

Research Article

Application of Adaptive Line Enhancer with LMS to Separate Heart Sound Signal from Lung Sound Signal at Real Time

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Abstract: This study presents a technique for separation of Heart Sound Signal (HSS) from Lung Sound Signal (LSS) using Adaptive Line Enhancer (ALE) with Least Mean Square (LMS) algorithm with real time recorded sound signal. While recording lung sounds, an incessant noise source takes place owing to heart sounds. This noise source severely contaminates the breath sound signal and interferes in the analysis of lung sounds. The proposed system is applied with different filter order and the results show the error rate of the Desired Sound Signal (DSS), Signal to Noise Ratio (SNR) and execution time.

Keywords: Error rate, execution time, heart sound signal, LMS algorithm, lung sound signal, signal to noise ratio

INTRODUCTION

Heart sounds interfere with lung sounds in a manner that obstructs the potential of respiratory sound analysis in terms of analysis of respiratory disease. Lung sounds are created by vortical and turbulent flow (Blake, 1986) within lung airways during inspiration and expiration of air (Hardin and Patterson Jr., 1979). Lung sounds recorded on the chest wall represent not only produced sound in lung airways but also the effects of thoracic tissues and sound sensor characteristics on sound conveyed from the lungs to a data acquisition system (Vovk *et al.*, 1995).

Lung sounds show a power spectral density that is broadband with power decreasing as frequency increases (Gavriely *et al.*, 1981). The logarithm of amplitude and the logarithm of frequency are almost linearly related in healthy subjects provided that the signals do not have adventitious sounds. As the flow in lung airways increases, sound intensity increases and several mathematical relations between lung sounds and airflow have been suggested (Hossain and Moussavi, 2002; Gavriely and Cugell, 1996). It is important to note that inspiratory and expiratory lung sounds be unlike in terms of both amplitude and frequency range. At comparable flows, inspiratory lung sounds will have more intensity than expiratory sounds (Manecke Jr. *et al.*, 1997; Gavriely *et al.*, 1995).

Heart sounds are created by the flow of blood into and out of the heart and by the movement of structures involved in the control of this flow. The first heart sound results when blood is pumped from the heart to the rest of the body, during the latter half of the cardiac cycle and it is comprised of sounds resulting from the

rise and release of pressure within the left ventricle along with the increase in ascending aortic pressure (Luisada, 1964).

After blood leaves the ventricles, the simultaneous closing of the semilunar valves, which connect the ventricles with the aorta and pulmonary arteries, causes the second heart sound. The electrocardiogram signifies the depolarization and repolarization of heart muscles during each cardiac cycle. Depolarization of ventricular muscles during ventricular contraction results in three signals known as the Q, R and S-waves of the electrocardiogram (Sherwood, 2001). The first heart sound immediately follows the QRS complex. In health, the last 30 -40% of the interval between successive R-wave peaks contains a period that is void of first and second heart sounds.

Characteristics of heart sound signals have been assessed in terms of both intensity and frequency. Though peak frequencies of heart sounds have been shown to be much lower than those of lung sounds (Arnott *et al.*, 1984), comparisons between lung sound recordings acquired over the anterior right upper lobe containing and excluding heart sounds (Gnitecki *et al.*, 2003) show that PSD in both cases is maximal below 150 Hz.

Separating the two signals, breath and cardiac sounds and several methods are used to solve the problem in terms of reduced interferences in a desired output signal. The adaptive algorithm is used to find the real signal recorded in both breath and cardiac sounds in a human and also in a stationary background noise (Rudnitski, 2001).

During the real time sound signal recording the source of interference is unavoidable but the separation

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of two sound signals is very challenging task because the output heart sound signal is very useful to find the diseases for the respiratory specialists and cardiologists (Pourazad *et al.*, 2005). The blind source extraction is introduced to separate the heart 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 (Ghaderi *et al.*, 2009).

