Penetration signal Analysis based on RLS Adaptive Wavelet Transformation

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***Abstract* — The hard target penetration process is very complex, it contains axial de-acceleration vibration and some weak interfering random signals. This is a non-stationary random vibrating signal. Although filtering method based on Hard threshold and Soft threshold method is effective on some extend, but it cannot completely remove the high-g acceleration signal components which is related to vibration and interference. As such this paper proposes a new method which analyzes the multivariate signal through Recursive Least Squares Adaptive Wavelet Transformation. Further with the similar signal and multivariate processing this paper compares the RLS Adaptive and Hard/Soft Threshold in Mat lab Simulation Software.**

***Keywords*— complex, random, soft threshold, hard threshold, RLS adaptive filtering,**

I. INTRODUCTION

Based on the filtering method that exists, the paper present a method which uses the different wavelet function to decompose noise. First the original measured signal is decompose by using wavelet function into low frequency and high frequency signal. Then, the block adaptive filter design selected for the Recursive Least Square algorithm allow the low frequency and high frequency signal to pass through this filter model. The method by which the error signal is detected by using Wavelet Transform'

Adaptive Filter (WTAF) [1] is known as Wavelet Transform Adaptive Signal processing. The adaptive filtering on the sub band signal which is obtained by wavelet decomposition and reconstruction is the application of WTAF. The Adaptive filtering technique provides good convergence and low computational complexity that can be easily adaptable to non-stationary signal as well. At the output the signal is reconstruct by using the filtered low frequency and the high frequency signal. In this paper, we choose similar penetration signal and RLS algorithm in mat lab simulation software.

II BLOCK WAVELET ALGORITHM

In terms of the type of the wavelet transform being used, the paper present the DWT based WTAF. D-WTAF could also be use for adaptive filtering in different time's domains, the Before-Reconstruction structure that correspond to adaptive filtering in the scale-domain and the After-Reconstruction structure that correspond to adaptive filtering in the time-domain. In the sub band error based D-WTAF, the error signal in each sub band is used as input to the LMS algorithm. In order to speed up the calculation, this paper developed the block RLS based WTAF, that modifies the weights of the adaptive filter block-by-block instead of sample-by-sample. We observe that the signal-to-noise ratio (SNR) was greatly increased by applying these WTAFs. This makes a possibility of lower sampling rate, and greatly improves the system speed to meet faster testing requirements.

A conventional block adaptive filter is depicted in Fig. 1.

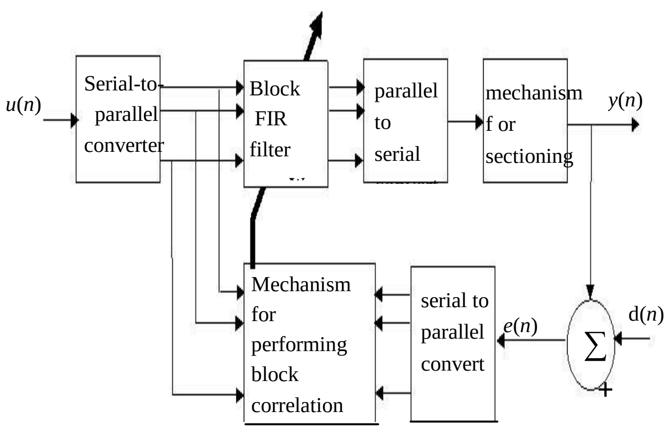

Fig. 1 Conventional block adaptive filter

The serial to parallel converter section the incoming data sequence u(n) into L- block[1] which is applied to an FIR filter of length M, one block at a time. The adaptive filter tap weight is held fixed and the filtering is done block by block rather than sample by sample as in standard LMS algorithm [2].

The Wavelet Transform (WT) processes both stationary as well as non-stationary signal, thus the WT is used to separate the signal into sub bands and is subsequently applied to block adaptive filtering algorithm. The Block Wavelet Transform Adaptive Filter [1]-[4] is shown in Fig. 2

