A fast running FIR Filter structure reducing computational complexity

Jae-Kyun Lee*, Chae-Wook Lee*
* Computer and Communication Engineering Department, , ljk6031@daegu.ac.kr
** Electronic Engineering Department, cwlee@daegu.ac.kr
Daegu University, 712-714, Korea

Abstract - In this paper, we propose a new fast running FIR filter structure that improves the convergence speed of adaptive signal processing and reduces the computational complexity. The proposed filter is applied to wavelet based adaptive algorithm. Actually we compared the performance of the proposed algorithm with other algorithm using computer simulation of adaptive noise canceler based on synthesis speech. As the result, We know the proposed algorithm is prefer than the existent algorithm.

Keywords: FIR filter, noise canceler, fast wavelet

1 Introduction

LMS (Least Mean Square) algorithm using steepest descent way in adaptive signal processing requires simple equation and is used widely because of the less complexity. But eigen-values change by width of input signals in time domain, so the rate of convergence becomes low. In order to solve this problem, FFT (Fast Fourier Transform) or DCT (Dispersion Cosine Transform) are used by removing the correlation rate between signals[1][2].

But FFT or DCT using accomplishment of adaptive algorithm of frequency domain is needed the additional complexity by changing time domain to frequency domain. On this purpose, WTLMS(Wavelet Transform LMS) adaptive algorithm is announced. WTLMS algorithm is better than existent frequency domain algorithm over the rate of convergence [3].

In this paper, we propose a new fast running FIR filter structure to reduce computational complexity. Adaptive algorithm of time domain is accomplished in frequency domain using orthogonal Wavelet. Appling to Wavelet based adaptive algorithm to reduce the rate of convergence improvement and complexity in frequency domain.

The proposed algorithm is applied to computer simulation by adaptive noise canceller. Each algorithm's complexity and the rate of convergence are compared. We know that the proposed algorithm has the same performance as existent algorithm for computational complexity.

2 Wavelet based adaptive algorithm

As eigen-values are changed by the widths of input signals in time domain, the rate of convergence becomes relatively low. Therefore, in this paper, we use Wavelet adaptive algorithms of frequency domain instead of time domain. Wavelet adaptive algorithm is shown in Figure 1.

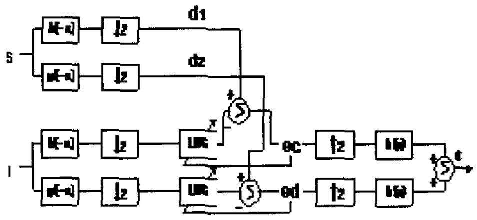

Figure.1. Wavelet adaptive algorithm

3 Fast wavelet algorithm

DWT causes many computations by actuality hardware implement because of conversion between input signal and Wavelet coefficient convolution. Fast convolution algorithm can divide two forms by reducing this convolution's calculation. One is FFT based on fast algorithm. And another is short-length fast running FIR algorithm. In case the order of filter is long, we use FFT based fast algorithm and in case the order of filter is short, we use short-length fast running FIR algorithm[4][5][6].

First of all, we assume that z transform of X(n) is X(z). We can see like that if z transform of sub-sample output signal y(n) calculation.

$$Y(z) = \frac{1}{N}\sum_{n=0}^{N-1} X(W_N{}^n z^{1/N}), W_N = e^{-j2\pi/N} \qquad (1)$$

The other way about up-sampling signal of input signal is converted like that.

$$Y(z) = X(z^N) \tag{2}$$

Convolution algorithm is processing effectively. The transform signal in z transform field shows multiplication form if input signal and impulse-response are converted to z transform. First, high speed convolution algorithm is like that if 'N=2' is sum-sample and there is 3 channel filter bank which has filter consisted.

$$h(z) = [z^{-1}, 1+z^{-1}, 1]^T \tag{3}$$

$$C(z) = diag[H_0(z^2), H_0(z^2)+H_1(z^2), H_1(z^2)]^T \tag{4}$$

$$g(z) = [1-z^{-1}, z^{-1}, z^{-2}-z^{-1}]^T \tag{5}$$

We make system output of figure 1 like that if we use both (1) and (2).

$$Y(z) = [z^{-1} \cdot H_0(z^2) + z^{-2} \cdot H_1(z^2)] \cdot X(z) \tag{6}$$

When H(z) is endowed with desired filter, we make difference method to $H_0(z^2), H_1(z^2)$.

$$H_0(z^2) = \frac{1}{2}[H(z) + H(-z)] \tag{7}$$

$$H_1(z^2) = \frac{1}{2}[H(z) - H(-z)] \tag{8}$$

If filter degree is 2K here, the filter degree of (7), (8) is decreased. We make (9) like that with using (7), (8).

$$Y(z) = z^{-1} \cdot H(z) \cdot X(z) \tag{9}$$

As you can see (7),(8) and (9), there is renewed 3 filter that is decreased in half than filter degree which has original FIR filter. After all, there is reducing multiplication amount per 25% output sample. In the other hand, there is an addition which needs one time addition per two time input and 3 time addition per two time out output. You can see picture no.2 to block diagram of algorithm

Short-length fast running FIR algorithm is shown in Figure 2.

