Performance Analysis of Adaptive Noise Canceller using (NLMS) algorithm for Speech Communication Systems

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Abstract

Any received speech signal can be distorted by interference with an additive noise, which often can be recuperated by using special type of algorithms. In numerous wireless speech communication applications, although the speech noise characteristics are very fast , therefore, an adaptive noise canceller should be used to cancel the noise of the transmitted signal. Adaptive noise Cancellation filter is an alternative technique of estimating signals corrupted by additive noise or interference; adaptive filters such as Normalized Least Mean Squares (NLMS) algorithm have been used in a wide range of signal processing applications due to its simplicity in implementation and computation of removing the background noise and distortion.

In this paper, an extensive study has been made to investigate the effects of different parameters, such as Filter length effect, step size effect , number of samples. The performance of NLMS filter for speech communication systems will be analyzed and discussed. Simulation results are presented to verify the analysis of the NLMS filter. All the results have been obtained using computer simulations built on MATLAB platform.

Keywords — Noise cancellation, Adaptive filters, Speech enhancement, NLMS filter, Signal to noise ratio (SNR). , spectral analysis.

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1. Introduction

In practical communication systems, transmitted speech signals are usually distributed partially, or at a time almost completely, by an additive noise from the transmitter, channel, and receiver [1] . Therefore, the typical method of estimating the corrupted speech signal by additive noise is to pass the corrupted signal through a specific filter that tends to suppress the noise while leaving the speech signal relatively unchanged. The design of that specific filters is the essence of optimal filtering [2] , were a special type of filters used for signal filtering can be divide into two categories Fixed or Adaptive [3] . The design of the fixed filters requires a priori knowledge of both the signal and the noise Adaptive filters. On the other hand, the second category of filters require little or no prior knowledge of the corrupted speech signal and its undesirable noise characteristics, due to the ability to adjust their impulse response to filter out or extract the correlated noise signal in the input. The adaptive filters in general has the property of exhibiting self-modification in its frequency response with respect to time, as a result it will allow the filter to adapt the response to the input signal characteristic's, which improve their performance and increases the flexibility the filter's structure [4] . Where an any adaptive filter may be expressed by the processed signal of the filter, also the structure of the filter will define how is the output signal of the filter is computed from its input signal, the inside parameters of the structure can be iteratively changed to alter the filters input-output relationship [5] , and the specific adaptive algorithm that describes how the parameters are adjusted from one instant time to the next, is shown in figure 1.

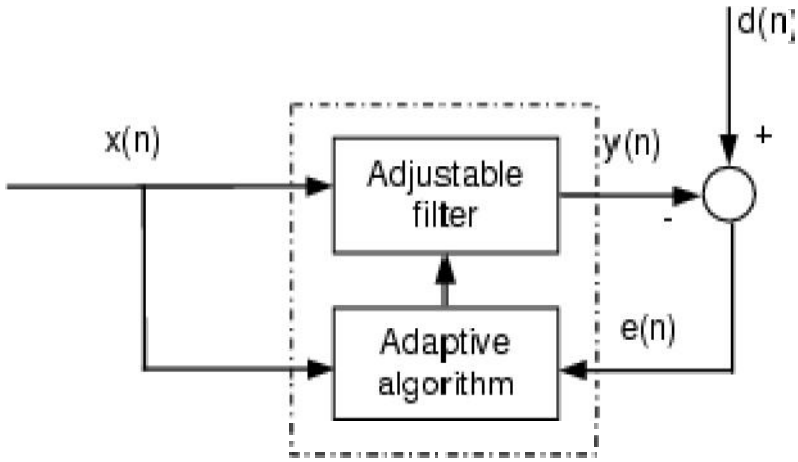

Figure (1) Block Diagram of General Adaptive Filter

Where x (n) is the received signal, d (n) is the desired signal, e (n) is the error signal, and y (n) is the output signal.

