(k)e_m(k)x^*_m(k)$. (17)
$f(k) = Tw(k)$ (18)

The closed loop configuration is summarized by the equations for all n.

$y(n)=f^T([n/D])x(n)$ (19)
$e(n) = d(n) - y(n)$ (20)
for all k = [n/D]
$x_m(k) = h^T_m x(kD)$ (21)
$e_m(k) = h^T_m e(kD)$ (22)
$Pm(k) = \frac{x^H_m(k)x_m(k)}{Lw}$ (23)
$\mu_m(k) = \mu / P_m(k)$ (24)
$w_m(k+1) = w_m(k) + \mu_m(k)e_m(k)x^*_m(k)$. (25)
$f(k+1) = Tw(k+1)$ (26)

subband-to-fullband transforms in combination with filter banks for the subband signal decompositions is presented.

III. FILTER BANK AND TRANSFORM CONFIGURATION

Uniform DFT Filter Banks

A uniform DFT Filter Bank consists of M analysis filters $H_m(z)$ (Fig. 3), which are modulated from a prototype filter H(z) according to[5].

$Hm(z)= h(zW^m_m)$

where $W_M = e^{-j2\pi/M}$. Let $Al(z)$, $l = 0, \ldots, M-1$, denote the polyphase components of the prototype filter $H(z)$

$$H(z) = \sum_{l=0}^{M-1} z^{-l} A_l(z^M). \quad (27)$$

Accordingly, the polyphase decomposition of all analysis filters is given[6]

$$H_m(z) = \sum_{l=0}^{M-1} z^{-l} A_l(z^M) W_M^{-ml}. \quad (28)$$

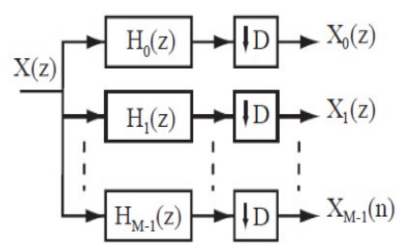

Fig.3 Direct Form Uniform-DFT Filter Bank

Analysis Filter Bank Design

In this section an analysis filter bank design procedure for the delayless subband adaptive filter is described. Lossless power complementary filter banks with minimum phase property is designed. The reason for the use of this design method is that the design procedure is applicable for both uniform-DFT (M-channel filter bank and tree structured filter banks (two-channel filter bank design). The power complementary constraint for an M-channel analysis filter bank is given by [7]

$$\sum_{m=0}^{M-1} \left|H_m(e^{j\omega})\right|^2 = 1,$$

Where, the analysis filters $H_m(z)$ are derived from a lowpass prototype analysis filter $H(z)$ with real coefficients according to $H_m(z) = H(zW_m^M)$. With the spectral factorization $Q(z) = H(z)H(z-1)$, where $Q(e^{j\omega})$ is a real-valued frequency function and $Q(e^{j\omega}) \geq 0$ [22], the power complementary constraint can be rewritten

$$\sum_{m=0}^{M-1} Q(e^{j\omega}W_M^m) = 1. \tag{29}$$

The transfer function $Q(z)$ is a zero phase FIR filter with coefficients qi [8]

$$Q(z) = \sum_{i=-L+1}^{L-1} q_i z^{-i}. \tag{30}$$

The power complementary constraint can be transformed to the time-domain according.

$$\sum_{m=0}^{M-1} q_i W_M^{-mi} = \delta_i, \tag{31}$$

where δi = 1 for $i = 0$ and zero otherwise. Since

$$\sum_{m=0}^{M-1} W_M^{-mi} = \begin{cases} M & \text{for } i = 0, \pm M, \pm 2M, \ldots \\ 0 & \text{otherwise} \end{cases}$$

IV. SIMULATED AND MEASURED RESULT

Open Loop delayless subband filter result

The identification of a length *Np* = 512 FIR system is considered. The input signal is a colored noise sequence generated by passing gaussian white noise by a first-order IIR filter with a pole located at $z = 0.9$. Experiments open loop delayless filter were performed with the fullband normalized LMS, and with the delayless subband structure of fig. 2 using perfect reconstruction cosine modulated filter banks and M = 2 subbands. The step-sizes were selected such that the best convergence rate μ= 0.1 presents the MSE evolutions. The new delayless subband structure presents a better convergence rate than the LMS algorithm, due to the power normalization of the step-sizes. It converges to an MSE of the order of the stop band attenuation of the analysis filter (which is around -18 db for $M = 4$, -16 dB due to the assumption of non overlapping non-adjacent analysis filters Table I [9].