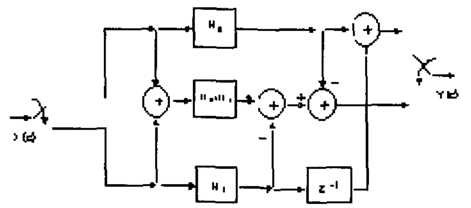

Figure 2. Short-length fast running FIR algorithm

When FIR filter's order supposed L, we compare mathematical complexity between existent FIR filter and fast FIR algorithm existent FIR filter need multiplication of L times and addition of L-1 times per output while fast FIR algorithm need multiplication of (3/4)L times and addition of (3/4)L times + (1/2)times per output.

4 Proposed algorithm

In case the apply short-length running fast FIR algorithm of forward of figure 2 to wavelet adaptive algorithm of figure 1, can take the place of down sampling's role as that omit one of two output part to short-length running FIR algorithm because accomplish down sampling in wavelet adaptive algorithm. removing output by even number times of two output part in this paper, Proposed algorithm calculates multiplication of (2/3)L times and addition of (1/4)L times than applying short-length running fast FIR algorithm in existent wavelet adaptive algorithm. And we know that it shows the same performance.

The proposed first fast running FIR algorithm is shown in Figure 3.

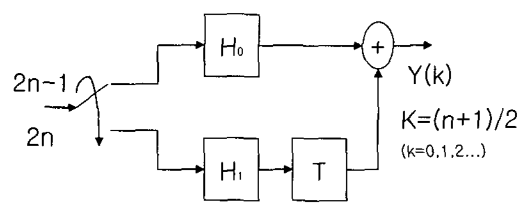

Figure 4. Proposed second fast running FIR algorithm

As you can see the figure 2, the proposed algorithm calculates the multiplication of (2/3)L times per output and addition of (3/4)L times per output than existent wavelet adaptive algorithm as that is up-sampling after LMS algorithm, so the multiplication of Daubechies wavelet filter coefficient of H1 is zero, therefore it is removed filter coefficient. And we know that it shows the same performance.

The proposed second fast running FIR algorithm is shown in Figure 4.

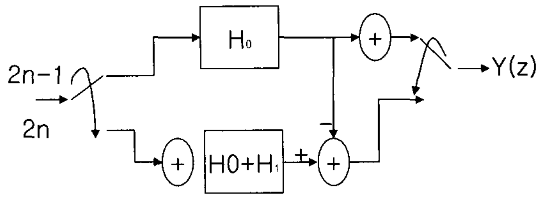

Figure 4. Proposed second fast running FIR algorithm

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Figure 5 is the diagram that applies the proposed fast running FIR algorithm to wavelet adaptive algorithm.

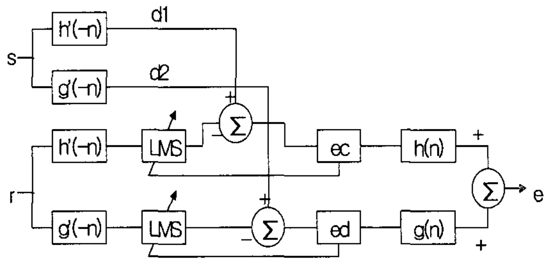

Figure.5. Block diagram that applying to Wavelet adaptive algorithm

5 Simulation

x(n) in simulation is input environment, and white Gaussian noise is used, which the average is zero and power is one. Original signal is 16 kHz sampling frequency and it was made woman complex sound of 16bit quantization. Transfer route of noise is approximated by third FIR filter. Input Speech SNR is -4.49dB. The order of filter is 32nd of each algorithm. In WLMS algorithm case, wavelet conversion is used to Daubechies D_4 wavelet filter. Adaptive constant is used to 0.06. We use other algorithm like the same type.

