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AN INSIGHT INTO ADAPTIVE NOISE CANCELLATION AND COMPARISON OF ALGORITHMS

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ABSTRACT

The principle of adaptive noise cancellation is to acquire an estimation of the unwanted interfering signal and subtract it from the corrupted signal. Adaptive Noise Cancellation technique is an approach for powerful noise cancellation. In this paper the noise cancellation is performed using adaptive noise cancellers using an adaptive finite impulse response (FIR) filter are presented for the estimation of a transfer function of a noisy channel in a communication link. Optimizations of the Least Mean Square (LMS) versions and Normalized LMS (NLMS), Weiner algorithms are also used to adapt the filter coefficients of the estimated transfer functions in order to minimize the effect of background noise effectively & analyzed their performances with respect to their estimation of weights.

Keywords: Finite Impulse response filters (FIR), Infinite Impulse response filters (IIR), Least-mean square algorithm (LMS), Normalized least mean square algorithm (NLMS), WIENER algorithm

1. INTRODUCTION

A Digital communication system consists of a transmitter, receiver and channel connected together. Typically the channel suffers from two major kinds of impairments: Inter symbol interference and Noise. Adaptive noise cancellation a specific type of interference cancellation uses cancellation of noise by subtracting noise from a received signal, an operation which is controlled in a manner which is adaptive for the purpose of improved signal to noise ratio (SNR). It is basically a dual-input, closed loop adaptive control system as shown in Figure 1.

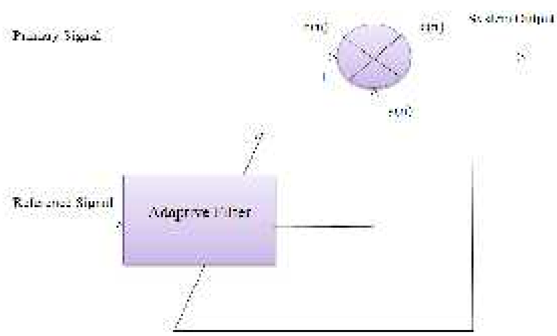

Figure 1: Noise Cancellations

Here the adaptive filter is used to cancel unknown interference contained in a primary signal, with the cancellation being optimized in smarter sense. The basic signal serves as the required response of the filter. The reference signal is the input to this filter. This paper studies and compares the performance analysis of three adaptive algorithms in noise cancelling. Simulations done based on different types of signals mixed with various types of noise & their response after application of three adaptive filters and are presented.

2. ADAPTIVE NOISE CANCELLER

Adaptive noise canceller is a dual input, closed loop adaptive feedback system. By estimating the noise signal and subtract it from corrupted signal. In simple form the two inputs to cancellation system are primary sensor and reference sensor.

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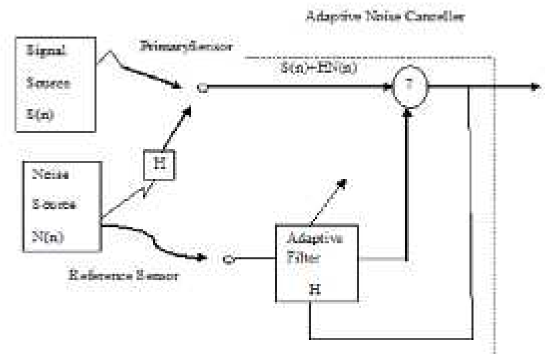

Figure 2: Adaptive Noise Canceller

As shown in above figure 2 the primary sensor not only records the signal from the required signal source but also picks up a delayed and/ or filtered noise originating from the noise source.

Recorded signal at primary sensor = S (n) +HN (n)

Let V (n)= HN (n) represents the noise signal at primary sensor and assume the desired signal and noise signal are uncorrelated with each other such that,

$$E [S (n) V (n-m)] = 0 \qquad \text{for all } m$$

Noise signal N (n) originated at the reference sensor is uncorrelated with signal S (n) i.e.

$$E [s (n) N (n-m)] = 0 \qquad \text{for all } m$$

However N (n) is correlated with delayed and filtered noise V (n) or HN (n) at the primary sensor output in an unknown way such that,

$$E [V (n) N (n-m)] = p (m) \qquad \text{for all } m$$

Where p(m) is unknown-correlation for lag m.

$$S^{*} (n) = S (n) + HN (n) - H^{*}N (n) = S (n) + (H-H^{*}) N (n) = e (n) \qquad (1)$$

From equation, the essential noise component is $(H-H^{*})$ N (n). This term can be minimized if $H = H^{*}$, which in turn leads to maximization of system output signal-to- noise (SNR) ratio.

