Investigate the Performance Characteristics of RLS Algorithm in Half Band FIR Filter Structure for WiMAX Application

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***Abstract*: In this paper, a novel recursive least square (RLS) algorithm for halfband FIR filter for WiMAX is proposed. A literature survey is carried out in the context of RLS adaptive filter for halfband FIR in WiMAX application. The Recursive least square algorithm has fastest convergence rate at the cost of high computation complexity. The sampling rate required for WiMAX is 8. The work is simulated in MATLAB are obtained.**

***Keywords*: FIR, Halfband, RLS, WiMAX.**

1. Introduction

Owing to the powerful digital signal processors and the development of advanced adaptive algorithms there are a great number of different applications in which adaptive filters are used. The number of different applications in which adaptive techniques are being successfully used has increased enormously during the last two decades. There is a wide variety of configurations that could be applied in different fields such as telecommunications, radar, sonar, video and audio signal processing, noise reduction, between others. The efficiency of the adaptive filters mainly depends on the design technique used and the algorithm of adaptation.

In this paper, a novel Recursive Least Square (RLS) algorithm for Halfband FIR for WiMAX is proposed. The sampling rate required for WiMAX is 8, which in turn reduces the device utilization and power consumption considerably. The work is simulated in MATLAB software. In existing system, the multirate filter has various magnitude response and phase response but it's not sufficient to solve the problem and not obtain the accurate result in real time system. The disadvantage of existing system is overcome here by introducing the Halfband filter (HBF) structure. A proposed algorithm of RLS has fastest convergence rate at the cost of high computation complexity.

This paper [1] describes new strategies to improve the transient response time of harmonic detection using adaptive filters applied to shunt active power filters. Two cases are presented and discussed, both using an adaptive notch filter, but one uses the least mean square algorithm to adjust the coefficients and the other uses the recursive least squares algorithm. This paper [2] presents advanced digital signal algorithms for adaptive filtering applied for noise cancellation and signal analysis in real-time. Correlated and not-correlated signal parts are distinguished by such methods based on the statistical characteristics of the received signals.

Due to the high flexibility and the high integrated of FPGA technology, this paper [3] propose a LMS algorithm using FPGA technology to achieve it, and well used in adaptive filter, not only meet the needs of flexibility and real-time for systems, but also easy to implement, compared with other algorithms it is more simple and more reliable. It has good prospects in the field of signal processing. This brief proposes [4] a novel normalized least mean square algorithm that is characterized by robustness against noisy input signals. To compensate for the bias caused by the input noise that is added at the filter input, a derivation method based on reasonable assumptions finds a bias-compensating vector. This paper [5] deals with the survey of design of adaptive filters using low power adder and multipliers using Very Large Scale Integrated Circuits. The evaluation of power, area and speed for different types of adders and multipliers will be taken into account and the adaptive filter will be designed with efficient combination of adders and multipliers for low power and high speed applications. Various tools will be used for the design of the adaptive filters.

Table 1
Filter design parameters for WiMAX application

Parameter	Values
Over Sampling Rate (Samples/sec)	8
Input sampling frequency *Fs* (MHz)	133.632
Pass band Edge (MHz)	8
Stop band Edge (MHz)	10
Pass band ripple (dB)	0.5
Stop band attenuation (dB)	39

The essential function is to decrease the sampling rate and to keep the passband aliasing within prescribed bounds. In this paper we are designing the halfband FIR filter structure for WiMAX standard as per the specification mentioned in the table 1. The sampling rate reduction required for WiMAX is 8, which is realized with multistage FIR based decimation filter structure, which reduces the computational complexity compared to single- stage realization and other structures considered for analysis.

2. Basics of adaptive filters techniques

The adaptive filters are efficient when compared to multirate filters because it gives the accurate response and analyze the performance metrics such as order of the filter or length of the filter, convergence rate, minimum mean square error [MSE], computational complexity, stability and robustness. There are four major types of adaptive filtering techniques: Adaptive system identification, Adaptive noise cancellation, Adaptive linear prediction and Adaptive inverse system.