Also the adaptive filters used in the above system are for non-stationary signals, where a sample-by-sample adaptation process is required [6] . Applications of adaptive filtering include multi-channel noise reduction, radar or sonar signal processing, channel equalization for cellular mobile phones, echo cancellation and low delay speech coding. The key point of the noise cancellation adaptive filters is removing the noise from the transmitted signal adaptively to increase the signal to noise (SNR) ratio, and therefore to improve the quality of the transmitted speech signal [7] .

This paper provides a study of the performance of an adaptive noise canceller employing NLMS (Normalized Least Mean Square) algorithm. The performance of the system is analyzed while varying a range of parameters such as Filter length, effect step size, and number of samples.

Because of its simplicity, the LMS algorithm is the most popular adaptive algorithm. However, the LMS algorithm suffers from slow and data dependent convergence behavior. The NLMS algorithm, an equally simple, but more robust variant of the LMS algorithm, exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good properties the NLMS has been largely used in real-time applications.

This paper is organized in four sections; section 2 gives an idea of an Adaptive Noise Cancellation (ANC) designed model, section 3 shows the simulation and the results of the system, and finally conclusion is discussed in section 4.

2. Principles of The Adaptive Noise Canceller

Adaptive Noise Cancellation ANC is one of the most common practical applications of adaptive filters [8] . Where many situations require the using of nontraditional filtering techniques for noise cancellation. Adaptive noise cancellation technique was firstly presented by the authors Glover and Widrow, where the objective was to filter out or extract the interference component by detecting and identifying a linear model between a measurable noise source and the corresponding un-measurable interference. Therefore, the ANC is a technique of estimating additive noise or interference through an Adaptive Filter after that subtracting this measured noise from the corrupted speech signal to get the actual signal, where it has been used successfully in real world applications [9] , such as interference cancellation in electrocardiograms (ECGs) [10] , echo elimination on long-distance telephone transmission lines [11], and antenna interference cancelling [12] . In [13] , the authors propose a novel least-mean-square (LMS) algorithm for filtering speech sounds in the adaptive noise cancellation (ANC) problem. It is depends on the minimization of the squared Euclidean space of the difference weight vector under a stability constraint defined over the a posteriori estimation error. One of the main drawbacks of the LMS algorithm is to have a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In real world this is infrequently achievable.

Most importantly the normalized LMS algorithm exhibits a rate of convergence that is potentially faster than that of the standard LMS algorithm for both uncorrelated and correlated input data. The main drawback of the pure LMS algorithm is that it is sensitive to the scaling of its input $x(n)$ [14] . This makes it very hard to choose a step size μ that guarantees stability of the algorithm. The Normalized least mean squares filter (NLMS) is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input.

The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm which avoids this issue by calculating maximum step size value. Step size value is calculated by using the following formula [15] .

Step size=1/dot product (input vector, input vector)

This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x (n). This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix [16] .

$$tr[R] = \sum_{i=0}^{N-1} E[X^2(n-i)] = E\left[\sum_{i=0}^{N-1} X^2(n-i)\right] \quad (1)$$

The recursion formula for the NLMS algorithm is stated in equation.

$$W(n+1) = W(n) + \frac{1}{X^T(n)X(n)} e(n)X(n) \quad (2)$$

The NLMS algorithm has been implemented in Matlab application. Where the step size parameter is chosen based on the current input values, the NLMS algorithm shows far greater stability with unknown signals. This combined with good convergence speed and relative computational simplicity make the NLMS algorithm ideal for the real time adaptive echo cancellation system [17] . As the NLMS is an extension of the standard LMS algorithm, the NLMS algorithms practical implementation is very similar to that of the LMS algorithm. Each iteration of the NLMS algorithm requires first the calculation of the output of the adaptive filter. [17]

$$y(n) = \sum_{i=0}^{N-1} w(n) \times (n-i) = W^T(n)X(n) \quad (3)$$

Then the calculation of the error signal as the difference between the

$$e(n) = d(n) - y(n) \quad (4)$$

And the step size value for the input vector.

