A convergence analysis of Block MADF algorithm for adaptive noise reduction•

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Abstract

When it calculates the optimum price of filter coefficient, the many operation quantity is necessary. Is like that the real-time control is difficult and the hardware embodiment expense is big. The case which does not know advance information of input signal or the case where the statistical nature changes with change of surroundings environment is necessary the adaptive filter. Every hour to change a coefficient automatically and system in order to reach to the condition of optimum oneself, the fact that is the adaptive filter. When it does not the quality of input signal or it does not know the environment of surroundings every hour changing, it does not emit not to be, in order to collect, the fact that is the adaptive filter.

The case of the Acoustic Echo Canceler does thousands filter coefficients in necessity. It reduces a many calculation quantity to respect, it uses the IIR filter from hour territory. Also it uses the block adaptive filter which has a block input signal and a block output signal. The former there is a weak point where the stability discrimination is always demanded. Consequently, The block adaptive filter is researched plentifully. This dissertation planned the block MADF adaptive filter used to MADF algorithm.

1. Introduction

The adaptive approach based on the filtering of background noise adjusts its parameters for each new available sample. The adaptive noise cancelling(ANC) method based on the Wiener theory exploits adaptive optimal filtering concepts proven to be useful in many signal processing applications[1]. The goal of the ANC algorithm is to eleminate background noise from the main signal, which is composed of the desired signal and background noise that has been correlated with noise from a reference measurement. The technique therefore relies upon access to a reference signal, located at the source of noise fields, as well as the main or primary signal[2].

In 1965, Widrow and Hoff developed adaptive algorithms including least mean squares(LMS), and was realized it to the adaptive noise cancelling system[3]. After this date, the ANC algorithm was successfully employed in many signal processing, seismic, and biomedical areas.

In order to assure the objectivity of biomedical signal, there must be groped for new methods with analysis and clinical diagnosis. Although the signal is nonstationary, the adaptive digital filter is designed by optimal coefficients in time varying. However the LMS algorithm is inappropriate for the fast-varying signals due to its slow convergence properity.

An alternative approach is the recursive least-square (RLS) method. The RLS algorithm has been widely used inreal-time system identification applications and fast start-up channel equalization because of its fast convergence rate and stable filter characteristics. However the RLS algorithm may be computationally costly for some application since it requires M^2 operation per time update[4,5].

One of modified LMS algorithm is the Sign algorithm, which can reduce the multiplication operation and obtain the stabilized effects in comparing with another algorithm.

After this date, many efforts to reduce the multiplication number have been by Lee and Un, they proposed the MADF(multiplication free adaptive digital filter) algorithm using the LMS and DPCM(Differential Pulse Code Modulation) method[6]. Also, Park, Youn and Cha[7] were experimented the MADF algorithm using the Sign and DPCM[8].

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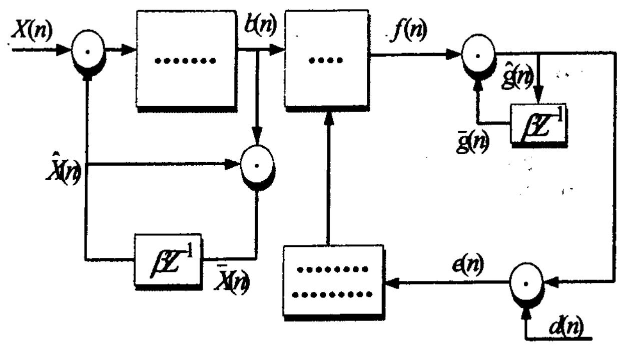
2. Structure of MADF and BMADF

2.1 MADF algorithm

The DPCM for the input signal of Sign algorithm is used from the MADF. The standard signal is used in the renewal equation for an adaptive filter coefficient. The D (N) it is a desired signal. The X (N) is an input signal. it shows the basic structure of MADF structure.



