Noise Cancellation Using Adaptive Filters of Speech Signal by RLS Algorithm in Matlab

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Abstract: ***In wireless communication, if the property of signal are given, fixed filter are used by us.. but when the property of signals are unknown then we require to adjust the filter, this type of filter is called adaptive filter. It is use for remove the background noise. Here, in this paper we are introducing a new method for noise cancellation through RLS algorithm in matlab. It is more efficiencally and effectively from the other method of noise cancellation. The update filter coefficient are auto considered, so this method is very fast response and find estimated error and getting the original noise. Here we are using a speech signal as a input signal that should being contained many type of noise.***

Keywords: noisy speech signal, adaptive filter, noise canceller, RLS algorithm, matlab program

1. Introduction

Noise is a unwanted electrical disturbance which gives to rise to audible or visual disturbance in the communication systems and errors in the digital communication.
the noise can rise from different types of sources..
1) Natural sources
2) Man made sources
3) Internal sources
If we know the parameter where fixed filter are used. The another filter is adaptive filter, it has the ability to adjust their impulse response to filter out the correlated signal in the input. Here, we are introducing the adaprive filter in RLS algorithm. it is a finite impulse response that is use to repeatvely. In the RLS adaptive filter are use to remove the noise from the input signal. The RLS adaptive filter uses the reffrence signal on a input port and the desired signal on the desired port to automatically match the filter response in the noise filter block. The filtered noise should be completely subtracted from the " noisey signal" of the input sine wave and noise input signal, and the "Error signal" should contain only the originally signal.

2. Related Work and Problem Identification

In many year, there are many method for noise cancelling in wireless communication system like LMS algorithm, average algorithm, RLS algorithm here, it has consist some limitation in the projects. firstly project proposed is based by using averages of both data and correction terms to find the updated value of tap weights of the ANC cotroller of the speech signal. In the 2nd paper present a simulation scheme to simulate an adaptive filter using LMS algorithm. In this project the input signal are speech signal and a refrence input containing noise. Another paper presented design of adaptive noise canceller using RLS filter in this paper we are using RLS algorithm for remove the noise from input signal. We consider as a input signal are sin wave. Another paper Presented statistical analysis of the LMS adaptive algorithm with uncorrelated Gaussian data. The outcome of this paper place fundamental limitations on the MSE performance and rate of convergence of the LMS adaptive scheme. One paper Presented statistical analysis of the LMS adaptive algorithm with uncorrelated Gaussian data. The outcome of this paper place fundamental limitations on the MSE performance and rate of convergence of the LMS adaptive scheme.

3. Methodology

3.1 Adaptive filters

Adaptive filters, on the other hand, have the ability to adjust their impulse response to filter out the correlated signal in the input. They require little or no a priori knowledge of the signal and noise characteristics. (If the signal is narrowband and noise broadband, which is usually the case, or vice versa, no a priori information is needed; otherwise they require a signal (desired response) that is correlated in some sense to the signal to be estimated.) Moreover adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions.

3.1. 1 Mean Square Error (MSE) adaptive filters

The mean squared error (MSE) of an estimator measures the average of the squares of the "errors", that is, the difference between the estimator and what is estimated. MSE is a risk function, corresponding to the expected value of the squared error loss or quadratic loss. The difference occurs because of randomness or because the estimator doesn't account for information that could produce a more accurate estimate.[

The difference between the desired signal d(n), and the actual output of the adaptive filter y(n)

$$\xi(n) = E[e^2(n)] = E[(d(n) - y(n))^2]$$

3.1.2 Recursive Least Squares (RLS) adaptive filters

They aim to minimize a cost function equal to the weighted sum of the squares of the difference between the desired and the actual output of the adaptive filter for different time instances.

$$\zeta(n) = \sum_{k=1}^{n} \lambda^{n-k} e_n^2(k)$$

Where k=1, 2, 3.n., k=1 corresponds to the time at which the RLS algorithm commences. Later we will see that in practice not all previous values are considered, rather only the previous N (corresponding to the filter order) error signals are considered.