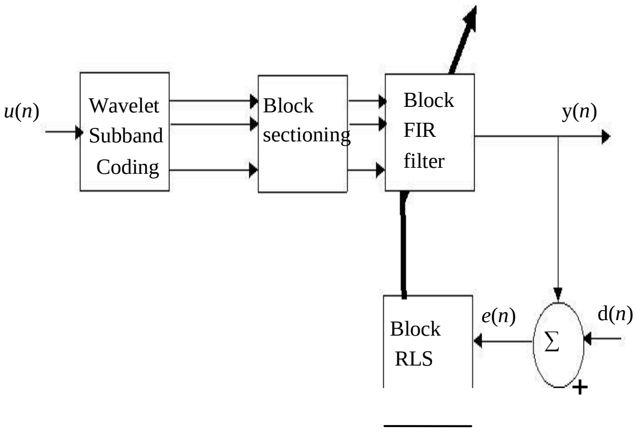

Fig. 2 Block wavelet transform Adaptive Filter

The input signal u (n) is processed by wavelet decomposition and reconstruction Filter banks, and the sub band signal **v**(n) is obtained. The properly sectioned sub band vector input of **v**(n) is subsequently processed by the Block FIR Filter [1], whose weights held fixed during the block. The estimated output y(n) is compared with the desired signal d(n) to obtain the error signal e(n) . The Block LMS algorithm finally adjust the weights of the block FIR filter based on the error signal e(n) , and the block FIR filter is now ready to process the next sub band block.

Let k refer to block time, and w(k) denote the tap-weight vector of the filter for the kth block, as shown by

$$\mathbf{w}(k) = [w_0(k), w1(k), \ldots, w_{M-1}(k)]^T, \quad k = 0, 1.. \qquad (1)$$

The index n is reserved for the original sample time, written in terms of the block time as follows

$$n = kL+i \ , \ i = 0, 1, \ldots, M-1 \qquad (2)$$

Let the input signal vector v(n) the same as before. Then, at time n the output y(n) produced by the filter in response to the input signal vector **v**(n) is defined by the inner product

$$y(n) = \mathbf{w}^T(k)\mathbf{v}(n) \qquad (3)$$

Equivalently, in light of (2) we may write

$$y(kL+i) = \mathbf{w}^T(k)\mathbf{v}(kL+i), \quad i=0, 1\ldots M-1 \qquad (4)$$

Let

$$d(n) = d(kL+i) \qquad (5)$$

Denote the corresponding value of the desired response. An error signal e(n) is produced by comparing the filter output y(n) against the desired response d(n) , as shown in Figure 1, which is defined by

$$e(n) = d(n)-y(n) \qquad (6)$$

or equivalently,

$$e(kL+i) = d(kL+i) - y(kL+i) \qquad (7)$$

Thus, the error signal is permitted to vary at the sampling rate as in the standard LMS algorithm. The error signal is sectioned into L-point blocks in a synchronous manner with **v**(n) and then used to calculate the modification of the tap weights of the filter. For each block of data we have different values of the error signal to use in the adaptive process. For the kth block, we define an average estimate of the gradient vector, as shown by[2]

$$\nabla(k) = \frac{2}{L}\sum_{i=0}^{L-1} v(kL+i)\, e(kL+i) \qquad (8)$$

Now using the optimum solution for Adaptive filter, we have the following update equation for the tap-weight vector of the block LMS algorithm

$$w(k+1) = w(k) + \mu\sum_{i=0} v(kL+i)e(kL+i) \qquad (9)$$

Where μ is the step-size parameter and the factor 1/L is absorbed into μ.

III HARD/SOFT THRESHOLD FILTERING

In the process of signal analysis and filtering hard threshold and soft threshold are often mentioned and are used for the purpose of filtering. The equation for hard threshold and soft threshold are given as

$$f(\omega) = \begin{cases} 0, & |\omega| < t \\ \omega I, & |\omega| \geq t \end{cases} \quad (10)$$

and

$$f(\omega) = \begin{cases} 0, & |\omega| < t \\ \omega - sgn(\omega)t, & |\omega| \geq t \end{cases} \quad (11)$$

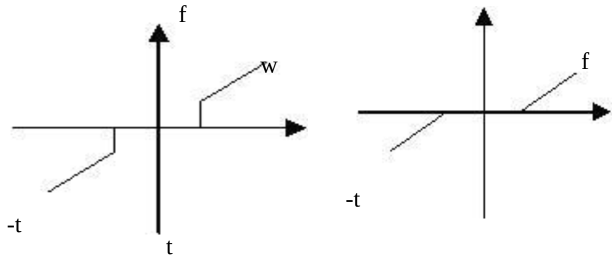

Fig.3 (a) hard threshold (b) Soft threshold

From the given equation and the diagram, it is found that when the absolute value is smaller than the threshold value, it is set to zero. In hard threshold filtering it generate interrupt at some points such that we get poor continuity of wavelet coefficient and form oscillatory reconstructed signal. Whereas soft threshold constantly shrink boundary by comparing discontinuous points and prevent interruption, and the reconstruction of the signal become very smooth. Now the decomposed continuous wavelet coefficients are good. However if the coefficient of the decomposed wavelet is large, deviation could appear with the real coefficient resulting reconstruction error.