$$\mu(n) = \frac{1}{X^T(n)X(n)} \quad (5)$$

Finally the updated filter tap weights in preparation for the next iteration [18]

$$W(n+1) = W(n) + \mu(n)e(n)X(n) \quad (6)$$

Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm. This is an acceptable increase considering the gains in stability and echo attenuation achieved

3. Simulation of Nlms Adaptive Filter and Results

In this section the performance of NLMS algorithms in noise cancellation was evaluated as shown in Figure 2. MATLAB software was used to simulate this scenario and to design the NLMS adaptive filter. The desired corrupted speech signal by noise and the noise signal are recorded in real environment. The desired speech signal corrupted with noise is fed to the primary input of NLMS filter. While the noise signal is fed to the other input of NLMS which is called reference input. The output of the NLMS filter is supposed to remove the unwanted noise and enhance the desired signal as can be seen in the following figure.

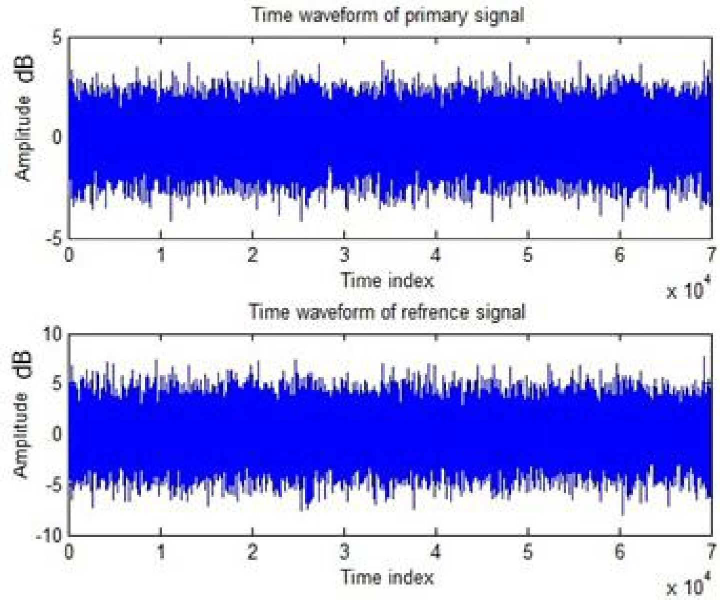

Figure (2) the primary and reference signals

Figure 2 shows the primary and the reference signals which are recorded in real environment. It can be seen from the primary signal that the desired signal is not clear and also cannot be recognized due to the present of noise.

First, the study of the effect of the changing filter length will be explained, step size and number of coefficients on estimation of desired signal and how these parameters affect the performance measures of the shown system, like convergence rate, MSE, computational complexity, and some calculation related to the frequency response, SNR improvement. Miss-adjustment will be used, and finally analyses and comparing the obtained results will be done. For visual inspection, the resulted curves were smoothed by running window of 1.000 samples.

A. Filter length effect

Figure 3 shows the behavior of the NLMS algorithm for different filter lengths (case1: length=4 taps, case 2: length=16 taps, case 3: length=32 taps) while fixing the other parameters like step size, number of iteration,, etc. (A step size of 0.01 were chosen because it's less than two so it matches the condition of choosing the step size for NLMS algorithm and also it ensure that the used algorithm is stable which is very important parameter for real-world applications).

As seen from figure 2, if the filter length will be increased, faster convergence can be obtained, i.e. less iteration is needed to reach the steady state. But on the other hand, the filter complexity will increase, because more computations is also needed like addition and multiplication.

