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The Effects of Changing Step-Size Parameter on an Adaptive Noise Cancellation Using Least Mean Square Algorithm (LMS)

Ouday Nidhal Ameen Hanosh1 , Baraa Wasfi Salim2

Abstract- In this project the Adaptive filter with Least-Mean Square (LMS) algorithm are used for denoising which is known also Adaptive Noise Cancellation. Noise cancellation is a special case of Optimal filter which is used tan o estimate the noise by filtering the reference input (for example the white Gaussian noise signal) and then subtracting this noise estimate from the corrupted input signal which containing both signal and noise.

The adaptive filter used for denoising or noise cancellation has a privilege lies in achieving levels of noise rejection that are impossible to achieve by other signal processing methods or direct methods of removing noise. Finally, this project was built using MATLAB R2012b

Index Terms— Adaptive filters, LMS algorithm, Noise Cancellation, Signals, Sampling frequency, Tap-weight length , Tap-weight vector, MATLAB Program.

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1 Introduction

It is clear that there is no a specific method to reduce the level of noise because the probability distribution of the noise are different [1]. Noise cancellation is used when some information about the reference noise signal is available. This technique has many applications, such as speech processing, echo cancellation and enhancement, biomedical signal and image processing, antenna array processing and many other applications [2].The basic methods of noise cancellation use only one essential signal [3].

Adaptive filters are digital filters with an impulse response; these filters can be adjust or changed over time to meet the characteristic of the desired system. On the contrary of fixed filters, adaptive filters don't need a priori information about the characteristics of the signal to be filters.

The least mean square (LMS) algorithm is one of the search algorithms that used to provide the strategy for adjusting the filter coefficients. There are a number of adaptive structures can be used for different applications in adaptive filtering, one of them is noise cancellation.

As a conclude comparison between our proposed system and the previous works, in this paper, the proposed system provides all facilities stated in the previous works that can be considered as an aggregate system of all the previous works. Hence, this system has multi-source, multi-level of destinations (level-1 receives signals from server of multi-source and level-2 receives signals from level-1), online receiving the incoming signals from the sources, online broadcasting them to level-1 of multi-destinations and online rebroadcasting them to level-2 of multi-destinations Clients. Also, many other flexible options provided with this system to be more efficient and powerful.

2 The Proposed System

Figure (1) shows the block diagram of the Adaptive noise cancellation using LMS. In this project, the desired signal d (n) is corrupted by a correlated additive noise V (n).

V (n) chose to be a white Gaussian noise generated using the following MATLAB code:

V=wgn(k,1,0)

Where the two firs numbers k and 1 represent the size of the vector V(n) which containing a white Gaussian noise of power 0dbW The noise signal V is a column vector of size k.

The desired signal d(n) is the addition of the signal S(n) and the noise signal V(n).

d (n) = S(n) + V (n) …..(1)

In this project S(n) generated using the following MATLAB code:

Fs = 44100; y = wavrecord (15*Fs,Fs,'int16');

Where Fs represents the sampling frequency, the duration time for this file is equal to 15 second.

This recorded file is stored using the following code:

wavwrite (S,File name)

wavwrite (S,'ouday.wav') . So S(n) signal is a collection of audio recorded data for a 15 sec time duration. Furthermore the FIR low pass filter is used to remove the high frequency noise from the S(n) signal.

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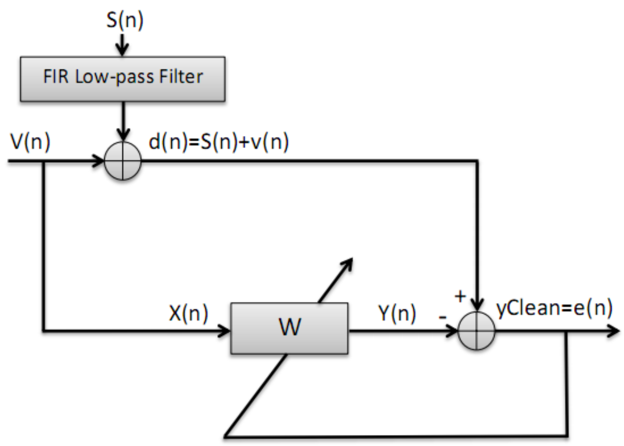


Fig. 1. Block Diagram of Adaptive LMS Noise Cancellation

The input signal to the adaptive filter X(n) is equal to the value of input noise signal where:

$X(n) = V(n)$ ….. (2)

While the adaptive filter's output Y (n) is equal to

$Y(n) = W(n)^T X(n)$ …..(3)

Where W(n) is the filter tap-weight vector at time (n).

