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Analysis of Noise Signal Cancellation using LMS, NLMS and RLS Algorithms of Adaptive Filter

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Abstract—*A microphone often picks up acoustical noise disturbances together with a speaker's voice (which is the signal of interest). There has been a tremendous amount of research over the years to control and cancel this noise level. The cancellation of noise can be effectively accomplished by using adaptive algorithms. An adaptive filter is one that self adjusts the coefficients of transfer function according to an algorithm driven by an error signal. The adaptive filter uses feedback in the form of an error signal to define its transfer function to match changing parameters. In this paper, we present an implementation of Least mean square(LMS), Normalized least mean square(NLMS), Recursive least square(RLS) algorithms on MATLAB platform with the intention to compare their performance in noise cancellation in terms of minimum mean square error(MMSE), the algorithm execution time, the required filter order, computational complexity and stability. The obtained results show that RLS has best performance but at the cost of computational complexity.*

Keywords*: Adaptive Filters, Least Mean Square(LMS), Normalized Mean Square(NLMS), Recursive Least Square(.RLS), Minimum mean square error(MMSE).*

1. INTRODUCTION

In communication system, generally different transformational operations are performed on a signal during information transmission. In signal processing, a signal containing useful information is passed through a system (e.g. filter, modulator, adder etc.) to process the signal. In noise cancelling, signal processing is concerned with filtering out the noise from the noise corrupted signal to recover the signal of interest. The statistics of the noise corrupting a signal is unknown in many situations and changes with time. Moreover, the power of noise may be greater than the power of the desired signal being transmitted. In these circumstances, adaptive filters may show satisfactory performances.

Adaptive filters are variable filters whose filter coefficients are adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter the filter to adapt the changes in the input signal characteristics. Adaptive filters have found use in many diverse applications such as telephone echo cancelling, radar signal processing, noise cancelling and biomedical signal enhancement. These filters incorporate different algorithms that allow the filter coefficients to adapt the signal statistics.

This paper, provides study of the performance of an adaptive noise canceller employing Least mean square(LMS), Normalized least mean square(NLMS), Recursive least square(RLS) algorithms on MATLAB platform with the intention to compare their performance in noise cancellation in terms of minimum mean square error(MMSE), the algorithm execution time, the required filter order, computational complexity and stability.

2. ADAPTIVE NOISE CANCELLER

Adaptive noise canceller is utilized to eliminate background noise from useful signals where a signal of interest becomes submerged in noise. One basic element of an adaptive noise cancelling system is adaptive filter. A digital filter having self-adjusting characteristics is known as adaptive filter. An adaptive filter gets adjusted automatically to the changes occurred in its inputs .The coefficients of the adaptive filter are not fixed, rather these can be changed to optimize some measure of the filter performance.

Fig. 1 shows a model of adaptive noise cancelling system. As seen in figure, an adaptive noise canceller consists of two inputs (known as primary input and reference input) and an adaptive filter. The noise-corrupted signal $(y = s + n)$ is applied as primary input. Reference input is the noise, $\bar{x}$, which is correlated with the main input, x in some way but uncorrelated with the signal, s. The noise reference input is applied to the adaptive filter and an output, $\hat{n}$ is estimated which is a close replica of n as much as between n and $\hat{n}$ is minimized during this process. Finally, the recovered signal is obtained by subtracting the estimated noise, $\hat{n}$ from the primary input.

The desired output of the adaptive noise canceller is given by

$$\hat{s} = e = y - \hat{n}$$
$$= s + n - \hat{n} \quad (1)$$

Where, s, n, y and e are termed as useful signal, band limited noise, the noise corrupted signal and the error signal.

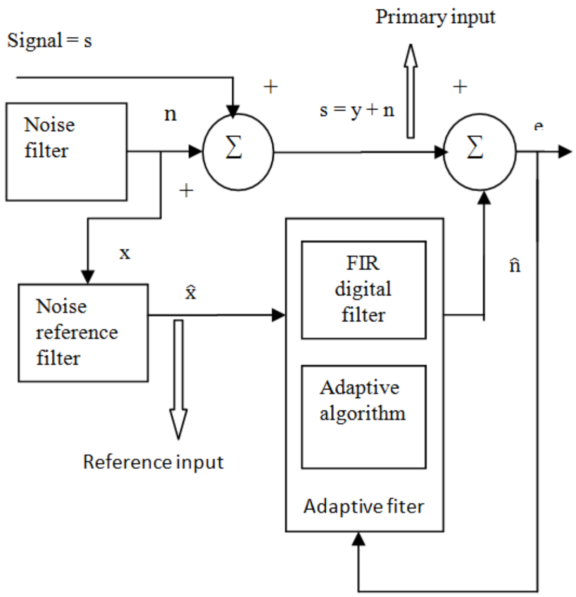

Fig. 1: Model of Adaptive Noise Filter

3. ADAPTIVE ALGORITHMS

Many adaptive algorithms have been proposed for implementing adaptive filter theory.