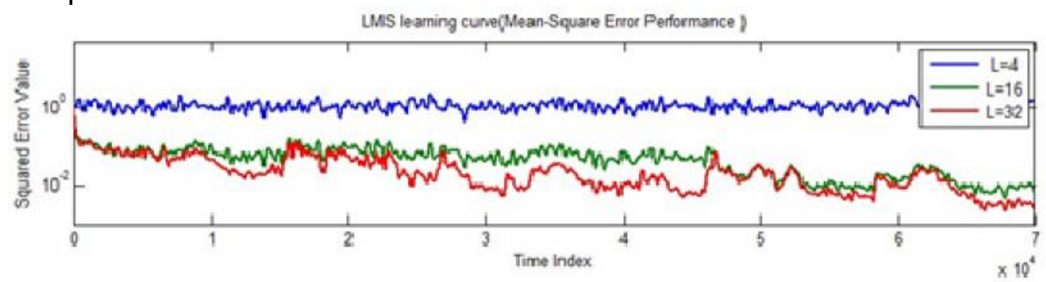

Figure (3) Behavior of NLMS algorithm for different filter length and with a step-size parameter **μ=0.05**

B. Step size effect

Figure 4 Shows the behavior of the NLMS algorithm for different step size (or it also called Convergence of speed) values (case1: step size=0.005, case 2: step size=0.05, case 3: step size=1) while the filter length is constant and equals to 32 taps.

As it can be seen in figure 4, when the step size value is small, the system will converge slowly. While choosing a large value of step size, the system will converge faster; also less number of iteration is needed to reach the steady state. For a large μ the output of the system will have some ripples (oscillations) and it maybe will result in unstable filter which should be avoid.

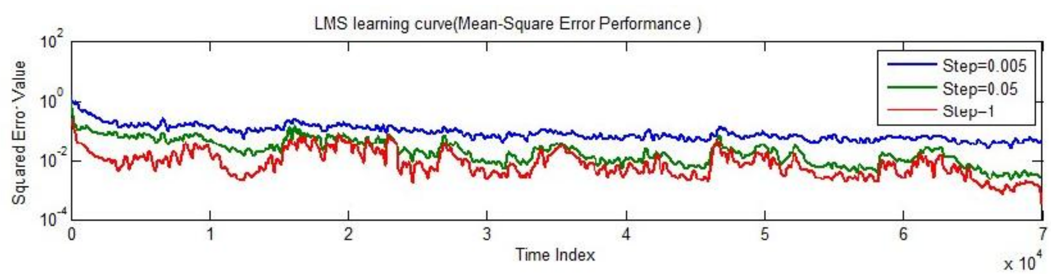

Figure (4) Evolution of the Mean Square Error for different values of the step size (μ) (L=32).

According to the last results above, it was found that if the step size 0.05 and filter length 32 taps were used, stable and reasonable results can be obtained. So these parameters will be used to setup NLMS filter and then some calculations will be done.
When setting the step size to 0.05 and filter length to 32, the NLMS filter output converging to the reference signal and the error signal converging to the desired signal are shown in the Figure 5.
While Figure 6 shows the frequency response of these signals. Figure 7 shows the weight tracks of different weights (only the first and the second weights will be tracked). Figure 8 Shows the learning curve (MSE of the output) for the NLMS filter at step sizeμ =0.05 and number of weights N=32.