The error signal or yClean is equal to :

$yClean = error = d(n) - Y(n)$ …..(4)

The overall output of this system is the error signal (yClean) and not the adaptive filter's output Y.

3 Least Mean Square (LMS) Algorithm

The LMS algorithm is an iterative technique developed by Widrow and Hoff in 1960 [4].

This algorithm uses a low computational complexity and because of its robustness is used to minimize the mean square error and also to update the tap-weight vector (W(n)) of the adaptive filter. Moreover the LMS algorithm can be written in matrix form:

$W(n+1) = W(n) + \mu e(n) X(n)^*$ …..(5)

Where:

$X(n) = [\, X(n) \;\; X(n-1) \;\; X(n-2) \ldots\ldots \; X(n-k+1) \,]^T$ is the input data

k: is length of the filter.

μ : is the step size

$W(n) = [\, W_0(n) \;\; W_1(n) \;\; W_2(n) \ldots\ldots\ldots W_{k-1}(n) \,]$ is the filter weight-taps at time (n).

In this project we suppose that yClean(n) = e(n) so the equation (5) can be rewritten in the following form:

$W(n+1) = W(n) + \mu yClean(n)\; X(n)^*$ …..(6)

4 Misadjustment

In the LMS algorithm, the step-size parameter μ must be suitable chosen so that its important to know what the Misadjustment term means. The Misadjustment (M) is defined as the dimensionless ratio of the steady state value of the average access mean-squared error to the minimum mean-squared error. It can be shown that:

$M = (\mu \sum \forall i \lambda i) / 2 - (\mu \sum \forall i \lambda i)$ ………(7)

For convergence

$\mu < (2/\lambda\, max)$.

If μ is small enough so that

$\mu < \sum \lambda i$.

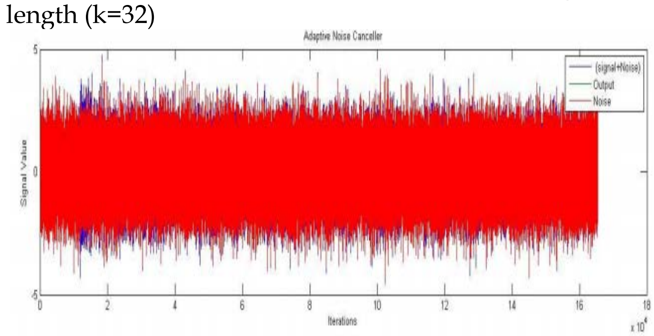
Then the Misadjustment M varies linearly with μ

5 Analysis of Simulation Results

In this project three different values of Step-Size factor μ were used in order to know the effect of this parameter on the output signal.

The first Step Size was chosen very small and equal to (μ=0.004), as a result there is a low output noise and a low signal distortion (see Fig(2) and Fig(3)).

For Step-Size Parameter μ=0.004 and tap-weight vector length (k=32)



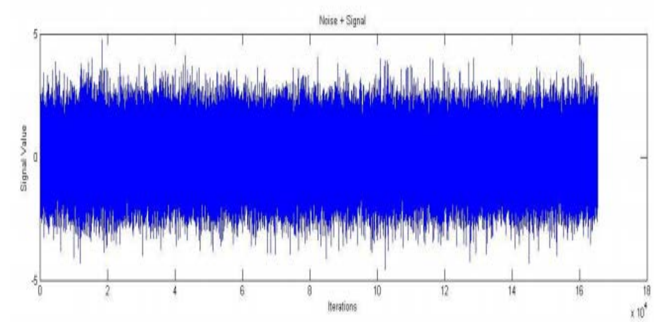


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Fig. 2.a. Shows the Combination of Desired Signal (d), Output Signal (yclean) and the Noise Signal (V).
b. Shows the Noise + Signal (Desired Signal d).