The adaptive noise cancellation system and its effectiveness depend on the following important factors:

i. The signal and noise at the output of the primary sensor are uncorrelated.
ii. The noise signal recorded at the reference sensor is highly correlated with the noise component in the primary sensor output.
iii. The desired signal component in the primary sensor output is undetectable a reference sensor.

After examining the rudimentary operation of noise cancellation device different types of algorithms which are required to implement the noise cancellation problem are presented in [1]. This type of noise cancellation which makes use of DSP's are known as adaptive filtering problem [1]. The different kinds of adaptive filter algorithms are LMS, NLMS and WIENER. These algorithms are discussed in the subsequent chapters.

3. ADAPTIVE FILTERS

By this we can define the term "adaptive filter" as the filter whose characteristics can be modified or altered to achieve an objective by automatic adaptation or modification of the filter parameters [2].

Adaptive filters are often realized as a set of program instructions running on an arithmetical processing device such as a microprocessor or a DSP chip or a set of logical operations implemented in a field programmable gate array (FPGA) or in a semi-custom or custom VLSI integrated circuit, the fundamental operation of an adaptive filter can be characterized independently of the specific physical realization that it takes [3].

The adaptive noise cancelling concept, an alternative method which estimating signals corrupted by interference or additive noise, where this method uses a primary input which contains the corrupted signal and a reference input having noise correlated in an unknown way with the primary noise is presented in [4].

The problem of noise cancellation and arrhythmia detection in ECG, using adaptive filtering techniques which are computationally simplified is presented in [5].

4. CLASSIFICATION OF ADAPTIVE FILTERS

The two most fundamental types of adaptive filters are:

a) Finite Impulse response filters (FIR)
b) Infinite Impulse response filters (IIR)

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5. FINITE IMPULSE RESPONSE FILTERS (FIR)

FIR filter get their name from naturally enough- the way they respond to their impulse. It is an input of value 1 lasting just to be sampled only once and only once of the response of the filter is finite then we say that filter is FIR filter. From practical point of view, finite response means that, when a unit impulse is given to a filter, the output response should return to zero after some time[2.]

We can write output signal y (n) as,

$$y(n) = \sum_{i=0}^{L-1} w_i(n)x(n-i) \qquad (2)$$

$$y(n) = w_0(n)\,x(n-0) + w_1(n)\,x(n-1) + w_2(n)\,x(n-2) \ldots\ldots w_{L-1}(n)\,x(n-L+1) \qquad (3)$$

$$y(n) = W^T(n)\,X(n) \qquad (4)$$

Where,

W (n) = the impulse response values of filter at time n.

X (n) = [x (n), x (n-1)… x (n-L+1] ^{T} denotes input signal vector and T denotes the vector transpose [2].

Where, h (n) = is the transfer function

H (n) = transfer function converted into frequency form

y (n) = filter output

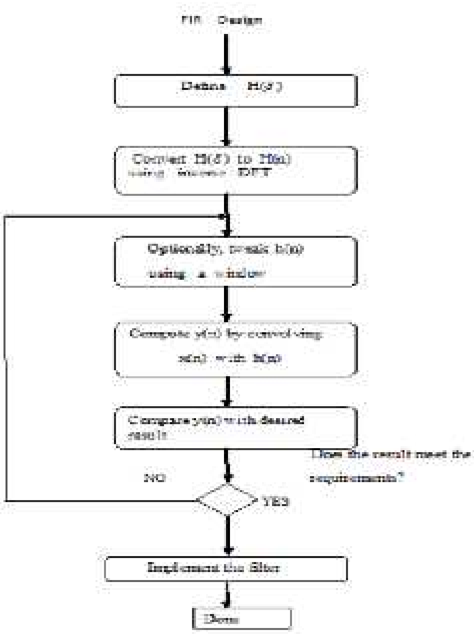

Figure 3: Algorithm for FIR filter

6. ALGORITHMS

The three famous algorithms are:

1. Weiner algorithm
2. LMS algorithm
3. NLMS algorithm

6.1 The Wiener Solution

For the FIR filter structure, the coefficient values in W (n) that minimize J_{MSE} (n) are well-defined if the statistics of the input and desired response signals are known. The formulation of this problem for continuous-time signals and the resulting solution was first derived by Wiener. Hence, this optimum coefficient vector W_{MSE} (n) is often called the Wiener solution to the adaptive filtering problem [6].

