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Review of Noise Cancellation of Speech Signal by Using Adaptive Filtering with RLS Algorithm

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Abstract: ***In communication system, if statistical property of the signal is known then we are use a fixed filter but when the property of the signal is unknown we used adaptive filter. Adaptive filter is one of the most important areas in DSP to remove background noise. In this paper, we describe the noise cancelling using recursive least square (RLS) algorithm to remove the noise from input signal. The RLS adaptive filter uses as a reference signal on the input port and desired signal on the desired port to automatically match the filter response in noise filter block. The filtered noise should be completely subtracted from the noise signal of the input speech signal and noise input signal and the error signal should contain only the original signal.***

Keywords: noise signal, adaptive filter ,noise canceller, RLS algorithm, matlab

1. Introduction

Linear filtering is required in a variety of application. A filter will be optimal only if the designed with some knowledge about the input data. If this information of the signal is not known then the adaptive filter are used for noise cancellation. we describe the noise cancellation using the Recursive Least Squares (RLS) to remove the noise from an input signal. The RLS adaptive filter uses the reference signal on the 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 "noisy signal" of the input Speech signal & noise input signal, and the "Error Signal" should contain only the original signal.

2. Related Work and Problem Identification

In the year 2014, Vijrala Mangamma et.al worked on noise cancellation of speech signal using adaptive filtering with average algorithm. This algorithm we 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 year 2013, Gyanendra Singh et.al worked in design of adaptive noise canceller using LMS algorithm. Here 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.

In the year 2012, Reena Rani et.al 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.

In the year 2010, Krishna E. H. 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.

In the year 2009, HOMANA et.al presented on echo canceller based on a system identification scheme with adaptive algorithm. The algorithm consider FIR filter with the taps chosen to minimize an error signal desired from the system according to a stochastic gradient based method.

3. Methodology

3.1 Adaptive Filters

If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications. The adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics. Because of their self adjusting performance and in- built flexibility, adaptive filters are used in many diverse applications such as echo cancellation, radar signal processing, navigation systems, and equalization of communication channels and in biomedical signal enhancement.

3.1.1 Mean Square Error (MSE) adaptive filters

MSE is an estimator measures the average of the square of 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.

3.2 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) =δ−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:

π n =P n−1 u n 3.3

k n =(n)λ+uH n π(n)………………………….3.4

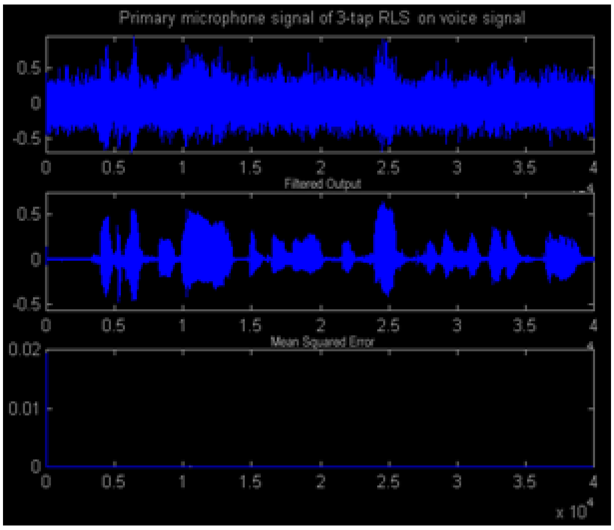
ξ n =d n −wH n−1 (n)……………..…3.5

w n =w n−1 +k n ξ n …………...….3.6 And

P n =λ−1p n−1 −λ−1k 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.

4. Conclusion

In the proposed work we are using speech signal as a input signal and RLS function are used for estimation process. In the work all the process are completed in matlab program. By Using RLS function the adaptive noise cancellation process are very faster as compare to other algorithm. It is very fast response and finding accurate original signal. It is reduces the computational complexity. So basically it is very excellent performance of finding the error by using adaptive filter.



5. Acknowledgements

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 . so this method are very fast response and find the estimated error. The filtered noise should be completely subtracted from thr noise signal of the input speech signal and noise input signal and the error signal should contain only the original sinal.

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