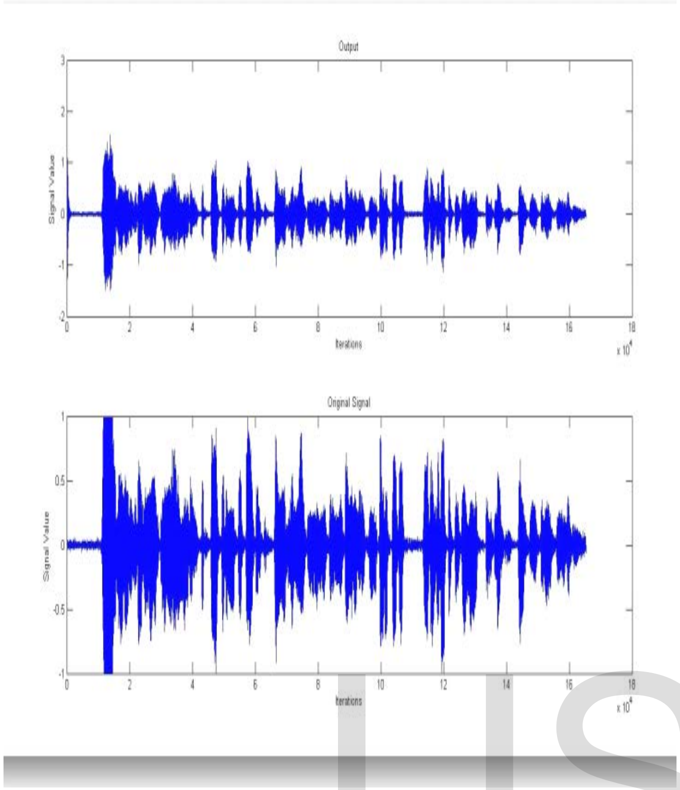

Fig .3. Output signal (yClean) vs. original signal (S)

Moreover it is easy to detect the level of noise for the output signal compared with the original signal.

The MATLAB code wavplay (filename) is used to listen to the original signal (S) and the output signal (yClean).

wavplay (yClean):Listening to the Output signal
wavplay (S): Listening to the Original signal.

For the value of μ =0.04 , it's clear that the level of the noise is higher than the level of noise For μ= 0.004 ((see Fig(4) and Fig(5)). Although the level of noise is clearly high but it is still possible to hear the recorded voice at the output signal.

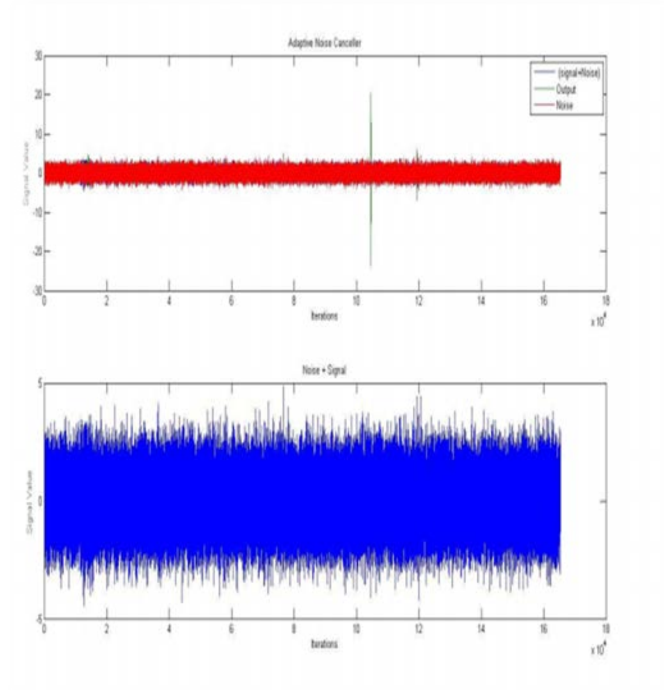

Fig .4.a. Shows the Combination of Desired Signal (d), Output signal (yClean) and the Noise Signal (V).
b. Shows the Noise + Signal (Desired Signal d)

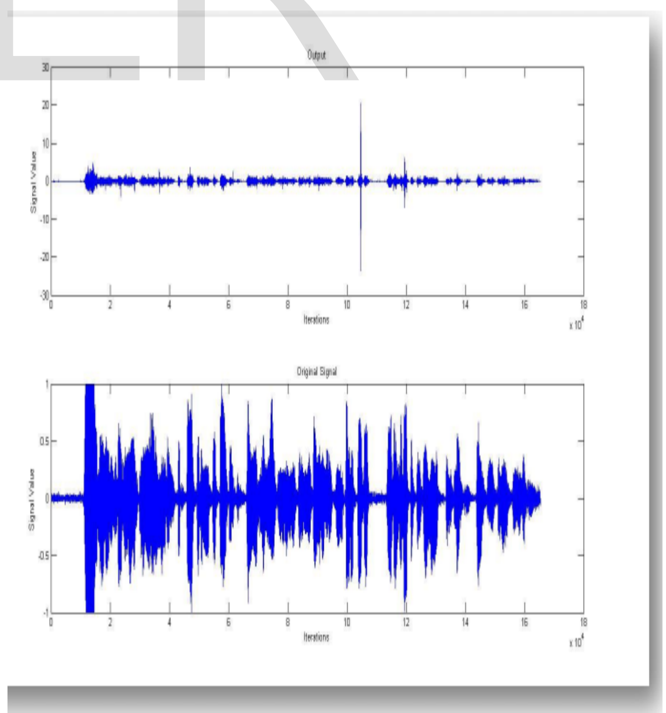

Fig (5): Output signal (yClean) vs. original signal (S)

On the other hand, for μ=0.4 it's impossible to cancel the noise because the value of that chosen for convergence not satisfied the condition $\mu < (2/\lambda max)$.