To determine W_{MSE} (n), we note that the function J_{MSE} (n) in is quadratic in the parameters $\{w_i(n)\}$, and the function is also differentiable. Thus, W_{MSE} (n) can be found from the solution to the system of equations [6]

$$\frac{\partial J_{MSE(n)}}{\partial w_i(n)} = 0, \qquad 0 \le i \le L-1 \qquad (5)$$

$$\frac{\partial J_{MSE}(n)}{\partial w_i(n)} = E\left\{ e(n)\frac{\partial e(n)}{\partial w_i(n)} \right\}$$

$$= -\left(E\{d(n)x(n-i)\} - \sum_{j=0}^{L-1} E\{x(n-i)x(n-j)w_j(n)\} \right).$$

The matrix R_{xx}(n) and vector P_{dx}(n) are defined as

$$R_{xx}(n) = E\{x(n)X^T(n)\} \qquad (6)$$

$$P_{dx}(n) = E\{d(n)X(n)\}, \qquad (7)$$

We can combine these two terms to obtain the system of equations in vector form as

$$R_{XX}(n)W_{MSE}(n) - P_{dX}(n) = 0$$

Where ‾ is the zero vector. As long as the R_{xx}(n) matrix is invertible, the optimum Wiener solution vector for this problem is

$$W_{MSE}(n) = R_{XX}^{-1}(n)P_{dX}(n) \qquad (8)$$

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A detailed discussion on the Wiener filter's quantitative performance behavior in the context of reduction of noise is given in [7].

6.2 Least Mean Square Algorithm:

LMS (Least mean square), an adaptive algorithm, uses a gradient-based method providing steepest decent. It uses the estimates of the gradient vector from the available data.

LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the mean square error which is of minimum value. LMS algorithm is relatively simpler compared to other algorithms; it does not require correlation function calculation nor does it require matrix inversions [3].

Consider a Uniform Linear Array (ULA) with N isotropic elements forming the adaptive beamforming system's integral part as shown in the below figure 4.

The antenna array's output is given by,

$$x(t) = s(t)a(\bar{}_0) + \sum_{i=1}^{N_u} u_i(t)a(\bar{}_i) + n(t) \qquad (9)$$

a ($\bar{}_0$) and a ($\bar{}_1$) represents the steering vectors for the desired signal and interfering respective signals. Therefore the desired signal is to be construct the from the received signal amid the interfering signal and additional noise n(t) [8].

As shown in figure 4 above the outputs of the individual sensors are linearly combined after being scaled using corresponding weights such that the antenna array pattern is optimized to have maximum possible gain in the direction of the desired signal and nulls in the direction of the interferers.

From the steepest descent's method, the weight vector equation is given by

$$\boldsymbol{w}(\boldsymbol{n}+1) = \boldsymbol{w}(\boldsymbol{n}) + \frac{1}{2}\text{«}[-\nabla(\boldsymbol{E}\{\boldsymbol{e}^2(\boldsymbol{n})\})] \qquad (10)$$

Where µ, controlling is the step-size parameter, is the convergence characteristics of the LMS algorithm

$e^2(n)$ is error in mean square about the beam former output y(n) and the reference signal which is given by,

$e^2(n) = [d^*(n) – w^h x(n)]^2$ (11)

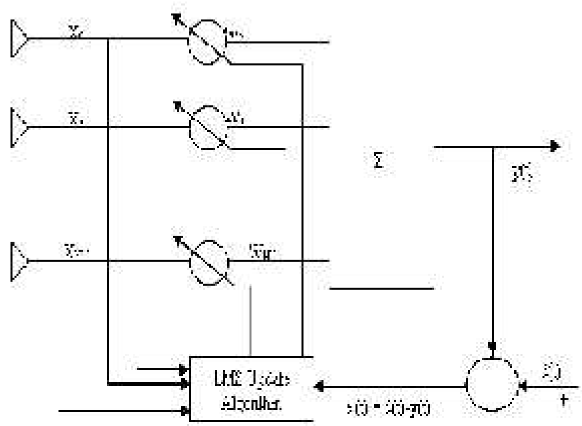

Figure 4: Adaptive beam forming network

The above weight update equation's gradient vector can be computed as

$$\nabla_w (E\{e^2(n)\}) = -2r + 2Rw(n) \qquad (12)$$

In the method of steepest descent the biggest problem is the computation involved in finding the values r and R matrices in real time. On the other hand, the LMS algorithm simplifies this by using the instantaneous values of covariance matrices r and R instead of their actual values i.e.