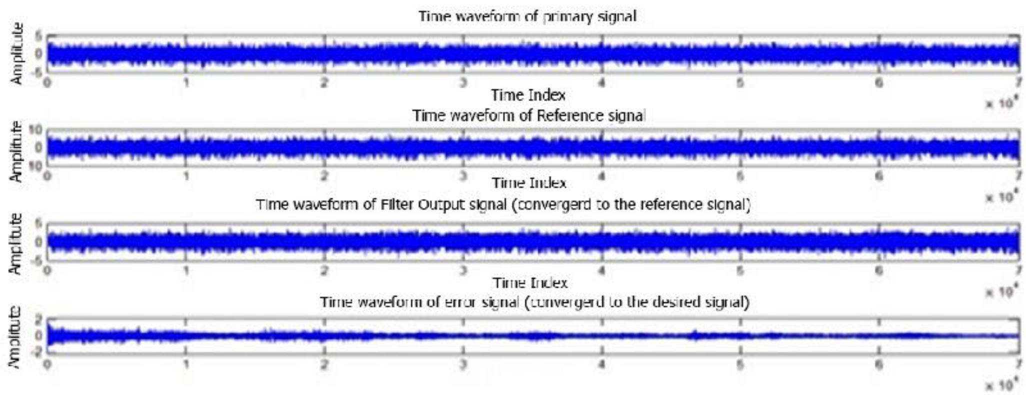

Figure (5) the primary, the reference, the filter output and the error signals (with μ=0.05 and N=32).

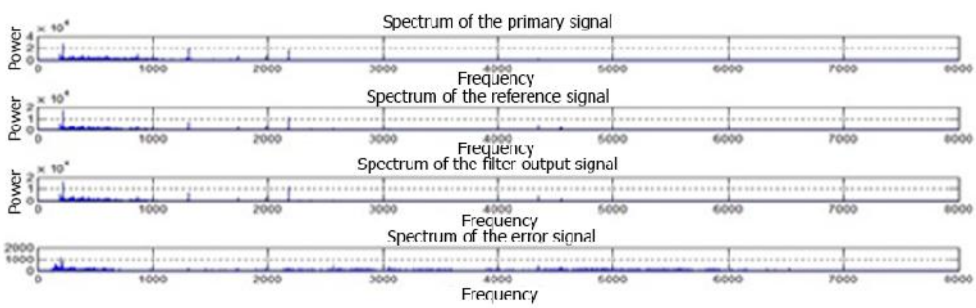

Figure (6) the frequency spectrum primary, the reference, the filter output and the error signals (μ=0.05, N=32).

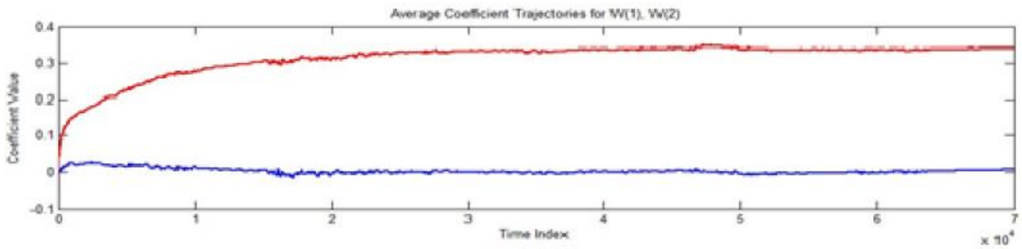

Figure (7) The Average Coefficient Trajectories for W (1), W (2) and the learning curve (with μ=0.05 and N=32).

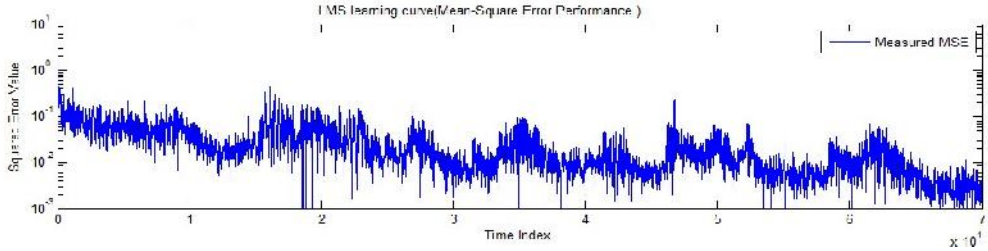

Figure (8) the learning curve (with μ =0.05 and N=32).

The choice of best algorithms is measured using performance Measure parameter like ERLE and SNR.