In addition to that, if the length of tap-weight vector W(n) is increased this also will increase the level of noise for the output signal (yClean) and the signal distortion. For tap-weight length k=500, the level of noise is so high and it's impossible to hear the recorded voice at the output signal (yClean) (see Fig(6)and Fig(7)). While for length equal 32, the level of noise and the signal distortion will be very low (see Fig(2) and Fig(3)).

Finally, the effect of causal approximation also considered in this project.

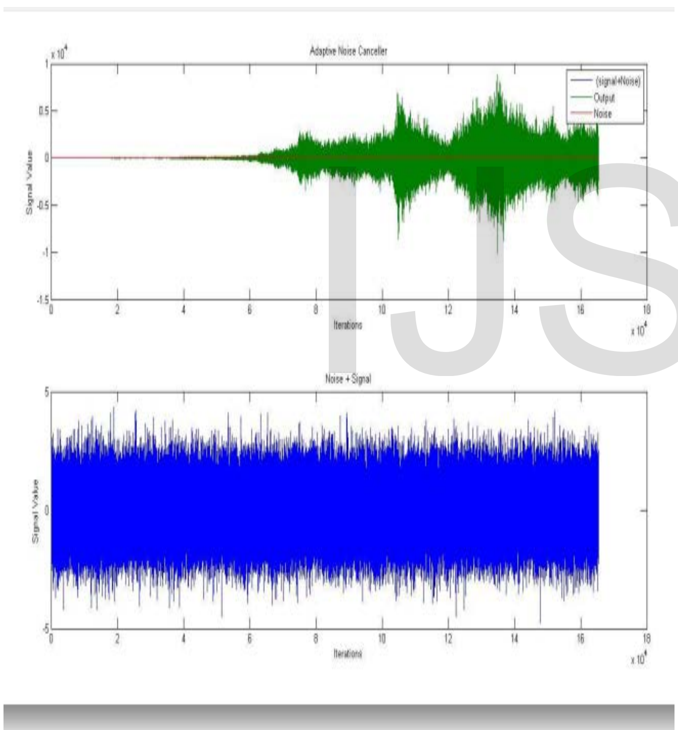

Fig. 6.a. Shows the Combination of Desired Signal (d), Output signal (yClean) and the Noise Signal (V).

b. Shows the Noise + Signal (Desired Signal d)

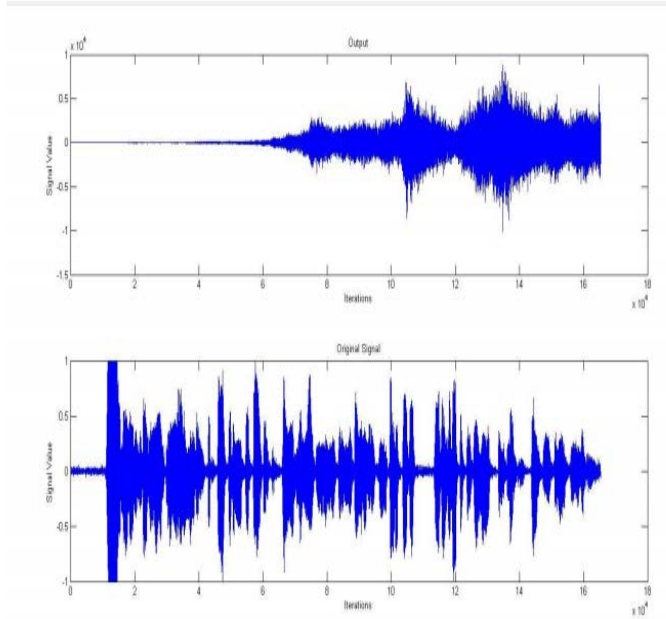

Fig .7. Output signal (yClean) vs. original signal (S)

6 CONCLUSION

In this project the Adaptive filter with least mean square (LMS) algorithm was programmed, simulated and tested for noise cancellation present in a corrupted signal. The adaptive filter is capable to process inputs with unknown properties and in some cases non-stationary. It is considered also that the output noise and the signal distortion are essentially lower than can be obtained with conventional optimal filter configurations. Furthermore, this project also studied the effect of changing the value of the step-size factor μ and also the length of the tap-weight vector W(n). As a result, the adaptive filter requires a very little computational power or memory while still maintaining the ability to handle complex signal processing.

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