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REAL TIME HEART AND LUNG SOUND SEPARATION USING ADAPTIVE LINE ENHANCER WITH NLMS

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ABSTRACT

The heart sound signal (HSS) separated from the recorded real time raw sound signal (RSS) in a human being. The signal is taken from the different age groups for both male and female. In a real time sound signal separation techniques the desired sound signals are difficult to achieve, now we have separated real time raw lung sound signal into the real heart sound signal. To the proposed technique we have introduced adaptive line enhancer (ALE) with normalized least mean square (NLMS) algorithm which is used to obtain the desired sound signal from real time sound signal and the linear predictive FIR filter are used to detect the other sound signals and the interferences. We have introduced the new adaptive filter algorithm such as the normalized least mean square algorithm to obtain the desired real heart sound signal because the NLMS is one of the factor µ is not constant and also the weights are updated for each iteration. So we get the expected the real heart signal without interferences in a real time recorded sound signal. The proposed system is implemented and in the results we have considered some parameters to find the error rate of the desired sound signal (DSS), signal to noise ratio (SNR) and computational time are verified using Matlab 2010.

Keywords: *Heart sound signal (HSS), Lung sound signal (LSS), Adaptive line enhancer (ALE), Normalized Least mean square (NLMS), Finite impulse response filter (FIR).*

1. INTRODUCTION

The ALE is the most effective architecture for separating two signals. The input signal is used to separate the HSS from LSS and the LSS control in the both combined HSS-LSS signal and the input signal of the adaptive line enhancer within a controllable level. In real time signal through lacking of ALE can be used to separate the signals HSS-LSS in a single channel [1]. The achievement of heart sound (HS) and lung sounds (LS) separation is based on the adaptive filtering system. In a desired signal the normal linear filtering approach is not able to separate the two sounds. Efficient architecture adaptive line enhancer (ALE) will help for signal tracking and noise elimination capability between the two signals [2],[18].The noise less heart sound signals help us to find various diseases and identifies the exact problem in a respiratory system. But during the real time Heart sound recording system the source of interference occurred, so this kind of problem can be avoided using the adaptive line enhancer with adaptive algorithm. Most widely used architecture ALE with least mean square (LMS) adaptive algorithm is used to eliminate the interferences but this kind of adaptive algorithms is not sufficient to give the desired output signal, because the co-efficient length and the processing time delay is very high [3][19]. The single input channel used to the adaptive line enhancer and the adaptive filters eliminate the interferences in the output signal. The de-correlated input signal and the noise are removed from the device output [4],[18],[21]. Separating the two signals, breath and cardiac sounds several approach is used to solve the problem in terms of reduced interferences in a desired output signal using several approaches. The adaptive algorithm is used to evaluate the real signal recorded in both breath and cardiac sounds in a human and also in a stationary background noise [5]. During the real time sound signal recording the source of interference is unavoidable but the separation of two sound signal is very challenging task because the output heart sound signal is very useful to find the diseases for the respiratory specialists and cardiologists [6],[17]. The blind source extraction is introduced to separate the heart

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sound and lung sound signal from the recorded input signal using the cyclic correlation matrix but in this technique a certain range of frequency reduces the interference [7]. Observing the heart sound signal from the chest wall of a human always have some interference in a recorded signal but the empirical mode decomposition technique is used to reduce the interference of the desired output signal. The EMD technique is used for both time and frequency domain analysis to give the desired output sound signal [8]. The variable gain amplifier is used for pre-processing compression stage in adaptive filter function. The least mean square algorithm helps to remove the heart sound interferences in adaptive filter algorithm [9]. To eliminate the interferences we use the reference signal when performing adaptive filtering. In [10],a proposed adaptive LMS algorithm based on the single recording. An ALC-RLS based architecture is used to separate or eliminate the LS from HS, though the experimental results are suffer from high computational load. LS reduced from HS using the Time-Frequency (TF) filtering [11], [12], and [13]. The above mentioned technique used to achieve the effective desired output sound signal and it is very efficient [12]. The major aspect of this paper is used to implement for first time, the ALE architecture for separation of HSS in LSS. This technique is used to consider the various parameters such as, spectral estimation, the objective of this paper is to use the ALE to mitigate the HSS in LSS and LMS is done for the first time. This technique has found to be applied in signal to noise ratio (SNR), mean square error (MSER),spectral estimation, frequency estimation and delay (D)[14] [15] [16].

