Adaptive Noise Reduction on the Frequency Domain using the Sign Algorithm.

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Abstract : We have proposed the adaptive noise reduction algorithm using the MDFT. The algorithm proposed use the linear prediction coefficients of the AR method based on Sign algorithm that is the modified LMS instead of the least mean square(LMS). The signals with a random noise tracking performance are examined through computer simulations and confirmed that the high speed adaptive noise reduction processing system is realized with rapid convergence.

Keywords: adaptive filter, sign algorithm, convergence speed, modified discrete Fourier transform(MDFT).

I. Introduction.

The convergence speed of a time domain adaptive filter is degraded when an input signal is colored since the eigenvalue spread become large. In order to accelerate the convergence speed of such a algorithm in th frequency domain, an adaptive filter has been proposed which the input autocorrelation matrix is approximately diagonalized by using the discrete Fourier transform(DFT) and normalized by the time-variable step size algorithm[1-2]. The FFT method guarantees stable convergence, but a continues output signal can't be obtained for block-processing[3].

The frequency sampling filter(FSF) method[3] is able to process sample by sample, but the repeated structure bring an unstable convergence and accumulated error[4]. In order to obtain the continued output signal to preserve stable convergence property, we have proposed the adaptive filter using MDFT on the frequency domain. In adding to improve its convergence performance proposed the spectral error method using the MDFT[5]. This method is increased the convergence speed.

In this paper, we proposed the high speed adaptive noise reduction algorithm on the frequency domain using the Sign algorithm. It is know that the proposed system is to evaluate linear prediction coefficient using the AR(auto-regressive) method. In order to increase the convergence speed, the noise reduction algorithm is adapted. By computing the existed algorithm, the proposed system is identified the fast convergence speed on frequency domain.

II. The properties of MDFT and SLMS.

When the input signal in DFT is real, the modified DFT(MDFT) is as follows[2],

$$X_{k,i} = \sum_{n=0}^{N-1} x_{i-n} \cos\frac{2\pi nk}{N} \tag{1}$$

$$x_i = \frac{X_{0,i}}{N} + \frac{2}{N}\sum_{k=1}^{\frac{N}{2}-1} X_{k,i} \tag{2}$$

Where N is the number of samples and assumed to be even in the following. The notes n, i and k are used as a time, another time and frequency index, respectively. We are proposed a block diagram to reduce the noise as figure 1.

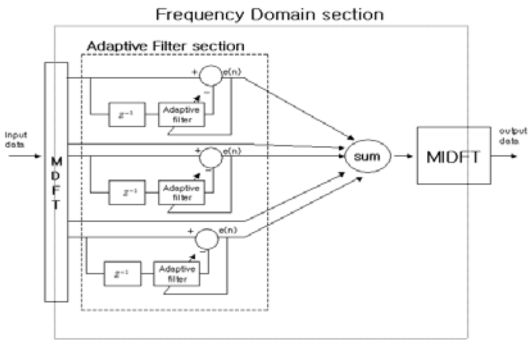


Fig. 1 The Sign algorithm structure

The proposed structure using the modified DFT(MDFT) has the minimum quantity of operations to enable nonblock in order to process from the stable convergence.

In order to improve the convergence speed is applied to Sign algorithm that the input auto-correlation matrix is approximately diagonalized by using the discrete Fourier transform(DFT) and normalized by the time-variable step size algorithm. In General, the normalizing by spectral power effect to improve the convergence speed[2].