$R(n) = x(n)x^h(n)$
$r(n) = d^*(n)\, x(n)$

Therefore the update weights of can be given by the equation,

w(n+1) = w (n) + µx (n) e*(n)
(18)

An arbitrary value w (0) is used to initiated the LMS algorithm for the weight vector at n=0.

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The weight vector's successive corrections eventually lead to the minimum value of the mean squared error [8].

Therefore the summarization of LMS algorithm can be in following equations;

Output, $y(n) = w^h x(n)$ (13)

Error, $e(n) = d^*(n) - y(n)$ (14)

Weight, $w(n+1) = w(n) + \mu x(n) e^*(n)$ (15)

An adaptive noise cancellation method having two stages, for enhancing an ideal signal submerged in noise, where the overall method uses two adaptive filters with reference and primary signals [9].

An insight into an algorithm combining multi-channel differencing and thus obtaining reference noise and KNLMS which adaptively cancel the unknown noise is presented in [10].

7. SIMULATION RESULTS & DISCUSSION

Considered the desired signal is a sine wave of 0.015 cycles/ sample and a cosine wave of 0.008 cycles/ sample. The input is a delayed version of the desired signal corrupted by white noise of variance 0.5. The LMS, NLMS adaptive filters object of length M, step size 0.2 and offset 1e-6are shown in figures 5,6,7,8 respectively.