< Fig 1. basic structure of MADF >

The $\hat{X}(n)$ and the $\dot{X}(n)$ each are forecast input of the DPCM and remaking input vector.
It is an input vector of the DPCM.

$$E(n)=[\varepsilon(n),\varepsilon(n-1),\varepsilon(n-2),\ \ldots,\varepsilon(n-N+1)]^T \qquad (1)$$

It is an output vector of the DPCM.

$$B(n)=[b(n),b(n-1),b(n-2),\ \ldots,b(n-N+1)]^T \qquad (2)$$

The next equations are showing a MADF structure.

$$\bar{X}(n)=\hat{X}(n)+B(n) \qquad (3)$$

$$f(n)=B^T(n)H(n) \qquad (4)$$

$$\hat{g}(n)=\beta\bar{g}(n-1) \qquad (5)$$

$$\bar{g}(n)=\hat{g}(n)+f(n) \qquad (6)$$

$$e(n)=d(n)-\bar{g}(n) \qquad (7)$$

It renews a coefficient it is last for.

$$H(n+1)=H(n)+\mu\,\nabla(n) \qquad (8)$$

2.2 Structure of block adaptive filter

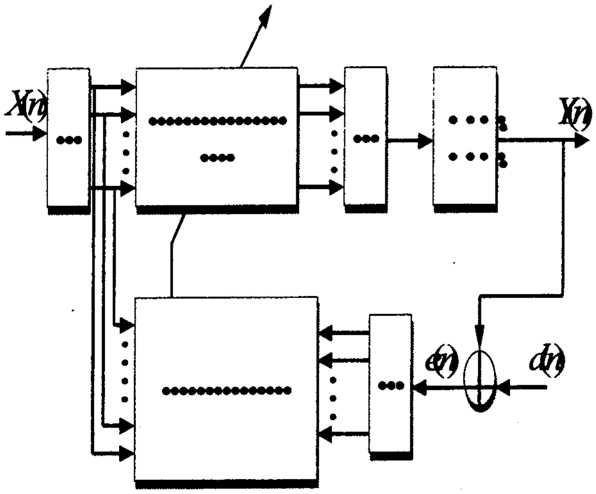
The execution of the block adaptive filter the arranging in a row processor will use an accumulation circuit or and it will be able to use efficiently. The filter subtracting is vector form. And tap input of the filter becomes the M• L procession. The selection of block length M is important from plan of the block adaptive filter. It selects a block length M and a filter subtracting L same from application of the block adaptive filter.

$$Y(n)=X(n)\,H(n) =[y(n)(M+1),\ y(n-1)(M+2),\ \cdots,y(n-N+1)(M)]^T \qquad (9)$$

$$X(n)=[x(n)(M+1),x(n-1)(M+2),\ \cdots,x(n-N+1)(M)]^T \qquad (10)$$

$$H(n)=[h(0,n),h(1,n),\ \cdots h(L-1,n)]^T \qquad (11)$$

The filter Y(n) which is a length M and the X(n) is procession of output vector and input vector. The H(n) is n filter coefficient.



< Fig 2. Basic structure of block adaptive filter >

2.3 Block adaptive algorithm

It does a block length M and a filter subtracting L same.

$$X(n)=[x(n)(M+1),x(n-1)(M+2),\ \cdots,x(n-N+1)(M)]^T \qquad (12)$$

$$\hat{X}(n)=[\hat{x}(n)(M+1),\hat{x}(n-1)(M+2),\ \cdots,\hat{x}(n-N+1)(M)]^T \qquad (13)$$

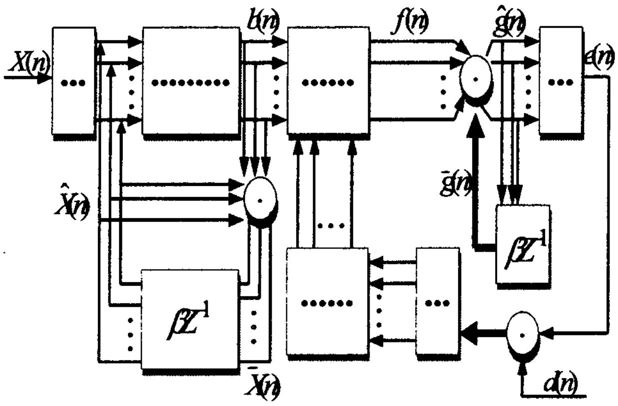
$$\bar{X}(n)=[\bar{x}(n)(M+1),\bar{x}(n-1)(M+2),\ \cdots,\bar{x}(n-N+1)(M)]^T \qquad (14)$$

It is an input vector of the DPCM.

$$B(n)=[b(n)(M+1),\ b(n-1)(M+2),\ \cdots,b(n-N+1),(M)]^T \tag{15}$$

It is an output vector of the DPCM.