If we apply an equation (1) to an analysis algorithm, a coefficient vector is as follows[4],

$$H(n+1) = H(n) + \mu X(n)e(n) \tag{3}$$

Where H(n) is the vector of coefficients with a time n index. X(n) is the primary input vector. μ is a time-variable convergence parameter, e(n) is the error for tracking of reference input d(n) using the X(n). We assume that an input correlation matrix is diagonalized approximately. To decrease a number of operations used the sign method to avoid multiplications instead of an error signal or data value of signal. Before starting the analysis, Let H_{opt} denote the optimal filter coefficient vector given by,

$$H_{OPT} = R^{-1}{}_{xx} R_{xd} \tag{4}$$

Also, define the misalignment vector as[4],

$$V(n) = H(n) - H_{OPT} \tag{5}$$

The Sign algorithm is defined by the adapted LMS for the misalignment vector[5],

$$H(n+1) = H(n) + \mu X(n) sign(e(n)) \tag{6}$$

$$E\{V(n+1)\} = (I - \frac{\mu}{\sigma e(n)}\sqrt{\frac{2}{\pi}} R_{xx}) E\{V(n)\} \tag{7}$$

$$R_{xx} = E\{X(n)X^T(n)\} = \sigma^2{}_\tau I \tag{8}$$

$$R_{xd} = E\{X(n)d(n)\} \tag{9}$$

The filter's properties doesn't changed from the filter output and the error calculation, but it degraded steeply the multiplication operations in the adjustable part of coefficients by calculating to only get the symbol of input signal and error signal in that. If the μ is very small, make the result of real properties. The $\sigma_e(n)$ as LMS error in time-variable n is[6],

$$\mu_s = \mu_L \sqrt{\frac{2}{\pi}} \sqrt{\xi_{\min}} \tag{10}$$

$$\xi\min = E\{d^2(n)\} - R^T{}_{xd}H_{opt} \tag{11}$$

The μ is chosen μ_s to get the mean square error in stable state between the LMS and the SLMS.

$$\mu = \mu_s = \mu_L \sqrt{\frac{2}{\pi}} \sqrt{\xi_{\min}} \tag{12}$$

Where μ_L is a convergence constant of a LMS.

III. Experimental results.

If we calculate the operational number. The multiplication and addition of the MDFT pair is as follows[2].

$$4(\frac{N}{2})\log N - \sum_{r=3}^{\log N} 2^{r-3}(r-2)) - N \tag{13}$$

$$2N\log N - 2\sum_{r=2}^{\log N} 2^{r-2}(r-1) + 2 - \frac{N}{2} \tag{14}$$

As a result, it is shown a few operations against DFT's [6]. The input signals are the normalized step size algorithms using its magnitude pass through the next state. The results of graphics are confirmed some differ between DFT and MDFT.

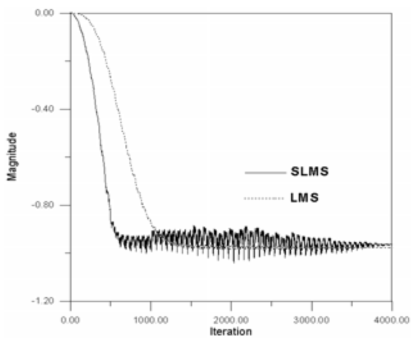


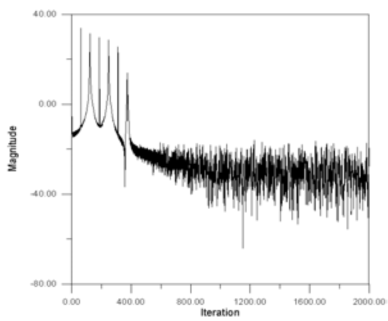
Fig. 2 Comparisons of the convergence speed.

Figure 2 shows the convergence speed between LMS and Sign algorithm. The result is shown from the Sign algorithm convergence speed.

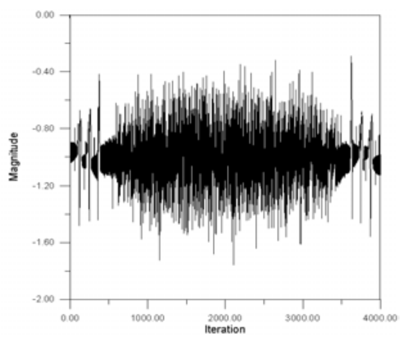
Also, if the μ is very small, there isn't a wide difference between LMS and SLMS in error of the stable state. The performances of the proposed system are confirmed through computer simulation.