2. ADAPTIVE LINE ENHANCER (ALE) ARCHITECTURE.

First architecture Adaptive Line Enhancer was introduced by Widrow et al. [14], and the adaptive filters with delay element implements is called Adaptive Line Enhancer. ALE is used to eliminate the external interferences and the lung sound from the desired heart sound. The below figure shows the architecture of the adaptive line enhancer (ALE) with normalized least mean square (NLMS) algorithm involved in the basic working principle of adaptive filter system.

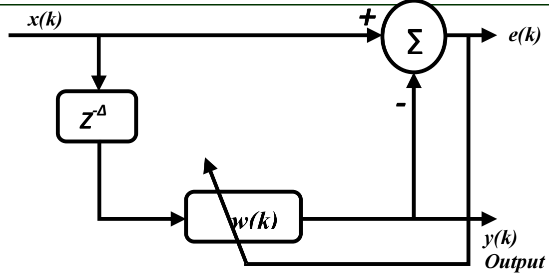

Figure1. Architecture of Adaptive line enhancer (ALE).

The ALE consists of weighted linear predictive FIR filter. In a ALE system the delayed input signal occur with well known normalized least-mean-square (NLMS) adaptive algorithm. The time domain analysis is as follows

$$x(k) = h(k) + \xi_0(k) + e(k) \qquad (1)$$

Where, k is the periodic sample time, $h(k)$ is the input periodic signal and $j(k)$ is the Gaussian noise signal, $\eta_0(n)$ is the narrow band noise at any time instance k, the output $y(k)$ is defined as:

$$y(k) = \sum_{l=0}^{L-1} w_k(k) x(k - l - \Delta) \qquad (2)$$

Where, Δ is the sampling interval or prediction distance, L is the filter length, and w_k is the filter weight or coefficients. To estimate the h(k) give the delay Δ exceeds the correlation time of $\xi_0(k)$ in output y(k). Choose the delay Δ should be equal to a interval for which the autocorrelation function of $\xi_0(k)$, $z_n(\tau)$ can be measured small relative to $z_n(0)$. To minimize the mean square error (MSE), choose the adaptive filter weights $w_k(l) = 0,......L-1$ and the MSE and it is defined as:

$$\xi = E[(x(k) - y(k))^2] \qquad (3)$$

Now, the input signal x(k)and its delayed element, $x(k-\Delta),......, x(k-\Delta-L+1)$ is the original periodic signal h(k), the mean square error is reduced when y(k) = x(k). In this manner the filter coefficients can be advertised in a normalized LMS algorithm so it reduces computational complexity and gives more accuracy.

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3. NORMALIZED LEAST MEAN SQUARE ALGORITHM (NLMS).

When least mean square (LMS) algorithm is modified from frequency normalization it is known as the Normalized Least Mean Square (NLMS) algorithm. The NLMS adaptive filter is used to remove the interferences from the input signal such as Gaussian white noise to obtain a output signal. The advantages of NLMS algorithm over the LMS algorithm are because of certain characteristics like faster computation and it's ability to reduce the error between the signals. In figure 2 shows the architecture of NLMS adaptive filter is shown below.

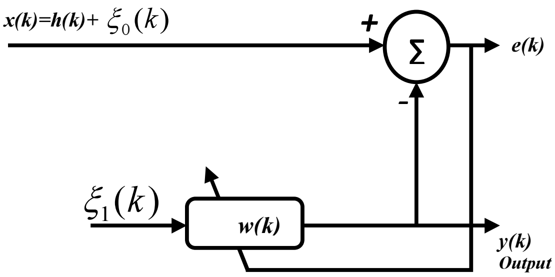

Figure2. Adaptive filter architecture.

In a above figure, *x(k)* is the primary input signal having the recorded heart sound signal *h(k)* and Gaussian white noise signal $\xi_0(k)$ and its components. The input to the NLMS filter is $\xi_1(k)$ which is estimated the filter coefficients or weights *w(k)*, to be close to $\xi_0(k)$, so that only the output sound signal *y(n)*is obtained as the error signal *e(k)* is the final output.