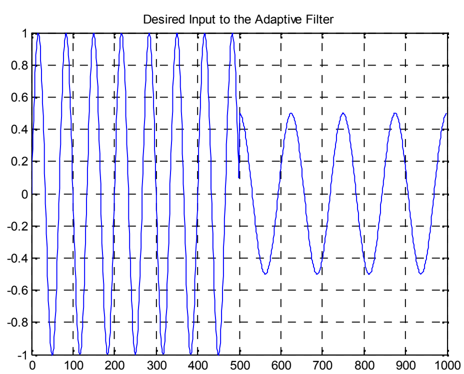

Figure 5: Desired Signal

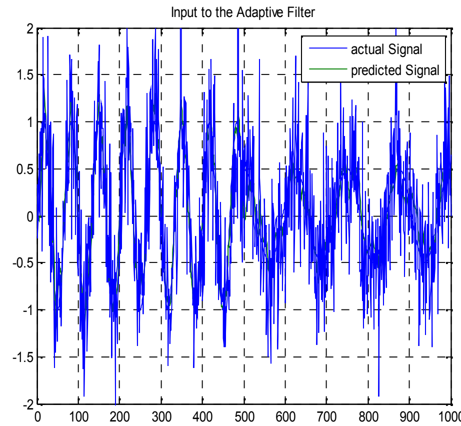

Figure 6: Signal with noise

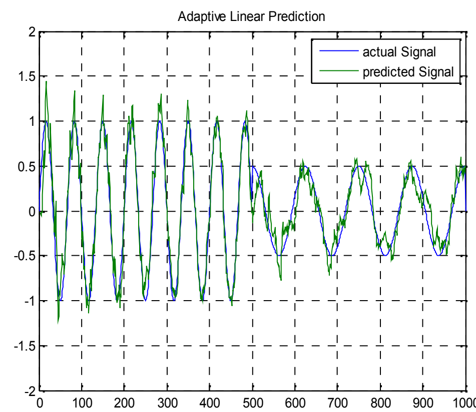

Figure 7: Noise Removed Using LMS

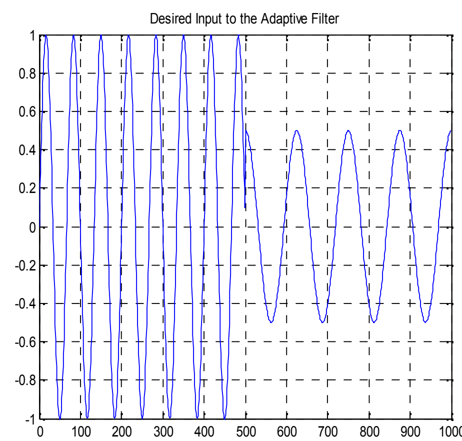

Figure 8: Desired Signal

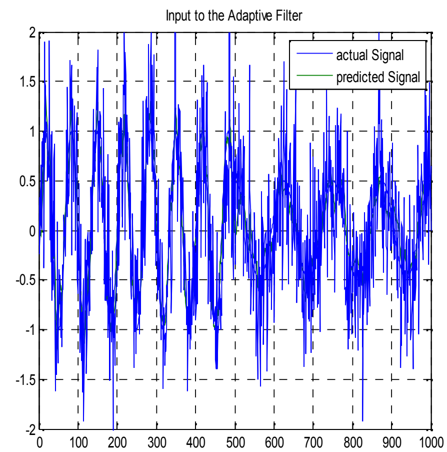

Figure 9: Signal with noise

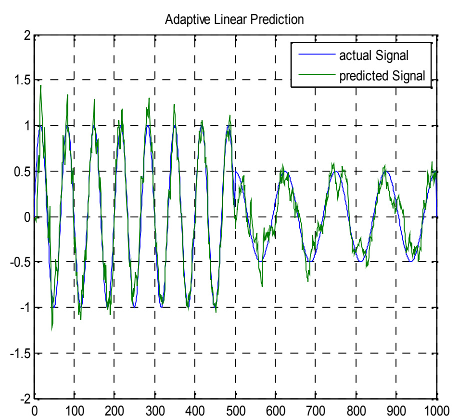

Figure 10: Noise removed using NLMS

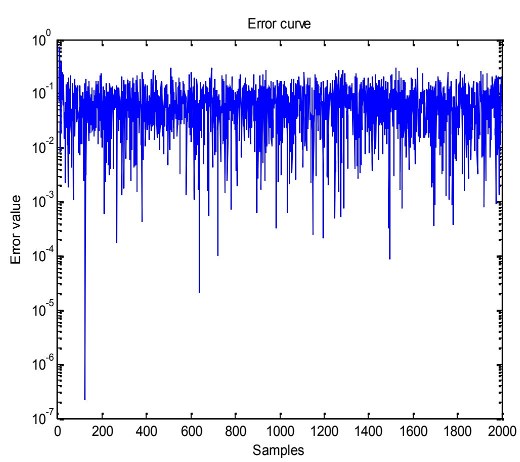

Figure 11: Estimation Error using LMS

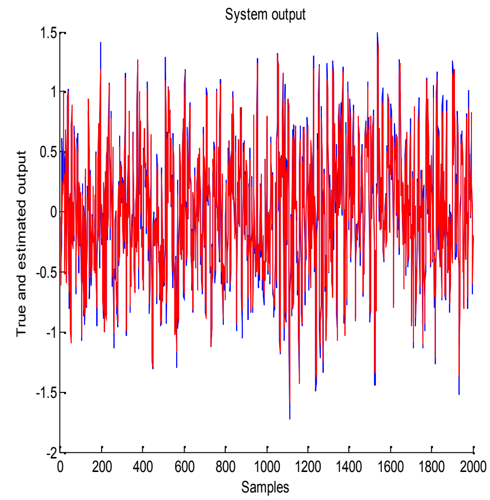

Figure 12: System Output's Signal Spectrum-LMS

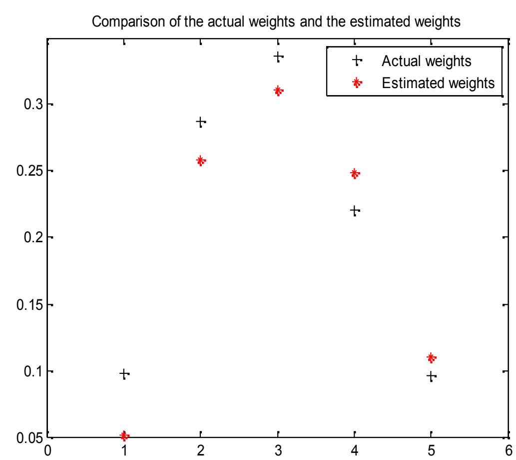

Figure 13: Comparison of LMS Weight

From the figures 12, 13, 14 we can conclude that NLMS algorithm has a better rate of convergence. The number of samples considered here is 2000. Also we see that the error is reduced to a great extent. The system output stabilizes in a very quick time as compared to the LMS algorithm. The estimated output is sometimes less and sometimes greater than the actual output.