The signals that used experiment was added to the noise that had the white Gaussian noise of variance 1 for confirming the noise reduction performance.

The number of samples N was 64, and the data point was 4000. The coefficient of filter degree was 1, and the μ set up 0.005.



(a) The input signal with noise.



(b) Adaptive coefficient signal.

Fig. 3 The waveforms.

Figure 3 shows (a) the input signal waveform with the white Gaussian noise, and (b) the adaptive signal waveform of the applied Sign algorithm in order to remove the noise signals.

Figure 4 shows the output signal using the Figure 3. Then the signal of canceling the noise is displayed. So, the Sign algorithm proposed regenerates exactly the signal from the input signal.

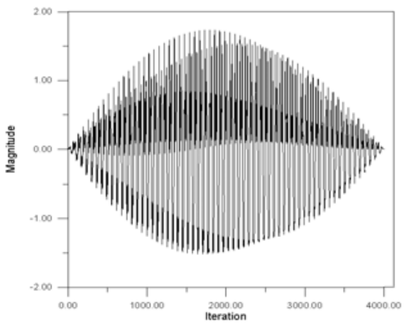


Fig. 4 The output signal waveform.

Figure 5 shows the processing speed in system. The proposed system is processed very faster than nonproposed.

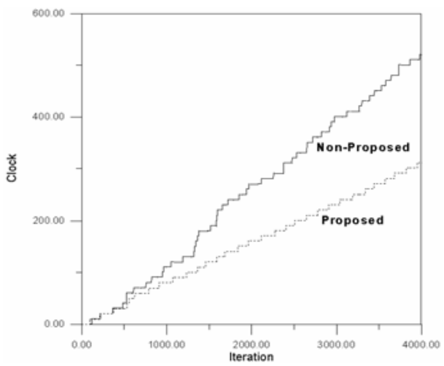


Fig. 5 The processing time estimation.

VI. Conclusion.

In this paper, we are proposed the high speed adaptive noise reduction on the frequency domain using the Sign algorithm, and its performances were confirmed through the simulations. The system proposed is advanced much better former research. It is necessary to method of correc-tion for the error in real states.

V. References

[1] I. Nakanishi, Y. Hamahashi, Y. Itoh, and Y. Fukui. “Performance Improvement of Frequency Domain Adaptive Filter,”in Proc. of ITC-CSCC'99, Sokcho, Korea, pp. 1483-1486, 1998

[2] Isao Nakanishi, Yoshihisa Hamahashi, Yoshio Itoh, Yutaka Fukui, “Frequency Domain Adaptive Noise Reduction using Modified DFT Pair,” in Proc. of ITC-CSCC’99, Sado, Nigata, Japan, pp. 1370-1373, 1999

[3] M. Dentino, J. McCool, and B. Widrow, “Adaptive Filtering in the Frequency Domain,” in Proc. IEEE, vol.66, no.12, pp. 1658-1659, Dec. 1978.

[4] Sung Ho Cho, “Convergence Analysis for Efficient Adaptive Digital Filtering Algorithms and Structures,” A diss. submitted to the faculty of The University of Utah in partial fulfillment of the requirements for the degree of.1989

[5] Metin Akay, “Biomedical Signal Processing”, in chapter10, pp195-251. 1994

[6] S. S. Narayan, A. M. Peterson, and M. J. Narasimha, “Transform Domain LMS Algorithm,” IEEE Trans. on Acoust. Speech & Signal Processing, vol. 31, no. 3, pp. 609-615, 1983.

[7] Dal Hwan Yoon and Chi Ho Lin, “Adaptive Noise Reduction of Biomedical Signals with Improved Multiplication Free Adaptive Digital Filter,” IEICE Transactions on Fundamentals of Electronics, Communic-ations and Computer Sciences, Vol. E86-A, No. 6, pp.1326-1334, June 2003