Table I

Np	M	μ	dB
512	2	0.1	-16
512	4	0.1	-15

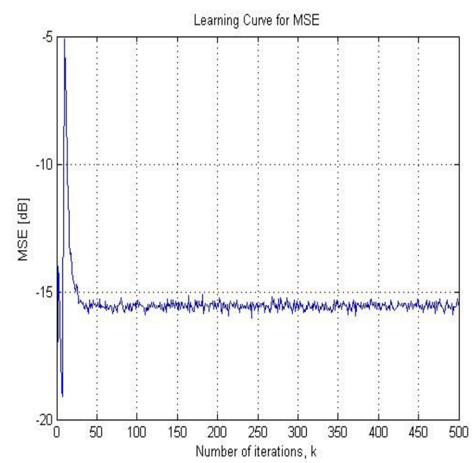

Fig.4 Simulation Result of open loop system z=. 9, M=2

Closed Loop Delayless Subband Adaptive result

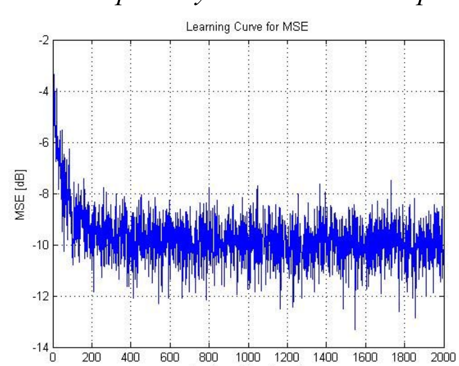

Fig. 5 Simulation Result of closed loop system M=8 Np=64

Table II

Np	M	μ	dB
64	8	0.1	-12

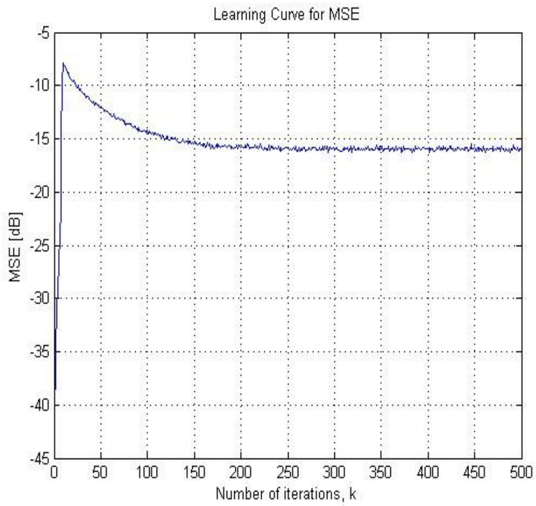

Fig.6 Simulation Result of open loop system z= .9 M=4

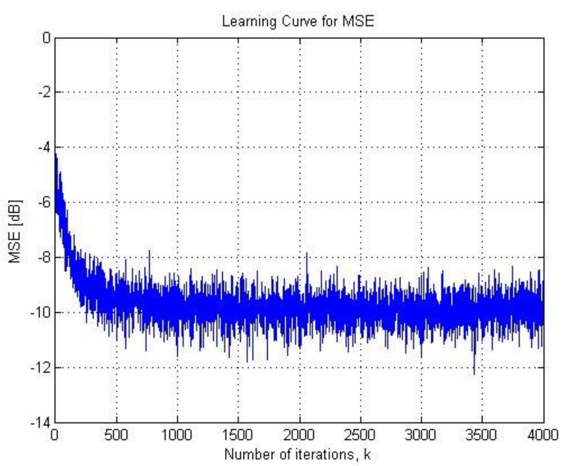

Fig.7 Simulation Result of closed loop system M=8 Np=128

Table III

Np	M	μ	dB
128	8	0.1	-12

V. CONCLUSIONS & FUTURE WORK

The convergence rate behavior of the open loop and closed loop configurations of the delayless subband adaptive filters architecture is studied. It is shown that the subband to-fullband transform greatly affects the performance in terms of the fullband mean square error for the open loop configuration and in terms of the convergence speed for the closed loop configuration. It is shown that based on the results for the closed loop case, a transform with optimal convergence performance can be derived. A novel delayless subband adaptive filter is presented, which employs polyphase adaptive filters This convergence has been analyses and compared to the behavior of the fullband LMS algorithm through of computer MATLAB simulations. We can observe that initially the oversampled subband structure presents better convergence rate. We proposed a closed loop structure with the following features:

1. Less MSE curve.
2. Better convergence.

VI. REFERENCES

[1] M. R. Petraglia, R. *G.* Alves and P. S. R. Diniz, “New Structures for Adaptive in Subbands with Critical Sampling,” *IEEE Trans. on Signal Processing,* vol. 48, No. 12, pp. 3316-3327, Dec. 2000.

[2] P. A. Naylor, O. Tanrikulu, and A. G. Constantinidis, “Subband Adaptive Filtering for Acoustic Echo Control using Allpass Polyphase IIR Filter Banks,” *IEEE Transactions on Speech and Audio Processing*, vol. 6, no. 2, pp. 143–155.

[3] R. Merched, P. S. R. Diniz, and M. R. Petraglia, “A New Delayless Subband Adaptive Filter Structure,” *IEEE Trans. On .Signal Processing,* vol. 47. No 6 1580-1591, Jun. 1999.

[4] Gilloire and M. Vetterli, “Adaptive Filtering in Subbands with Critical Sampling: Analysis, Experimentand Application to Acoustic Echo Cancellation,” *IEEE Trans. Signal Processing,*pp. 1862- 1875, vol. 40, no. 8, Aug. 1992.

[5] N. Hirayama, H. Sakai, and S. Miyagi, “Delayless Subband Adaptive Filtering using the Hadamard Transform,” *IEEE Transactions on Signal Processing*, vol.47, no. 6, pp. 1731–1734, June 1999

[6] J. Huo, S. Nordholm, and Z. Zang, "New Weight Transform Schemes for DelaylessAdaptive Filtering," in *IEEE Global Telecommunications Conference*, 2001,pp. 197–201.

[7] M. R. Petraglia and *S.* K. Mitra, "Performance Analysis of Adaptive Filter Structures based on Subband Decomposition," in *Proc. IEEE Int. Symp. Circ. Syst.,* , Chicago, IL, May 1993

[8] D. R. Morgan and J. C. Thi, "A Delayless Subband Adaptive Filter Architecture,"*IEEE Transactions on Signal Processing*, vol 43 no 8 pp. 1818–1830, August 1995.

[9] S. S. Pradhan and V. U. Reddy, "A New Approach to Subband Adaptive Filtering,"*IEEE Transactions on Signal Processing*, vol. 47 no 3 pp. 655–664, March 1999.