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 method is used to decrease the interference of the desired output signal. The EMD method is used for both time and frequency domain analysis to give the desired output sound signal (Mondal *et al.*, 2011). The variable gain amplifier is used for pre-processing compression stage in adaptive filter function. The least mean square algorithm helps to eliminate the heart sound interferences in adaptive filter algorithm (Guangbin *et al.*, 1992). 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 is presented by Sathesh and Muniraj (2014).

This study presents a technique for separation of Heart Sound Signal (HSS) from Lung Sound Signal (LSS) using Adaptive Line Enhancer (ALE) with Least Mean Square (LMS) algorithm with real time recorded sound signal obtained using digital stethoscope and its details are given in Appendix. The results show the error rate of the Desired Sound Signal (DSS), Signal to Noise Ratio (SNR) and execution time for different filter order.

METHODOLOGY

Adaptive Line Enhancer (ALE) architecture: First architecture Adaptive Line Enhancer was introduced by Widrow *et al.* (1975) and the adaptive filters with delay element implements are called Adaptive Line Enhancer. ALE is used to eliminate the external interferences and the lung sound from the desired heart sound. Figure 1 shows the architecture of the Adaptive Line Enhancer (ALE) with Least Mean Square (LMS) algorithm involved in the basic working principle of adaptive filter system.

Figure 1 shows the main components of ALE filter as follows:

- Input of the primary signal $x(n)$
- Input of reference signal $r1(n)$
- Filter output $y(n)$
- Finally the estimation error $e(n)$

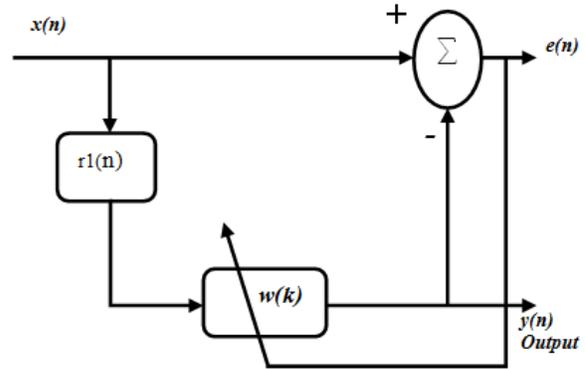


Fig. 1: Method of adaptive noise cancellation

The primary input signal consists of the desirable signal $L(n)$ and noise signal $r0(n)$ as shown in Eq. (1):

$$x(n) = L(n) + r0(n) \quad (1)$$

The reference input $r1(n)$ is moderated to a complex signal, which contains HS and generated signal NI simulated by WGN. Therefore, the generated signal NI accompanied with recording process of LS, which is generated from instruments and surrounding environment of LS recording. The generated NI is based on the methods presented by Tsalaila *et al.* (2008) and Karagiannis and Constantinou (2010). The methods proposed temporally to generate WGN on real quasi period. It is added to the synthetic HS in various magnitude scales in order to obtain synthetic bio-signals corrupted by WGN at a wide range of Signal to Noise Ratios (SNR). Only SNR higher than 0 dB is considered, because significant noise level distorts ECG in such a degree that low magnitude complexes are not identifiable. However, this method is evaluated on both synthetic periodic signals of the known HS period combined with WGN on real quasi period. Equation (2) is the reference modulated HS signal, $h(n)$ and generated WGN, $d(n)$:

$$r1(n) = h(n) + d(n) \quad (2)$$

The output of the Adaptive filter, $e(n)$, is the Minimum MSE (MMSE) estimate of the component of the primary signal as shown in Eq. (3):

$$e(n) = x(n) - y(n) = [L(n) + r1(n)] - y(n) \quad (3)$$

where, $y(n)$ is the structure of the FIR, which can be represented in Eq. (4):

$$y(n) = \sum_{k=0}^L wk \times r1(n - k) \quad (4)$$

The weights are updated with each iteration, n , based on feedback of the estimation error to the adaptive filter unit.