$$E(n)=[\varepsilon(n)(M+1),\ \varepsilon(n-1)(M+2),\ \cdots,\varepsilon(n-N+1),(M)]^T \tag{16}$$



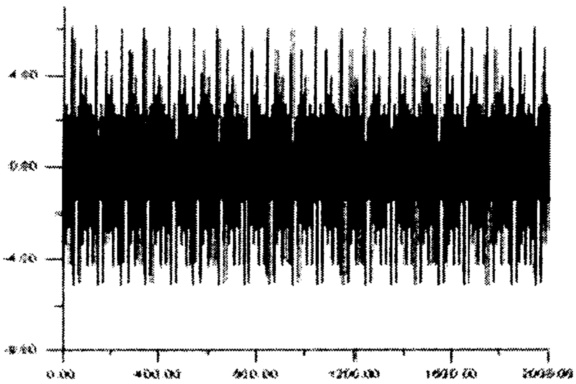
<Fig 3. Structure of block MADF >

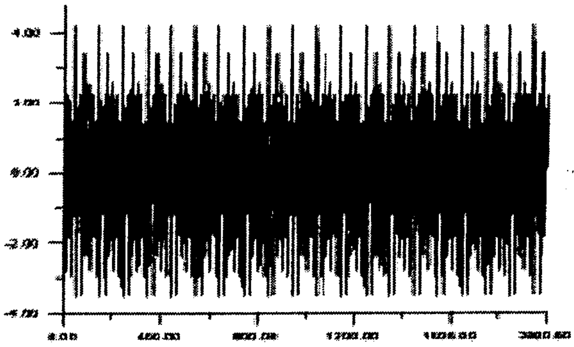
It renews a coefficient it was cool lastly for

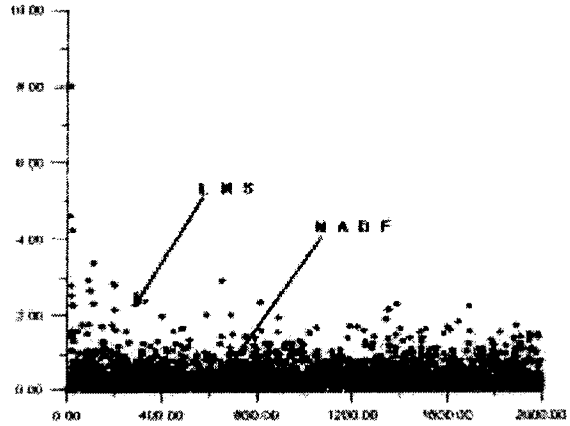
$$H(L,n+1)=H(L,n)+\mu\nabla(n) \tag{17}$$

3. Experimental results

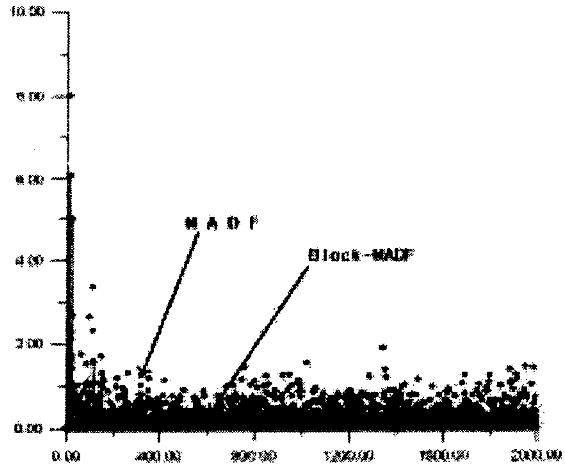
For the efficiency analysis of the block MADF algorithm which it proposes from this dissertation It applied in the adaptive noise removal flag and computer simulation it did. The input X (N) did in Gaussian noise. Whole input data possibility it is becoming 2000 samples.


<Fig 4. Voice signal of origin >


< Fig 5. The signal where the noise is added >



< Fig 6. Collecting quality of the LMS and the MADF it compares >



< Fig 7. Collecting quality of the MADF and the Block - MADF it compares >

4. Conclusion

The problem point against an inevitably many calculation quantity follows from the application highly subtracting in necessity. It solves this problem point to respect, It changed a MADF structure from dissertation and it proposed a block MADF algorithm.

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