A. *Adaptive system identification*

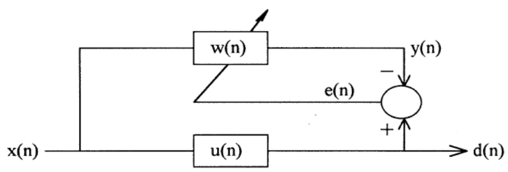

Fig. 1. Adaptive system identification

If the input x(n) is applied to the adaptive filter and the unknown system from which the outputs are compared this fig. 1, The output of the adaptive filter y(n) is subtracted from the output of the unknown system resulting in a desired signal d(n). The resulting difference is an error signal e(n) used to manipulate the filter coefficients of the adaptive system trending towards an error signal of zero. After a number of iterations of this process are performed, and if the system is designed correctly, the adaptive filter's transfer function will converge to, or near to, the unknown system's transfer function. For this configuration, the error signal does not have to go to zero, although convergence to zero is the ideal situation, to closely approximate the given system.

B. *Adaptive inverse system*

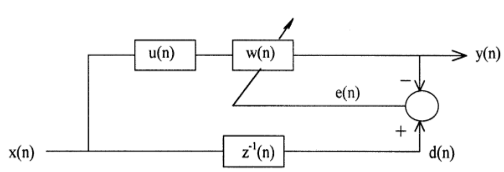

Fig. 2. Adaptive inverse system

When the adaptive inverse system configuration as shown in fig. 2, If the input x(n) is sent through the unknown filter u(n) and then through the adaptive filter resulting in an output y(n). Thus the input is also sent through a delay to attain d(n). As the error signal is converging to zero, the adaptive filter coefficients w(n) are converging to the inverse of the unknown system u(n).

C. *Adaptive noise cancellation*

The fig. 3, is the adaptive noise cancellation as the input x(n) and a noise source N_1(n) is compared with a desired signal d(n) which consists of a signal s(n) corrupted by another noise N_0(n). The adaptive filter coefficients adapt to cause the error signal to be a noiseless version of the signal s(n). Both of the noise signals for this system need to be uncorrelated to the signal s(n). In addition, the noise sources must be correlated to each other in some way, preferably equal to get the best results.

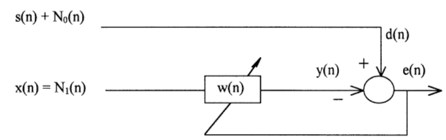

Fig. 3. Adaptive noise cancellation

D. *Adaptive linear prediction*

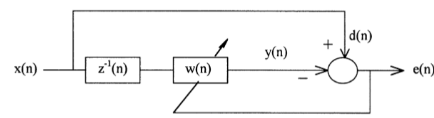

Fig. 4. Adaptive linear prediction

This technique essentially performs two operations as see in fig. 4, The first operation, if the output is taken from the error signal e(n) is linear prediction. The adaptive filter coefficients are being trained to predict from the statistics of the input signal x(n), what the next input signal will be. The second operation, if the output is taken from y(n) is a noise filter. This is true for the linear prediction output because if the error signal did converge to zero, this would mean that the input signal x(n) is entirely deterministic, in which case we would not need to transmit any information at all

3. RLS algorithm

The recursive least square error (RLS) filter is a sample-adaptive, time- update. For stationary signals, the RLS filter converges to the same optimal filter coefficients as the Wiener filter. For non-stationary signals, the RLS filter tracks the time variations of the process. The RLS filter has a relatively fast rate of convergence to the optimal filter coefficients. This is useful in applications such as speech enhancement, channel equalization, echo cancellation and radar where the filter should be able to track relatively fast changes in the signal process. In the recursive least square algorithm, the adaptation starts with some initial filter state, and successive samples of the input signals are used to adapt the filter coefficients.