The NLMS algorithm considers the standard LMS recursion, for which we select a variable step size parameter $\mu(k)$. This parameter will be minimized using the updated filter tap weights, *w(k+1)* and the current input vector *x(k)*.

$$\mu(k) = \frac{1}{x^T(k)x(k)+\psi} \quad (4)$$

The variable step size μ(k) is substituted into the standard LMS recursion replacing μ, resulting in the following .

$$w(k+1) = w(k) + 2\mu e(k)x(k)$$

$$w(k+1) = w(k) + \frac{1}{x^T(k)x(k)} e(k)x(k) \quad (5)$$

A small modification of the standard NLMS algorithm is detailed above. The value of ψ is maintained on a small positive constant in order to avoid division by zero when the values of the input vector are zero.

$$w(k+1) = w(k) + \frac{1}{x^T(k)x(k)+\psi} e(k)x(k) \quad (6)$$

Finally The filter tap weights or coefficient *w(k+1)* are updated and every iteration is defined as,

$$w(k+1) = w(k) + \mu(k)e(k)x(k) \quad (7)$$

An error signal is calculated as the difference between the input signal x*(k)* and the desired filter output and *y(k)* is defined as,

$$e(k) = h(k) - y(k) \quad (8)$$

The step size value $\mu(k)$ is calculated from the input signal *x(k)* is defined as,

$$\mu(k) = \frac{\mu}{x^T(k)x(k)+\psi} \quad (9)$$

The output of the adaptive filter is calculated based on the length of filter with delay element Δ is defined as,

$$y(k) = \sum_{l=0}^{L-1} w_k(k)x(k-l-\Delta) \quad (10)$$

From the equ-10, we get the output of the ALE – NLMS adaptive filter algorithm i.e. the desired output heart sound signal. In this equation the input signal *x(k)* with the filter length of *l* and the delay element Δ with the filter coefficient or weight computed up to the length of filter *L-1* to obtain the desired output. The coefficient weights are updated using the equ-7 to implement the every iteration at any stage of the adaptive filter. Typically, the MSE for the NLMS-ALE converges geometrically with a time constant $\tau_{NLMS-ALE}$ as:

$$\tau_{NLMS-ALE} \approx \frac{1}{4\mu\lambda_{\min}} \quad (11)$$

An expression for the signal to noise ratio (SNR) gain due to processing by the ALE for sinusoids with white noise has been given in the expression and it is simplified as,

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$$\frac{SNR_{out}}{SNR_{in}} = \frac{1}{\frac{2}{L} + \mu\xi_{\min} L(1+SNR_{in})} \qquad (12)$$

The real time recorded sound signal de-correlated with the noise components using the delay element in a adaptive filter x(k) with the phase delay between the periodic signals. To eliminate the noise component using the phase difference of the periodic signal is adjusted so that they cancel the other summing points of the adaptive filter. The input data alone is used to minimize the error signal combined with the other interferences.

4. RESULTS.

In a real time signal simulation, we get the noise less heart sound signal using the ALE – NLMS architecture. The proposed architecture provides the sufficient output heart sound signal to find the particular diseases in respiratory system. The entire architecture design simulated and the output verified using the MATLAB.

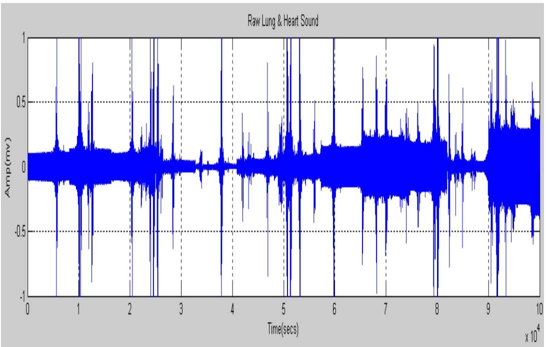

Figure 3. Shows the real time recorded sound signal. It contains the both heart and lung sound signal with the sources of interferences.

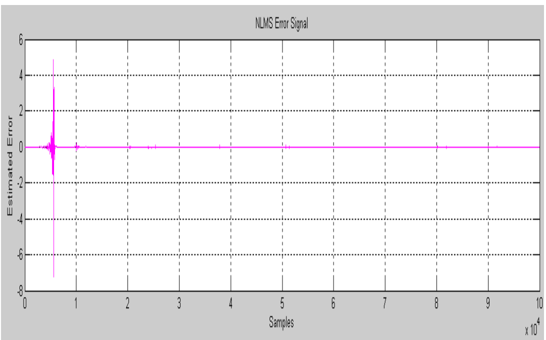

Figure 4. Shows the desired output signal e(n). The output is seperated from the input siganl x(n) to the excepted normalized LMS filter algorithm output y(n).