LMS algorithm: The LMS algorithm choice is based on its simplicity and it does not require measurements of the pertinent correlation function. It also requires matrix inversion. It uses Eq. (3) to get square error $e^2(n)$, which is the Square Error (MSE) between the output $y(n)$ (equal to $w_x(n)$) and the reference signal $r_1(n)$, as given Eq. (5):

$$e^2(n) = [r_1(n) - \sum_{k=0}^L w_k \times r_1(n - k)]^2 \quad (5)$$

Shortly, it is possible to calculate the filter coefficient vector for each iteration k having information about the previous coefficients w_k and gradient μ , multiplied by a constant, as shown in Eq. (6):

$$w_k(n + 1) = w_k(n) - \mu[-\nabla_k] \quad (6)$$

The gradient vector is defined by Eq. (7):

$$\nabla_k = \frac{\partial\{e^2(n)\}}{\partial w_k(n)} \quad (7)$$

This equation with Eq. (6) Lead to get Eq. (8):

$$w_k(n + 1) = w_k(n) - \mu e(n) \times (n - k) \quad (8)$$

The coefficient μ is constant, which must be chosen for quick adaptation without losing stability.

The filter is stable if μ satisfies the following condition:

$$0 < \mu < 1/(10 \times L \times P_{xx}) \quad (9)$$

where,

L = The filter order

P_{xx} = The power of the input signal (Garcés, 2007)

The LMS algorithm is initiated with an arbitrary value $w(0)$ for the weight vector at $n = 0$. The successive corrections of the weight vector eventually leads to the minimum value of the mean squared error.

RESULTS AND DISCUSSION

In a real time signal simulation, the noise less heart sound signal using the ALE-LMS architecture is obtained. 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 2 shows the real time recorded sound signal. It contains the both heart and lung sound signal with the sources of interferences.

Figure 3 shows the desired output signal $e(n)$. The output is seperated from the input signal $x(n)$ to the excepted LMS filter algorithm output $y(n)$.