$$\hat{x}(m) = \boldsymbol{w}^{\mathrm{T}}(m)\boldsymbol{y}(m)$$

where x (m) is an estimate of the desired signal x(m). The filter error signal is defined as

$$\begin{aligned} e(m) &= x(m) - \hat{x}(m) \\ &= x(m) - \boldsymbol{w}^{\mathrm{T}}(m)\boldsymbol{y}(m) \end{aligned}$$

The adaptation process is based on the minimization of the mean square error criterion defined as

$$\begin{aligned} \mathcal{E}[e^2(m)] &= \mathcal{E}\left\{\left[x(m) - \boldsymbol{w}^{\mathrm{T}}(m)\boldsymbol{y}(m)\right]^2\right\} \\ &= \mathcal{E}[x^2(m)] - 2\boldsymbol{w}^{\mathrm{T}}(m)\mathcal{E}[\boldsymbol{y}(m)x(m)] + \boldsymbol{w}^{\mathrm{T}}(m)\mathcal{E}[\boldsymbol{y}(m)\boldsymbol{y}^{\mathrm{T}}(m)]\,\boldsymbol{w}(m) \\ &= r_{xx}(0) - 2\boldsymbol{w}^{\mathrm{T}}(m)\boldsymbol{r}_{yx}(m) + \boldsymbol{w}^{\mathrm{T}}(m)\boldsymbol{R}_{yy}(m)\boldsymbol{w}(m) \end{aligned}$$

The Wiener filter is obtained by minimizing the mean square error with respect to the filter coefficients.

$$w = R_{yy}^{-1}\, r_{yx}$$

where R_{yy} is the autocorrelation matrix of the input signal and r_{yx} is the cross-correlation vector of the input and the target signals.

$$R_{yy} = Y^{T}Y = \sum_{m=0}^{N-1} y(m)\,y^{T}(m)$$

where y(m)=[y(m), ..., y (m–P)]T. Now, the sum of vector product

$$R_{yy}(m) = R_{yy}(m-1) + y(m)\,y^{T}(m)$$

To introduce adaptability to the time variations of the signal statistics,

$$R_{yy}(m) = \lambda R_{yy}(m-1) + y(m)\,y^{T}(m)$$

where λ is the so-called adaptation, or forgetting factor, and is in the range 0> λ >1. Similarly, the cross-correlation vector is given by

$$r_{yx} = \sum_{m=0}^{N-1} y(m)x(m)$$

The sum of products can be calculated in recursive form as r_{yx} (m)

$$r_{yx}(m) = r_{yx}(m-1) + y(m)x(m)$$

Again this equation can be made adaptive using an exponentially decaying

$$r_{yx}(m) = \lambda r_{yx}(m-1) + y(m)x(m)$$

For a recursive solution of the least square error, we need to obtain a recursive time-update formula for the inverse matrix in the form.

4. Graphical analysis

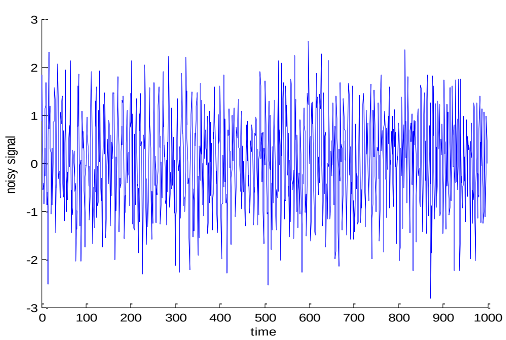

Fig. 5. Time domain of input and error

In the above Fig. 5, shows that the time domain of input and error.

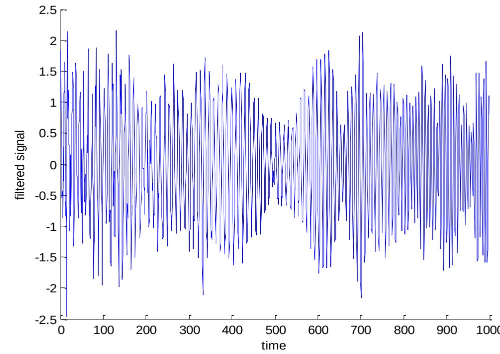

Fig. 6. Mean Squared Error

The Fig. 6, shows that the mean squared error.