4. Recursive Least Square

The Recursive Least Squares (RLS) algorithm is based on the well-known least squares method [6]. The least-squares method is a mathematical procedure for finding the best fitting curve to a given set of data points. This is done by minimizing the sum of the squares of the offsets of the points from the curve. The RLS algorithm recursively solves the least squares problem. In the following equations, the constants λ and δ are parameters set by the user that represent the forgetting factor and regularization parameter respectively. The forgetting factor is a positive constant less than unity, which is roughly a measure of the memory of the algorithm; and the regularization parameter's value is determined by the signal-to-noise ratio (SNR) of the signals. The vector wˆ represents the adaptive filter's weight vector and the M-by-M matrix P is referred to as the inverse correlation matrix. The vector π is used as an intermediary step to computing the gain vector k. This gain vector is multiplied by the a priori estimation error ξ (n) and added to the weight vector to update the weights. Once the weights have been updated the inverse correlation matrix is recalculated, and the training resumes with the new input values. Here k=1 is the time at which the RLS algorithm commences and λ is a small positive constant very close to, but smaller than 1 With values of λ<1 more importance is given to the most recent error estimates and thus the more recent input samples, this results in a scheme that places more emphasis on recent samples of observed data and tends to forget the past .

$$\zeta(n) = \sum_{k=1}^{n} \lambda^{n-k} e_n^2(k)$$

Unlike the LMS algorithm and its derivatives, the RLS algorithm directly considers the values of previous error estimations. RLS algorithms are known for excellent performance when working in time varying environments. these advantages come with the cost of an increased computational complexity and some stability problems .

$WH\ 0 = 0$ ……………………………….…….3.1

P (0) $=\delta-1$………………………….……………...3.2

Where δ = Small positive constant for high SNR Large positive constant for low SNR For each instance of time n =1, 2, 3 …compute:

$\pi\ n = P\ n-1\ u\ n$ 3.3

$k\ n = (n)\lambda + uH\ n\ \pi(n)$…………………………3.4

$\xi\ n = d\ n - wH\ n-1\ (n)$………….…3.5

$w\ n = w\ n-1 + k\ n\ \xi\ n$ …………..….3.6

and

$P\ n = \lambda-1 p\ n-1 - \lambda-1 k\ n\ uH\ n\ p\ n-1$ ……3.7 An adaptive filter trained with the RLS algorithm can converge up to an order of magnitude faster than the LMS filter at the expense of increased computational complexity.

We rewrite to the form of m-file as a file *rls1.m* :

```
1 function y=rls1 (u);
2 global Wo Wn Wm Po Pn Pm;
3 [m,n]=size(u);
4 uin=u(1:m-2,1);
5 yout=u(m-1,1);
6 lambda=u(m,1);
7 Wo=Wm;
8 e= yout-uin'*Wo;
9 Po=(1/lambda)*Pm;
10 Pn=Po - ((Po*uin)*(uin'*Po))/(1 + uin'*Po*uin);
11 Wn=Wo+((Pn*uin)*e);
12 y=Wn;
13 Pm=Pn;
14 Wm=Wn;
```

In the first line we define the *rls1* function, which has one input *u* and one output *y*. In the second line we declare auxiliary matrixes *P* and vectors *W*. In the third line we ascertain size of input data. In the lines 4, 5 and 6 we divide input data to appropriate vectors *uin* = **u**k+1, *yout* = *d*k+1 and *lambda*. The constant *lambda* is auxiliary constant (neglecting factor). In the line 7 we write to the auxiliary vector of weight Wo the weights from former time step. In the following line we evaluate inaccuracy of prediction, i.e., as it is termed in Eq. 2, the prior remainder, according the relation: (*d*k+1–**u**k+1 T**w**LS(*k*)). In line 9 we set down covariance matrix from previous time step into matrix *P*o. Next line represents the actualisation of covariance matrix according Eq. 1. In the line 11 we actualise the weights of adaptive filter. Actualised weights are inscribed to the output of function in the line 12. In the last two lines we take down the covariance matrix and weight vector for the use in the following time step

5. Result

The RLS algorithm was simulated using Matlab of the speech signal. This algorithm proved to be very effective. Figure 5.1 is total output ,Figure 5.2 shows the input signal. Figure 5.3 shows error signal. Figure 5.4 is desired signal, 5.5 shows filter output. Figure 5.6 shows mean square error, Figure 5.7 show attenuation.