C. Echo Return Loss Enhancement

The ERLE is defined as the ratio of send-in power (Pd) and the power of a residual error signal immediately after the cancellation (Pe), and it is measured in dB. The ERLE measures the amount of loss introduced by the adaptive filter alone. ERLE depends on the size of the adaptive filter and the algorithm design. The higher the value of ERLE, the better the echo canceller. ERLE is a measure of the echo suppression achieved and is given by

$$ERLE = 10\log_{10}\frac{P_d}{P_e}$$

D. Signal to Noise Ratio

SNR is defined as the ratio of signal power to the noise power corrupting the signal. The Signal to Noise Ratio is the defining factor when it comes to the measurement of quality of signal. A high SNR means good quality of signal with low distortions.

$$SNR = 20\ \log_{10}\left[\frac{rms\ (speech)}{rms\ (noise)}\right]$$

Figure 9 shows SNR improvement by finding the Echo Return Loss Enhancement (ERLE) which can be used to evaluate the performance of the noise cancelling system. Clearly, the NLMS algorithm is able to remove the noise signal and enhance the desired signal.

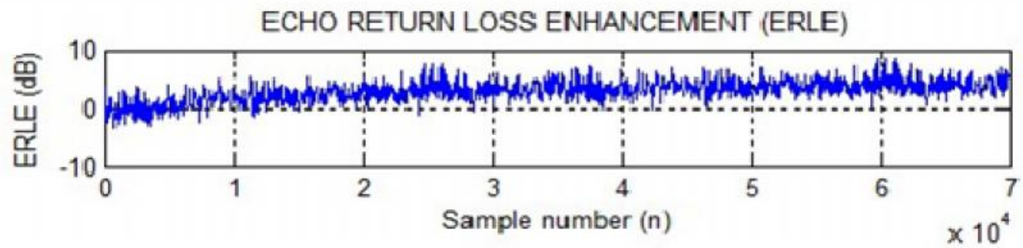

Figure (9) Echo Return Loss Enhancement (ERLE) [dB] (withμ =0.05 and N=32).

Figure 10 shows the plotting of the performance surface contours for the 32 weights filter case and also shows plotting the optimum solution obtained by Wiener filter. This figure also shows that by increasing the iterations the weights will move around the optimum values. Actually the weight vector does not just go to optimum values and stop there. In fact, it will undergo Brownian motion and hover around optimal values. Similarly, the MSE will hover around minimum MSE (Min. MSE).Thus, the asymptotic MSE (Final MSE) will actually be larger than the minimum MSE. That is, there will be an excess MSE.

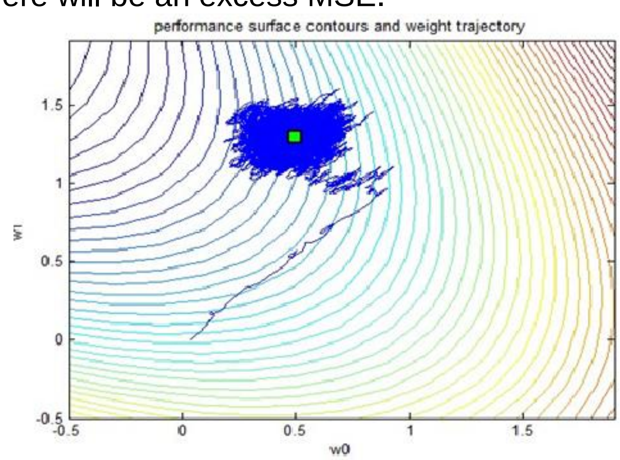

Fig (10) Performance surface contours and weight trajectory for 32 weights filter case and the Optimum solution obtained by Wiener filter.

Figure 11 shows the learning curve for our filter and shows also the Min MSE and Final MSE, so the miss adjustment M can be calculated, which is defined as the ratio of the excess MSE in the steady-state and the minimum MSE.