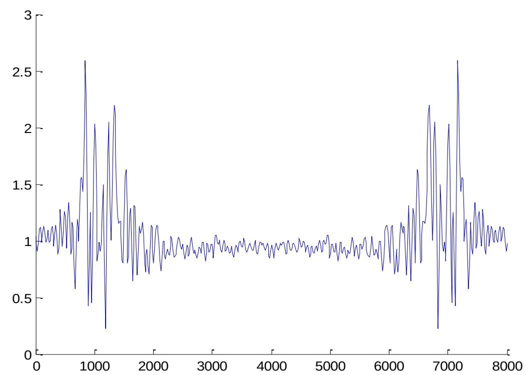
Fig. 7. Frequency spectrum of filtered signals

The Fig. 7, shows that the frequency spectrum of filtered signals.

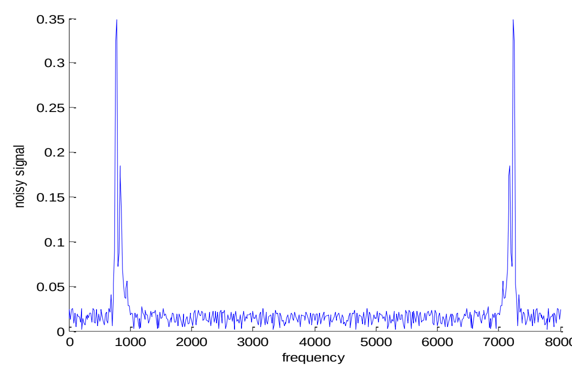

Fig. 8. Frequency spectrum of unfiltered signals

The Fig. 8, shows that the frequency spectrum of unfiltered signals.

The Fig. 9, shows that the final coefficient and mean squared error for filtered signals.

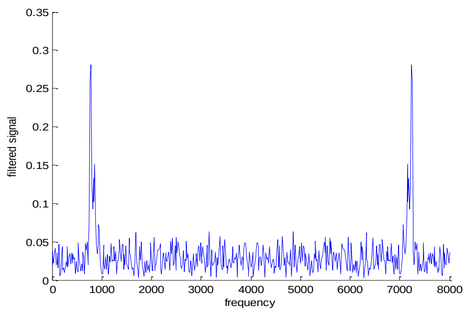

Fig. 9. Final coefficient and MSE for filtered signals

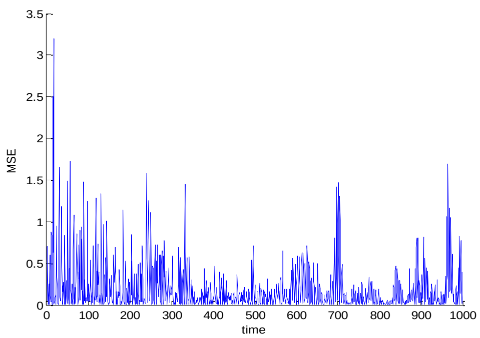

Fig. 10. Final coefficient and MSE for unfiltered signals

The Fig. 10, shows that the final coefficient and mean squared error for unfiltered signals.

5. Conclusion

Different Adaptive filters techniques are discussed and the results shows that the complexity can be reduced and compared to multistage realization techniques. Hence these results are analyzed with MATLAB using RLS algorithm, so investigated the frequency spectrum of halfband filter techniques are successfully implemented.

6. Future work

It is possible to design the filter structure in MATLAB - Simulink model and finally it is realize in Virtex-V FPGA using Xilinx system generator will be done in future. So far we analysed the different Adaptive filter methods and the results show that the complexity can be reduced and to found the different performance metrics using WiMAX application.

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