The Adaptive filtering is also used to remove noise signal. The effect of its filtering depends on filtering algorithm [8]. During the filtering process it update and adjust the weighting coefficient for each sample of the input signal x(n) as per the algorithm used. We can obtain the Recursive least square error between the output sequence and the desired signal sequence. The diagram for the RLS adaptive filtering is shown in figure 4.

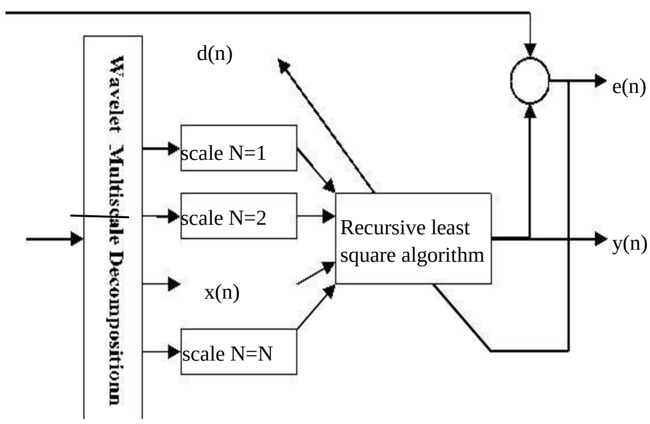

Fig.4 RLS adaptive filtering based on a wavelet filtering

The main determining factors for filtering in this model are:

1. Type of wavelet: Different type of wavelet show different decomposition. So paper used Daubechies 2 as mother wavelet which has the orthogonality, compactness and proximity with penetration [2].
2. Wavelet decomposition layers: The number of layer to be used with Daubechies 2 [2] wavelet requires special analysis.

Adoptive algorithm: There are many algorithms that determine the filtering effect of the high frequency coefficient of wavelet decomposed and the reconstructed signal after filtering. The paper uses the Recursive least square algorithm for verification.

IV PROCESSING METHODS

To verify the proposed adaptive filter model, the paper adopts the white noise signal that is similar to penetration signal and the simulation is done in mat lab. The paper uses Heursure soft threshold method, Sqtwolog hard threshold method and the proposed method.