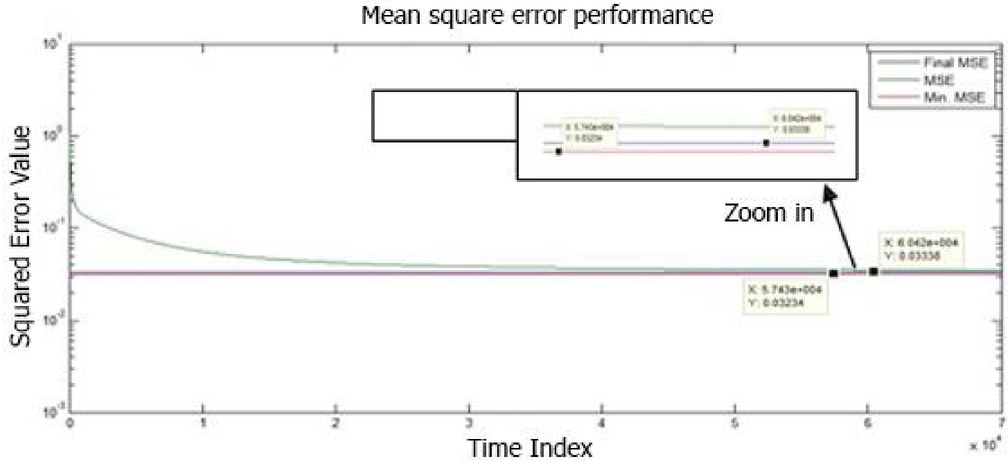

Figure (11) the learning curve

4. CONCLUSION

Adaptive filters have been shown to be one of the powerful tool for cancelling unwanted noise signals that present in a noisy speech signal.

In this paper, the performance of adaptive noise canceller using NLMS algorithm has been evaluated with varying different parameters of the system. The adaptation capability of the system to any input noise situation as well as the effects of step size, Filter length effect, number of samples and input noise level on the performance of the system are thoroughly studied considering a speech signal as useful signal. It is evident that individually each of these parameters has an optimum value at which the adaptive noise canceller showed best performance.

The SNR improvement by finding the Echo Return Loss Enhancement (ERLE) is obtained, which can be used to evaluate the performance of the noise cancelling system. Clearly, the NLMS algorithm is able to remove the noise signal and enhance the desired signal.

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تحليل الأداء لمزيل الضوضاء التكيفية باستخدام خوارزمية NLMS لأنظمة الاتصالات الصوتية

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المستخلص

ان الاشارة المرسلة من مصدر الى مستلم و التي تحتوي على معلومات صوتية قد تتشوه عند الاستقبال عن طريق التداخل مع الضوضاء المضافة خلال الارسال، والتي غالبا ما يمكن استعادتها باستخدام نوع خاص من الخوارزميات الموجودة في العديد من التطبيقات اللاسلكية التي يستخدم فيها الصوت كنوع من الاشارة المرسلة، ان هذة التطبيقات المرسلة للصوت تكون فيها خصائص إشارة الصوت او الكلام مرسلة و مستلمة بسرعة جدا، وبالتالي، ينبغي أن يستخدم طريقة لازالة الضوضاء و الغاء ضجيج الإشارة المرسلة. لذلك يعد فلتر إلغاء الضوضاء التكيفي طريقة بديلة لتقدير و ارجاع الإشارات التي تضررت بسبب الضوضاء أو التداخل الإضافي، وقد استخدمت مرشحات تكيفية مثل خوارزمية أقل متوسطات المربعات (NLMS) في مجموعة واسعة من تطبيقات معالجة الإشارات بسبب بساطتها في التنفيذ وحسابها الدقيق في إزالة الضوضاء الخلفية والتشويه. في هذه البحث، سيتم تحليل ومناقشة مرشح (NLMS) لأنظمة الاتصالات الكلامية ومناقشتها. و سيتم عرض نتائج المحاكاة للتحقق من تحليل مرشح(NLMS) .

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