In this work, we have applied Least mean square(LMS), Normalized least mean square(NLMS), Recursive least square(RLS) algorithms for studying adaptive noise cancelling system performance. In the following subsections, LMS, NLMS, RLS algorithms are illustrated.

3.1 Least Mean Square(LMS) Algorithm

This algorithm modifies the filter coefficients in such a way that error gets minimized in the mean-square sense. It is a sequential mechanism which can be employed to adjust the filter tap weights by continuously observing its input and desired output.

If the filter input vector is x(n) and the desired output vector is d(n), then the filter output, y(n) and estimated error signal, e(n) can be written as equation (2) and (3) respectively .

$$y(n) = w'(n)x(n) \quad (2)$$

$$e(n) = d(n) - y(n) \quad (3)$$

where, w(n) is the filter tap weight vector.

For LMS algorithm, at each iteration, the weight vector is updated by a small amount according to the equation:

$$w(n+1) = w(n) + 2\mu e(n)x(n) \quad (4)$$

where, μ is called the algorithm step size. μ is a convergence factor, whose value decides by which

amount the tap weight vector will be changed at each iteration.

3.2 Normalised Mean Square(NLMS) Algorithm

The NLMS (Normalized Least Mean Square) scheme can be seen as a special implementation of the Least Mean Square (LMS) algorithm which considers the variations in the signal level at the input of the filter and chooses a normalized step-size parameter that yields in a stable adaptation algorithm having fast convergence rate. In NLMS algorithm the step size parameter μ is normalized to μ_n the tap weights are updated according to the following equation :

$$w(n+1) = w(n) + \frac{\mu e(n)x(n)}{x'(n)x(n)+\delta} \quad (5)$$

Here,

$$\mu_n = \frac{\mu e(n)x(n)}{x'(n)x(n)+\delta}$$

where, δ is a small constant, used to avoid the numerical instability of algorithm that may arise, and x(n), e(n), w(n) and μ represent the filter input vector, the estimated error signal, the filter tap weight vector and the step size respectively.

Normalized LMS algorithm can be viewed as an LMS algorithm with a time-varying step size parameter. In addition, normalized LMS algorithm leads to faster rate of convergence as compared with that of the standard LMS algorithm both for correlated and uncorrelated input data.

3.3 Recursive Least Square (RLS) Algorithm

The Recursive least squares (RLS) adaptive filter is an algorithm which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. This is in contrast to other algorithms such as the least mean squares (LMS) that aim to reduce the mean square error. In the derivation of the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are considered stochastic. Compared to most of its competitors, the RLS exhibits extremely fast convergence. However, this benefit comes at the cost of hh computational complexity.

The error implicitly depends on the filter coefficients through the estimate y (n)

$$e(n) = d(n) - y(n)$$

The weighted least squares error function C- function has to minimize e (n) being a function of is therefore also dependent on the filter coefficients.

$$C(W_k) = \sum_{n=0}^{k} \lambda^{n-1} e^2(n)$$

Where $0 < \lambda > 1$ is the "forgetting factor" which gives exponentially less weight to older error samples.

Finally the RLS algorithm for an p^th order filter can be

$$\alpha(n) = y(n) - x'(n)w(n-1) \tag{6}$$

$$g(n) = p(n-1)x^*(n)\{\lambda + x'(n)p(n-1)x^*(n)\}^{-1} \tag{7}$$

$$p(n) = \lambda^{-1}p(n-1) - g(n)x'(n)\lambda^{-1}p(n-1) \tag{8}$$

$$w(n) = w(n-1) + \alpha(n)g(n) \tag{9}$$

Where,

p=filter order

λ = forgetting factor

δ = value to initialize p_0

W(n) = 0

$p_0 = \delta^{-1}I$, where I is the identity matrix of rank p+1

4. EXPERIMENTAL SETUP

For Matlab simulation of noise canceller system, appropriate speech database is required. One clean sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" from Hindi Speech Database has been taken as test sample for real time speech signal as shown in figure2. The noisy version of this sentence was prepared by adding car noise from NOISEX-92 database to this clean sentence. This is shown in figure2. Experiments are performed with LMS, NLMS and RLS algorithms.

