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Ambient Noise Canceller in Pulmonary Sound Using WHT Transform Domain Adaptive Filter

Wei-Shun Liao[1], Bor-Shyh Lin[1], Bor-Shing Lin[1], Huey-Dong Wu[2], Fok-Ching Chong[1]

[1] Institute of Electrical Engineer, National Taiwan University, Taiwan R.O.C.

[2]Department of Internal Medicine, National Taiwan University Hospital, Taiwan R.O.C.

Abstract - **In the process of signal processing of pulmonary sounds, the cancellation of ambient noise is very important. Because the ambient noise is a wide band signal in frequency domain, it usually uses adaptive noise canceller (ANC) structure to cancel the ambient noise. However, the usually used algorithm, normalized least mean square (nLMS) algorithm, may fail when dealing with the ambient noise due to time-varying system because of its low convergence speed. In this paper we use transform domain adaptive filter (TDAF) with Walsh-Hadamard transform (WHT) to improve the convergence speed. The simulation results show that this structure would cancel the ambient noise more efficiently.**

Keywords **– Pulmonary sound, adaptive filter, noise cancellation, biomedical signal processing, WHT**

I. INTRODUCTION

In the signal processing of pulmonary sound, it is important to remove the ambient noise recoded in pulmonary sound. The frequency band of pulmonary sound with significant characteristics is about 200Hz to 800Hz, and the frequency band of ambient noise would overlap large position of pulmonary sound. It is the reason why the ANC structure is usually adopted [1]. Adaptive algorithms need time to train the filter output and the variation of the outer environment may interrupt converge time. Thus the recording system should have an algorithm that has to converge fast enough so that the ambient noise would be properly kept to a minimum. The most usually used adaptive algorithm is nLMS algorithm, but it has low convergence speed that using it alone might not suitable for recording pulmonary sounds. Therefore, we propose a structure that uses TDAF with WHT to improve the convergence speed, thus the performance of entire adaptive noise canceller is improved.

II. METHODOLOGY

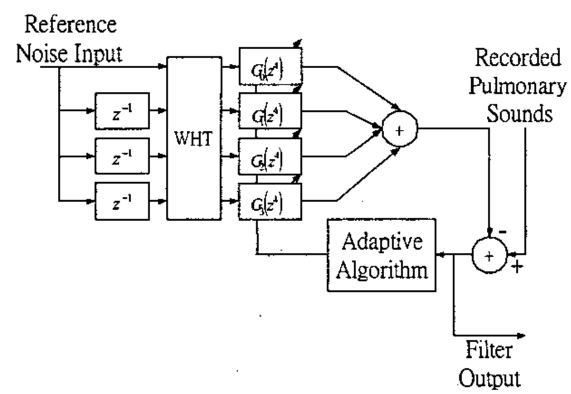

Fig. 1. Structure of TDAF.

The structure of this TDAF is shown in Fig.1 [2]. This structure uses a transform matrix to transform the input into a mathematical domain prior to input to the adaptive filter. From adaptive filter theory, when the degree of the eigen-value spread in the input autocorrelation matrix is great, the convergence speed is slow, when the degree of spread is small, the signal is close to white noise, and the convergence speed will be faster [3].

When the autocorrelation matrix is transformed to the mathematical domain, the eigen-value spread will become smaller. This meant that the input signal is transformed as white noise to ensure faster convergence speed. There are many type of mathematical transform. In this paper WHT is used .

III. RESULTS

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A. Performance Comparison Simulation

We use a pure pulmonary sound pattern for two seconds with sampling frequency of 3200Hz. Next we make a pattern out of a segment of musical voice. This musical voice must undergo a 10 times 8 order MA time variant system for 2 second. The pure pulmonary sound is added to this pattern to create a –6dB SNR composite pattern for the ANC input. The original musical pattern is taken as the reference signal. Table I is the result obtained for TDAF and traditional adaptive filter. The result shows the MSE values in different combinations of step size and tap number. The adaptive algorithm used is nLMS algorithm.

TABLE I MSE of TDAF and Traditional ADF

Tap Number	Step Size	MSE	
		TDAF	Traditional ADF
4	0.01	0.0115	0.0115
	0.02	0.0064	0.0065
	0.03	0.0047	0.0047
8	0.01	0.0069	0.0123
	0.02	0.0041	0.0068
	0.03	0.0033	0.0048
12	0.01	0.0057	0.0124
	0.02	0.0038	0.0068
	0.03	0.0034	0.0049

As the TDAF uses 4 channels, the total filter tap number is a multiple of 4. When the taps on each channel exceed two taps, TDAF outperform traditional structure. The reason is that the convergence speed of TDAF is faster and is effective to deal with environment changes.

B. Real-time Simulation

Fig.2 is the simulation results using real-time recorded pulmonary sound. Its sampling frequency is 3200Hz. The TDAF is with 4 channels, 8 taps, and step size 0.001. The adaptive algorithm used is nLMS. The result obtained for TDAF is fig.2 (a) and for ADF is fig.2 (b). From this figure, you can notice that there is significant noise reduction in TDAF.

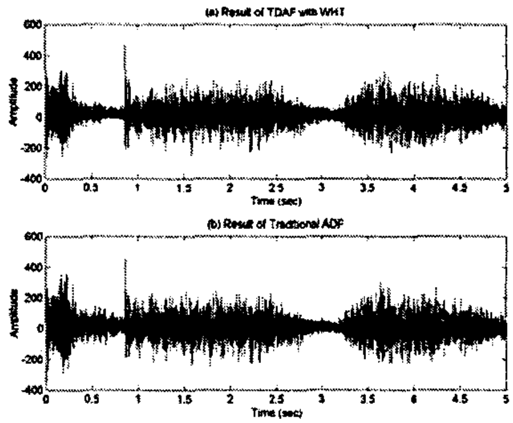

Fig. 2. Output of ambient noise canceller

(a) TDAF output, (b) Traditional ADF output.

IV. CONCLUSION

From the above result, we can conclude that using TDAF structure outperform traditional ADF. As the complexity of computation of WHT is low, it is effective to decrease the circuit cost. Moreover, its high convergence speed enables TDAF to be more adapted to environment changes. TDAF is indeed an effective ambient noise canceller structure.

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