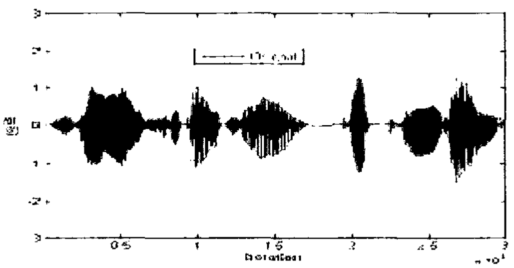

Figure 6. Graph of original signal

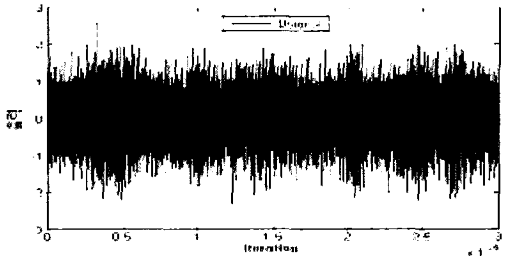

Figure 7. Signal including noises

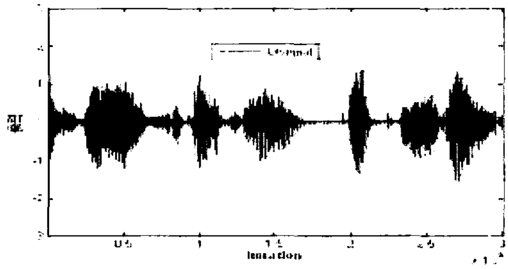

Figure 8. Graph of output signals

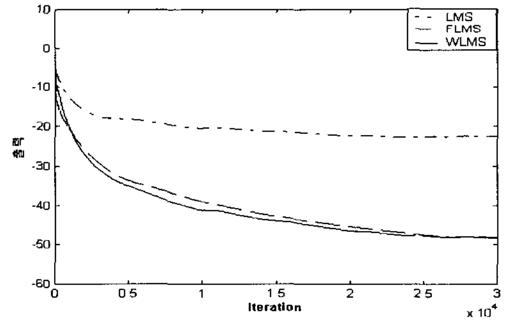

Figure 9. MSE performance curve of each algorithm

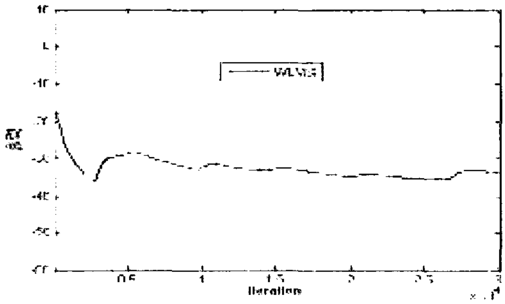

Figure 10. MSE performance curve of Wavelet adaptive algorithm

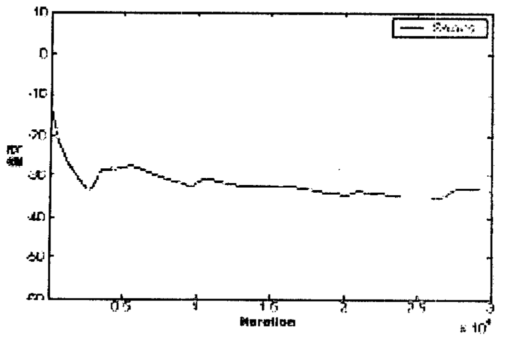

Figure 11. MSE performance curve of short-length running fast FIR algorithm

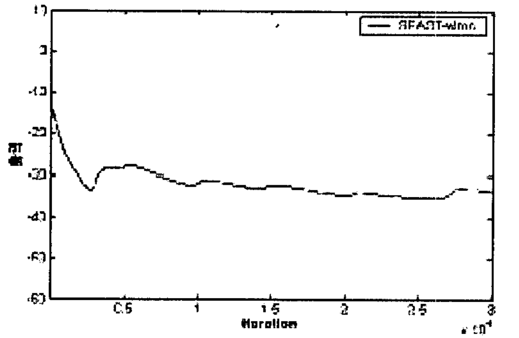

Fig.ure 12. MSE performance curve of fast1-short-length running fast FIR algorithm

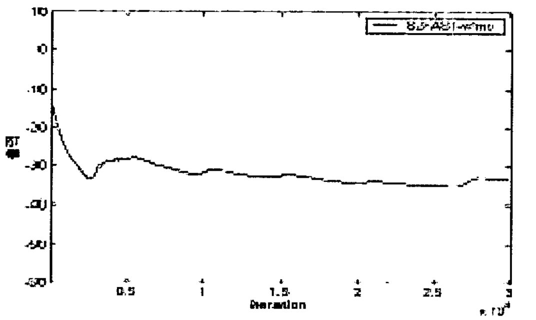

Figure 13. MSE performance curve of fast2-short-length running fast FIR algorithm

6 Conclusion

In this paper, we researched wavelet based adaptive algorithm in order to improve the rate of convergence using a fast running FIR fliter structure. The proposed algorithm, which is conducted with high-speed and the complexity of multiplication is reduced, becomes more faster algorithm. As the result, the proposed algorithm shows more stable and faster rate of convergence character. The proposed algorithm might apply to speech application to adaptive signal processing field. After we will study about math analysis of wavelet conversion character and many field of adaptive signal processing.

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