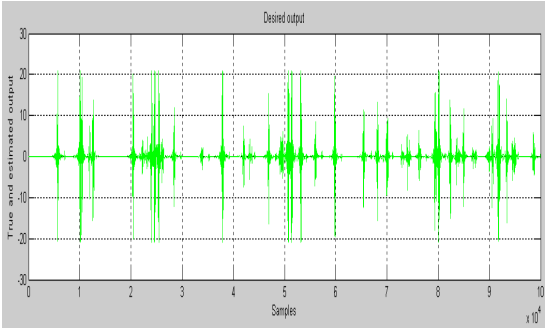

Figure 5. Shows the desired output signal of the ALE - normalized LMS adaptive filter algorithm y(n). The output depends on the each iteration update, the filter coefficient and the length of the adaptive filter.

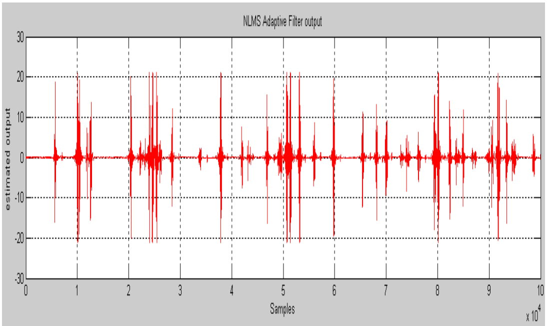

Figure 6. Shows the output of the normalized filter output with the filter coefficient updated w(k+1).

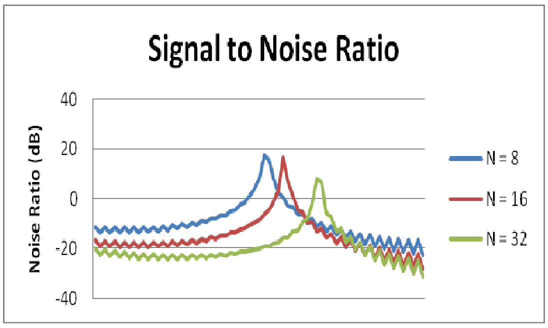

Figure 7. Shows the compression between the different FIR filter tap order such as N=8,16&32. The minimum signal-to-noise ratio achieved using the 32 tap order in a adaptive filter.

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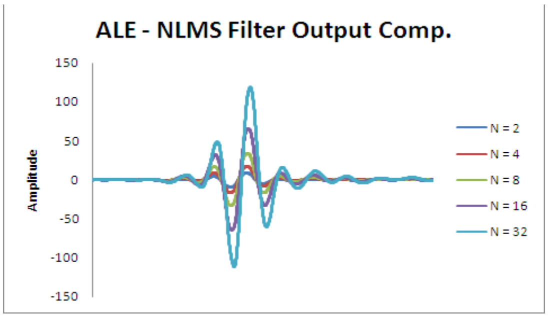

Figure 8. Shows the output compression of ALE – NLMS in different adaptive filter tap order.

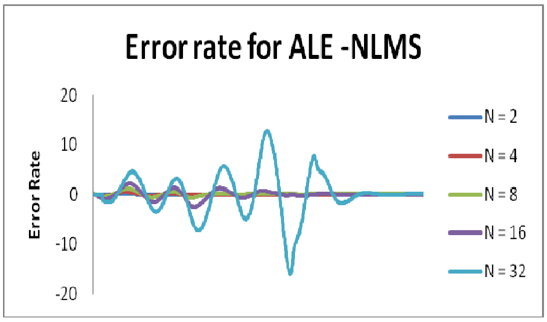

Figure 9. Shows the mean square error (MSE) rate at different filter order.

Table I. Analysis of the noise level in different FIR filter tap order (N=2, 4,8,16 and 32).

Performance Analysis for ALE – NLMS Algorithm			
Filter Order (N)	**SNR**	**MSE**	**Time (sec)**
N = 2	3.0403e+002	16.3226	2.864222
N = 4	84.0255	11.5418	2.655170
N = 8	66.1555	8.1613	2.949751
N= 16	68.8313	5.7709	2.798186
N = 32	18.7299	4.0807	2.554970

5. CONCLUSION.

A real time signal taken from the digital stethoscope from different age group of humans. The proposed architecture of ALE-NLMS adaptive algorithm provides the desired output heart sound signal. In this architecture the delay component Δ is adjusted, length of the filter l and the order of the adaptive filter (FIR filter) are changed to separate the noise less heart sound signal. Finally we have concluded that adaptive filter using the 16th filter orders provide more efficient output compared to the other filter order. The normalized LMS adaptive filter algorithm is used to minimize the computation complexity, time and minimum allowable signal-to-noise ratio (SNR). The results are measured sensitively by adjusting the parameters and by clear verification.

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