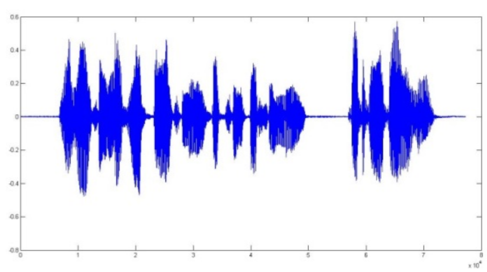

Fig. 2: Real Time Speech Signal

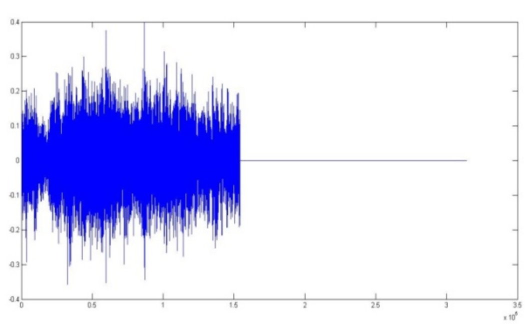

Fig. 3 : Noisy Version Of Real Time Speech Signal

5. SIMULATION AND RESULTS

5.1 Least Mean Square (LMS) Algorithm Results are obtained using a constant step size of 0.007 and a filter length of 50, 100, 200, and 1000 respectively.

Table 1: Results for LMS algorithm

Filter Length	Mean Square Error	Convrgence Time(seconds)	Complexity
50	0.0133	40.93	2N+1
100	0.0107	40.12	2N+1
200	0.0098	39.16	2N+1
1000	0.0089	38.81	2N+1

5.2 Normalised Mean Square (NLMS) Algorithm

Results are obtained using a constant normalised step size of 0.5 and a filter length of 50, 100, 200, and 1000 respectively.

Table 2: Results for NLMS algorithm

Filter Length	Mean Square Error	Convergence Time(seconds)	Complexity
50	0.010	23.74	3N+1
100	0.085	23.55	3N+1
200	0.0064	23.0	3N+1
1000	0.0055	22.78	3N+1

5.3 Recursive Least Square (RLS) Algorithm

Here results are obtained using a constant value of forgetting factor λ of 0.9 and a constant value of δ as 0.3. Filter order of 50, 100, 200, 1000 are taken

Table 3: Results for RLS agorithm

Filter Length	Mean Square Error	Convergence Time(seconds)	Complexity
50	0.0082	17.90	$4N^2$
100	0.008	16.2	$4N^2$
200	0.006	14.4	$4N^2$
1000	0.0048	13.9	$4N^2$

6. CONCLUSION

Simulation results revealed that LMS is least stable, NLMS is stable, where as RLS is highly stable algorithm. Also, convergence time taken by LMS is highest and that of RLS is least.

Overall, RLS shows best performance in reducing the noise level and minimizing the mean square error among the three but at the cost of highest computational complexity.

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REFERENCES

[1] V.K.Gupta, M.Chandra, S.N.Sharan,“Hand Free Telecommunication using Variable Step Size Algorithm”,*Radio Engineering*, VOL.22, NO.1, APRIL2013.

[2] Jyoti Dhiman, Shadab Ahmad, Kuldeep Gulia, “Comparison between Adaptive Filter Algorithms (LMS, NLMS and RLS)”,*International Journal of Science Engineering and Technology (IJSETR)*Volume2,Issue5, May2013.

[3] Hansler, E.,1Schmidt, G.,“Acoustic Echo and Noise Control:A Practical Approach. Hoboken (NJ, USA)” *Wiley*, 2004.

[4] Haykin, S.“Adaptive Filter Theory,4th. Upper Saddle River(NJ, USA)”*Prentice Hall*, 2002.

[5] Sayed, A. H.,“Fundamentals of Adaptive Filtering. Hoboken(NJ, USA)” *Wiley*, 2003.

[6] Greenberg, J. E., “Modified LMS algorithms for speech processing with an adaptive noise canceller”*IEEE Transactions on Speech and Audio Processing*, 1998, vol. 6, no. 4, p. 338 - 358.

[7] Raj Kumar Thenua and S.K. Agarwal“Simulation and Performance Analysis of Adaptive Filter in Noise Cancellation”*International Journal of Engineering Science and Technology* Vol. 2(9), 2010, 4373-4378.

[8] Paulo S.R. Diniz“Adaptive Filtering: Algorithms and Practical Implementation”,*Kluwer Academic Publishers*, 1997, ISBN 0-7923-9912-9.

[9] J.Gorriz and J.Ramrez, “A Novel LMS Algorithm Applied to Adaptive Noise Cancellation”,*IEEE Signal Process Letters*, vol. 16, no. 1, Jan. 2009.