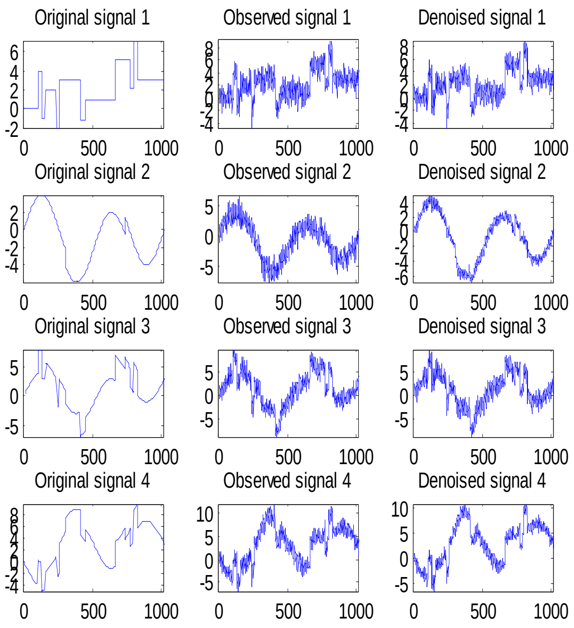

Fig.5 Hard threshold De-noising

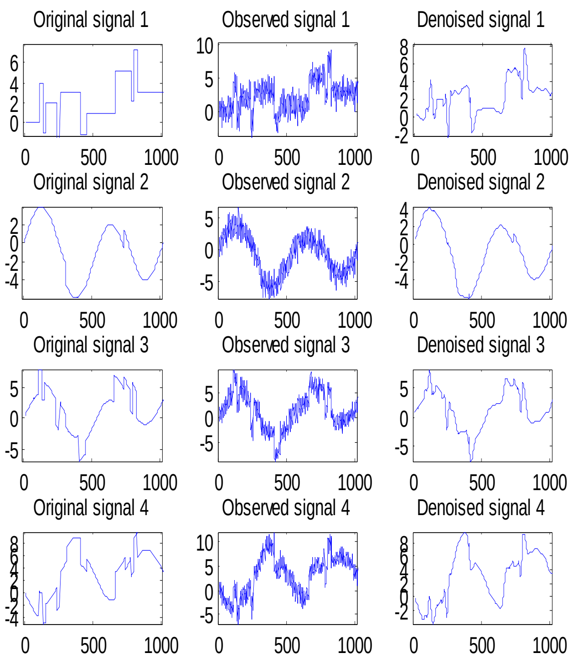

Fig.6 Soft threshold De-noising

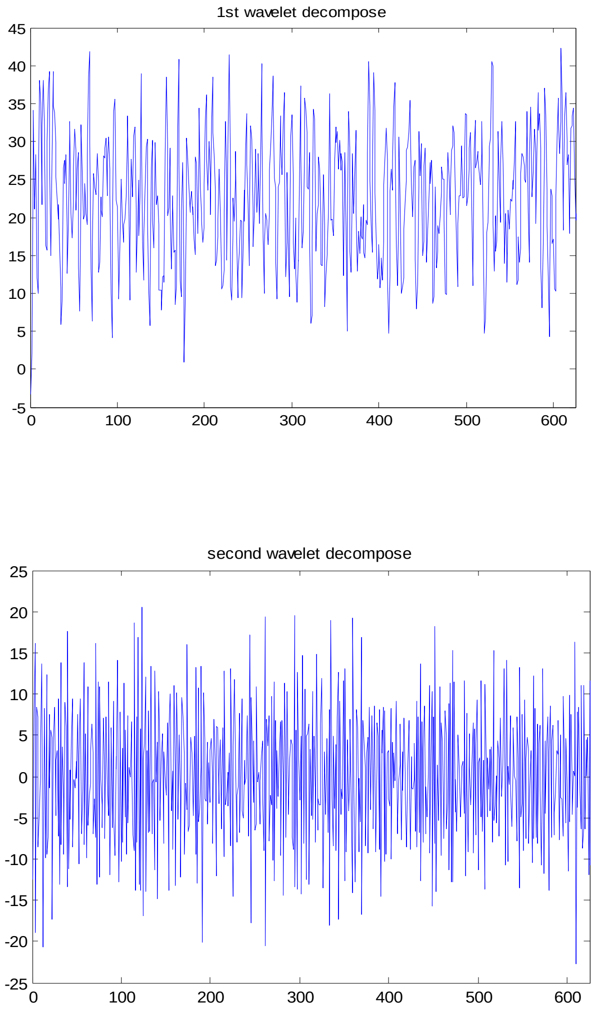

Fig.7 db2 wavelet decomposition

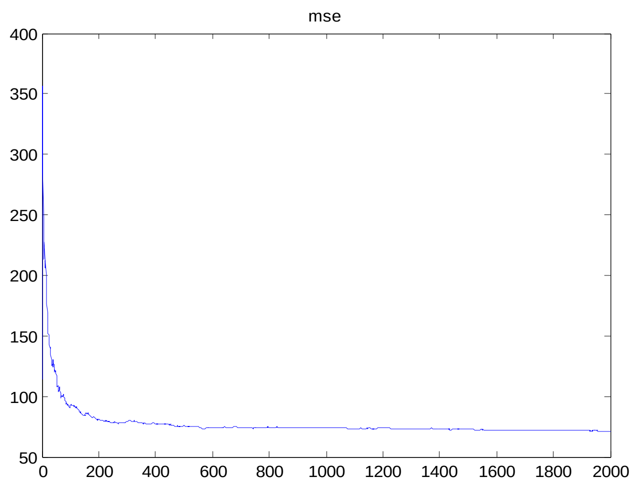

Fig.8 MSE

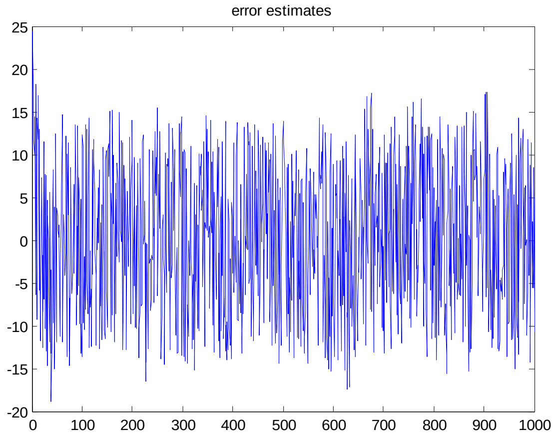

Fig 9. Error Estimate

V CONCLUSION

The fast and easy Computation make Wavelet RLS adaptive filter very attractive. RLS Adaptive filtering based on wavelet decomposition method not only provides good signal waveform but also better signal to noise ratio (SNR) as compared to Hard/Soft Threshold method.

This method of filtering is done for non-stationary random vibrating signal with reference to high-g penetrating signal filtering. Since many penetrating signal are relatively complex, more detail analysis are required, also the performance of this filtering may vary with different type of signal under consideration.

Table SNR OF THREE FILTER METHOD

Filtering Method	Hard Threshold	Soft Threshold	RLS adaptive decomposition
SNR	6.275	20.396	40.369

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