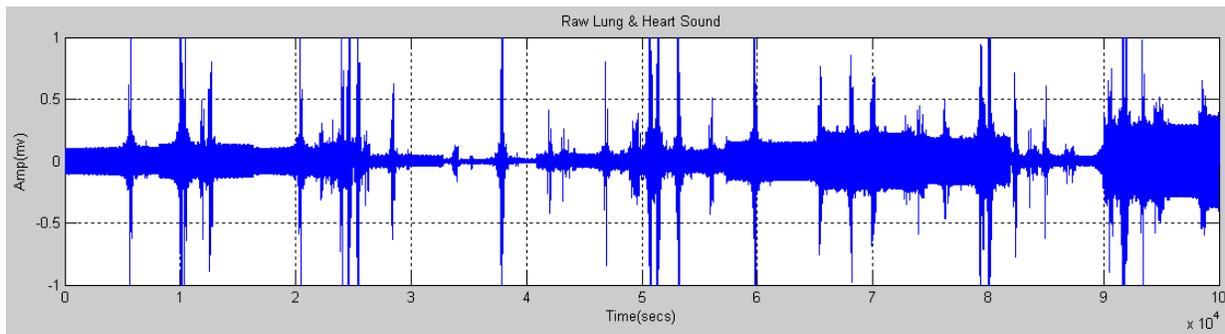


Fig. 2: The real time recorded sound signal

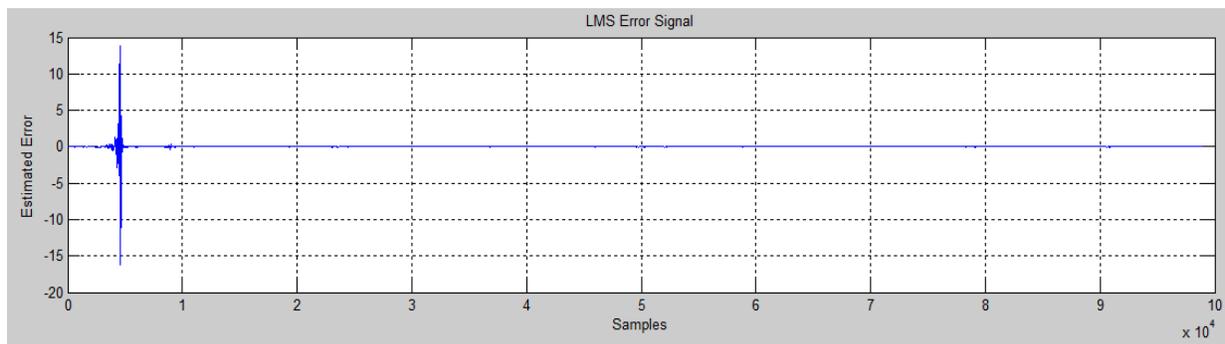


Fig. 3: The desired output signal $e(n)$

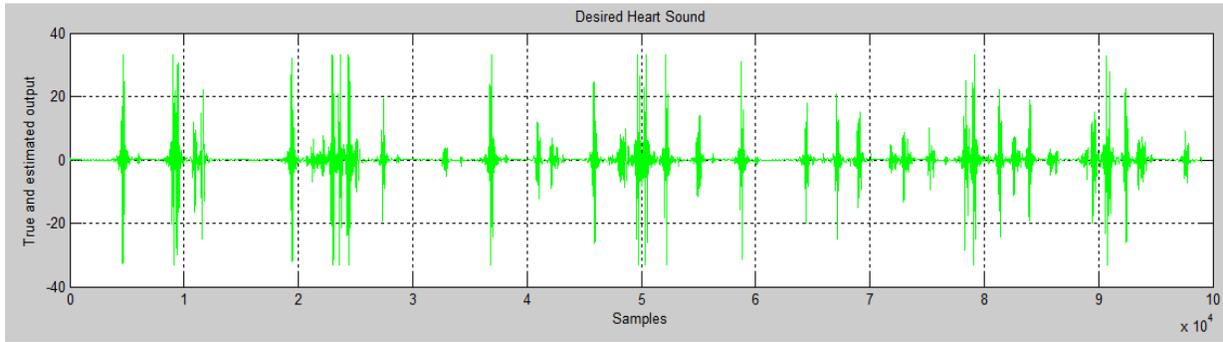


Fig. 4: The desired output signal of the ALE-LMS adaptive filter algorithm $y(n)$