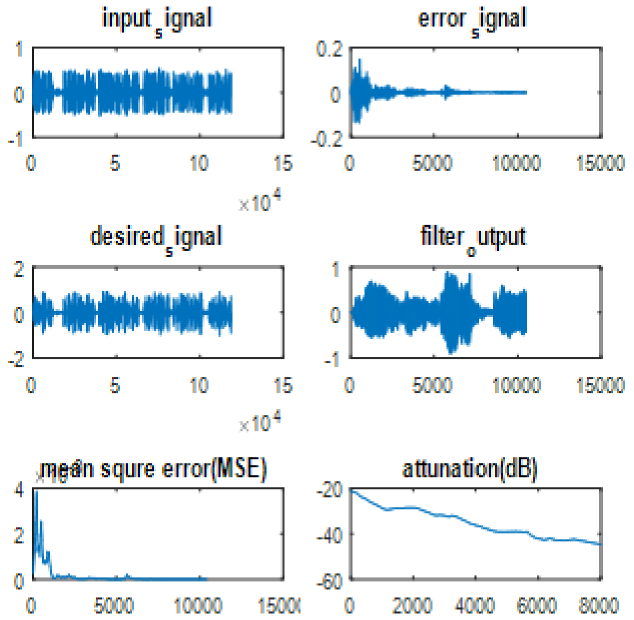

Figure 5.1

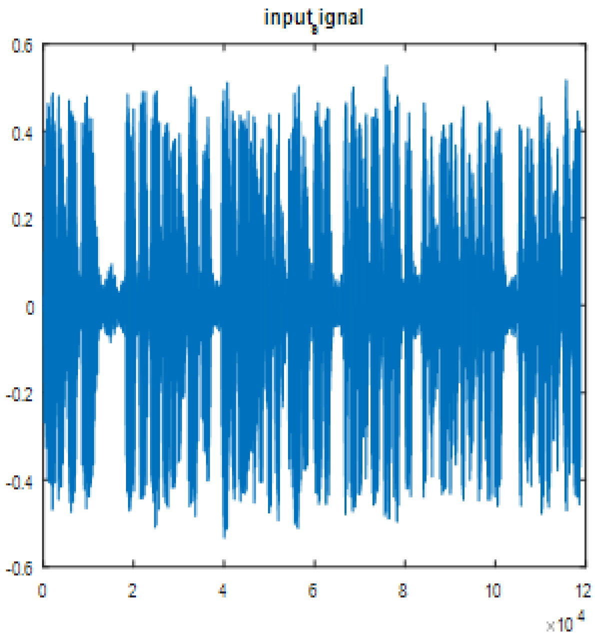

Figure 5.2

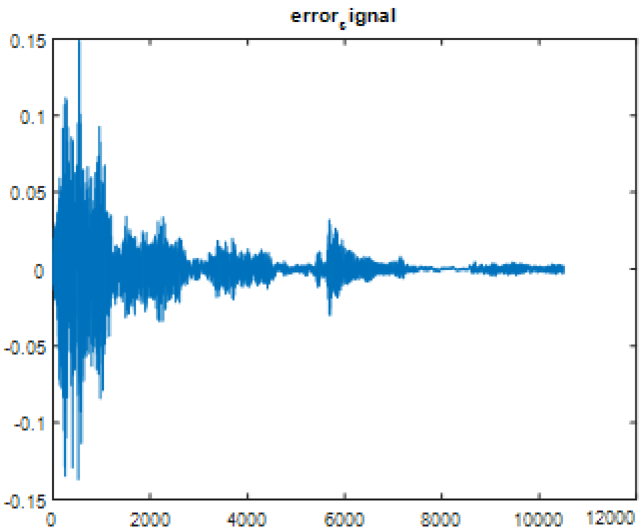

Figure 5.3

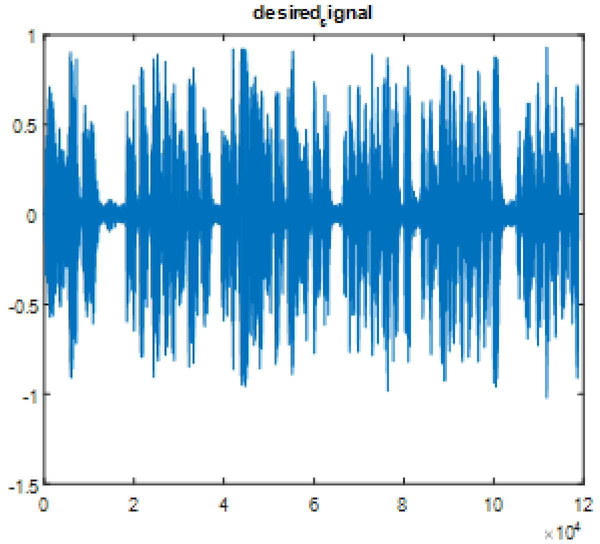

Figure5.4

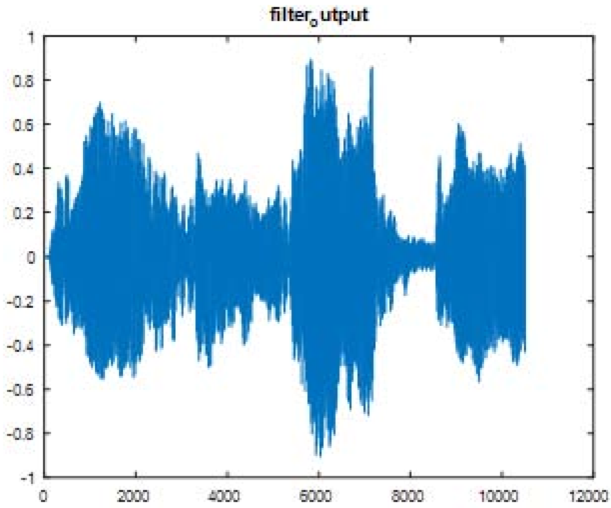

Figure 5.5

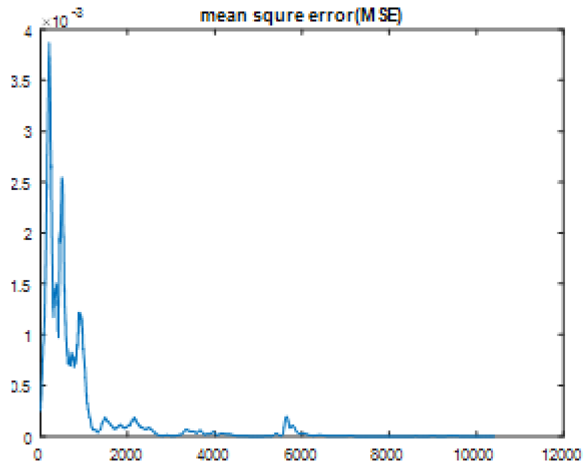

Figure 5.6

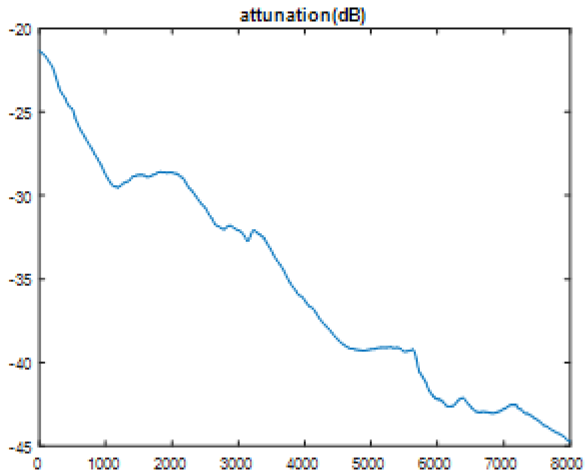

Figure 5.7

6. Conclusion

In this project we used the RLS algorithm for adaptive noise cancellation from the input signals. Here we use the speech signal as a input signal. In this algorithm the updated filter coefficient are automatic considered. It has the greatest attenuation of any algorithm, and converges much faster than the LMS algorithm. This performance comes at the cost of computational complexity and considering the large FIR order required for echo cancellation, this is not feasible for real time implementation. In future we can also perform this echo cancellation with other adaptive algorithms.

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