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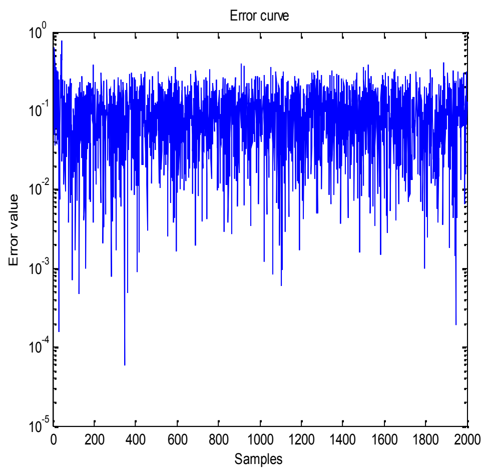

Figure 14: Estimation Error using NLMS

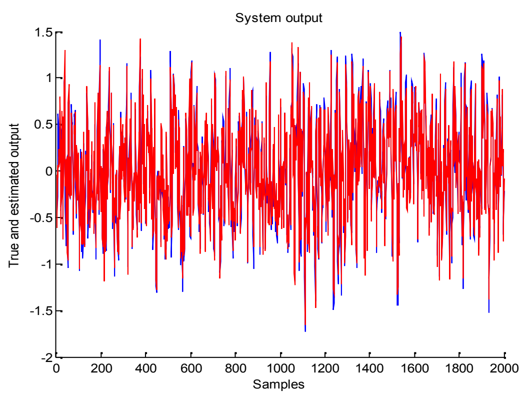

Figure 15: System Output's spectrum-NLMS

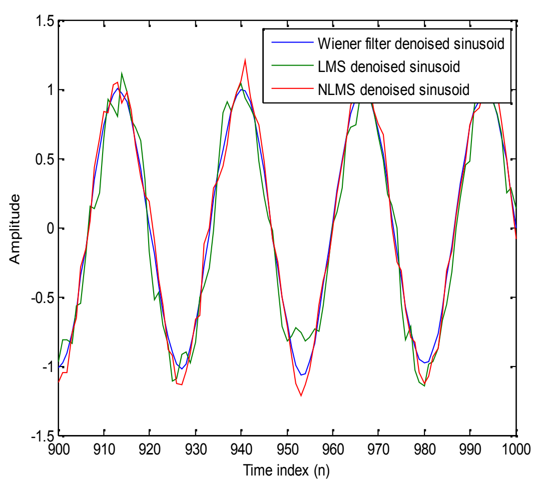

Figure 16: Comparison of Wiener-LMS& NLMS Algorithms

8. CONCLUSION

The adaptive filtering techniques are using in almost every electronic equipment. So it has become imperative to develop an algorithm for the adaptive filter.

We have explored different ways of finding a suitable adaptive filter algorithm which improves the adaptive filter performance. All the parameters regarding the adaptive filter algorithms are calculated. First the LMS algorithm discussed was most widely used, as it is computationally easy and numerically robust. But this algorithm has a disadvantage that the rate of convergence of this algorithm is very slow.

NLMS algorithm is a better choice as it has a better rate of convergence compared to the LMS algorithm. From the Figure 16 we can conclude that Wiener algorithm is a better algorithm compared to NLMS algorithm by considering the estimation weights & errors. New knowledge this work has created is that Weiner filter is the best noise reduction filter which can be used in the fields of signal processing.

This work is necessary because, one of the major applications of this work would be to save life during heart attacks and other deadly diseases, by providing most accurate signal by providing the best possible noise elimination from the acquired signals such as ECG. Future work may include a complete evaluation of our real-time system and investigation of optimal design criteria for the pre-emphasis filters, as well as alternate means of sub-band signal whitening. Wiener filter in hybrid with various algorithms can provide best solutions to sensor array processing.

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