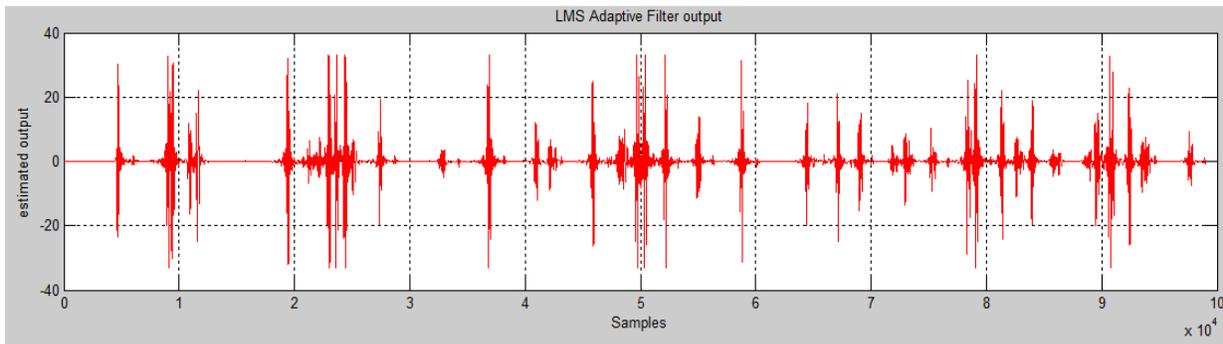


Fig. 5: The output of the filter coefficient updated $w(k+1)$

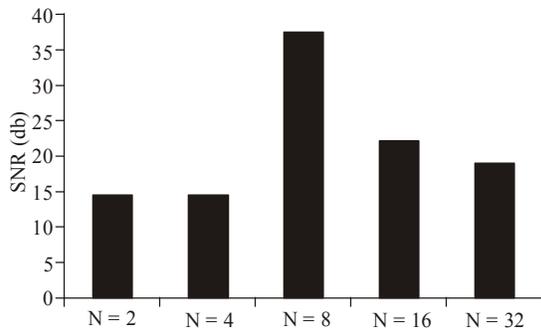


Fig. 6: Compression between the different FIR filter tap order such as $N = 2, 4, 8, 16$ and 32 , respectively

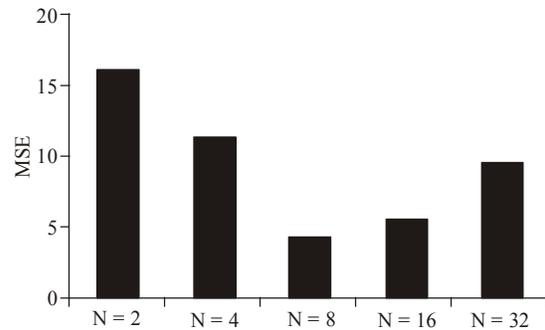


Fig. 8: MSE rate at different filter order

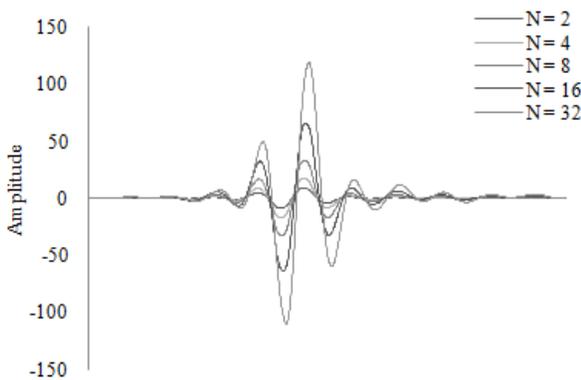


Fig. 7: The output compression of ALE-LMS in different adaptive filter tap order

Figure 4 shows the desired output signal of the ALE-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 5 shows the output of the filter coefficient updated $w(k+1)$. Figure 6 shows the compression between the different FIR filter tap order such as $N = 2, 4, 8, 16$ and 32 , respectively. The minimum signal-to-noise ratio achieved using the 8 tap order in an adaptive filter.

Figure 7. Shows the output compression of ALE-LMS in different adaptive filter tap order.

Figure 8 shows the Mean Square Error (MSE) rate at different filter order.

Table 1: Analysis of the noise level in different FIR filter tap order (N = 2, 4, 8, 16 and 32, respectively)

| Filter order | Signal to noise ratio | M.S.E. | Time (sec) | Avg. error |
|--------------|-----------------------|---------|------------|------------|
| N = 2 | 14.92170 | 16.5416 | 2.951093 | 0.001471 |
| N = 4 | 15.11387 | 11.6967 | 2.456030 | 0.000422 |
| N = 8 | 36.26540 | 4.1354 | 2.884758 | 0.000137 |
| N = 16 | 23.00630 | 5.8483 | 3.130054 | 0.002311 |
| N = 32 | 18.81720 | 8.2708 | 2.949939 | 0.003187 |

M.S.E.: Mean Square Error; Avg.: Average

The analysis of the noise level in different FIR filter tap order (N = 2, 4, 8, 16 and 32, respectively) is provided in Table 1.

CONCLUSION

A real time signal taken from the digital stethoscope from different age group of humans. The proposed architecture of ALE-LMS 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 8th filter orders provide more efficient output compared to the other filter order. The 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.

APPENDIX

Digital stethoscope details:

Hardware used-digital stethoscope
 Sampling frequency used-44.1 khz
 Open source software used-Thinklabs phonocardiography

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