Performance Analysis of LMS, NLMS Adaptive Algorithms for Speech Enhancement in Noisy Environment

Ch. D. Umasankar, M Satya Sairam

Abstract**: *The speech enhancement is an important technique to remove noise from corrupted speech signal. Here several Adaptive Algorithms were proposed to improve quality of speech signal. In this paper to estimate the speech enhancement performance with variety of noise reduction algorithms using adaptive filters like LMS, NLMS. Simulations were performed on noisy data which was prepared by adding machinegun, Factory, vehicle and Traffic noise at 0dB, 5dB and 10dB SNR levels to clean speech samples The performance comparison of adaptive noise cancellation (ANC) system using LMS and NLMS algorithms was carried by means of signal to noise ratio (SNR), mean square error (MSE) and root mean square error (RMSE). Based on performance analysis, the NLMS algorithm was found to be a better optimal adaptive noise canceller for speech signal***

Keywords : Adaptive Filtering, Adaptive Noise Cancellation (ANC), LMS (Least Mean Squares), NLMS (Normalized LMS), SNR, MSE, RMSE.

I. INTRODUCTION

Speech communication is the transfer of information from one person to another person via speech. Speech signals are the best way for humans' communication. Speech signals are used very often in many practical applications like speech recognition, speaker identification, general classification system and speech pre processing for hearing impaired persons' gadgets. Speech signals are corrupted in real time environment by different forms of noise such as outside speakers, background noise and also they are subject to distortion caused by communication channels such as room reverberation, low quality microphones etc [1]. In all the cases, extraction of high resolution signals is the primary task. In this manner filtering techniques come into picture.

Speech enhancement is a technique that improves quality of speech signal. De-noising techniques in Speech processing are used to enhance the corrupted signal by reducing noise. The objective of speech enhancement is to improve speech quality from degraded speech signal.

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In numerous speech communication frame works, the background interference affects the quality of speech to degrade. Therefore the signal has to be cleaned up using noise cancellation techniques before it is stored, analyzed, transmitted or processed [2].Several noise reduction techniques aim to mitigate the noise in a noisy corrupted speech signal without disturbing original (clean) signal.

II. ADAPTIVE FILTERING TECHNIQUES

An adaptive filter is a digital filter with coefficients that are determined and updated by an adaptive algorithm. The fundamental working principle of adaptive filter involves two iterations[3]: **filtering** that produces an output signal with respect to a given input signal and **adaptation** process that concentrates to manage the parameters of filters to the current environment. The whole performance & design flexibility of the adaptive filters have been widely used in various applications. Some of important applications are: Noise cancellation, signal prediction, echo cancellation, radar and biometrics signal processing. [4]

Adaptive filters are classified into two types namely linear and non-linear. In linear adaptive filters, the computation is done by estimating the desired response with the combination of the available set of observables supplied to the i/p of the filter. If not, the adaptive filter is said to be non-linear. Adaptive noise cancellation (ANC) is one of the most common applications of adaptive filters.

2.1 Adaptive Noise Cancellation (ANC):

The basic principle of an adaptive noise cancellation algorithm is apply the corrupted signal to a filter which helps to suppress the noise while leaving the signal unchanged.
ANC is an efficient procedure of receiving a signal which is affected by additive noise & is the primal core area of the digital signal processing(DSP) [5].
Below figure shows that Adaptive Noise Cancellation
d(n)=s(n)+n(n)

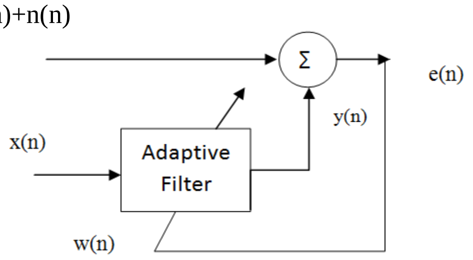


Figure.1: Adaptive Noise Canceller Block Diagram

A signal s(n) is being Txed through a channel to sensor that receives the signal which is combined with a noise n(n) and is mismatched with the signal. The primary input d(n) to the canceller is mixture of desired message signal in addition with noise i.e.

d(n)=s(n)+n(n).

The referred input ***x(n)*** are mean to consider a measuring for a distorted signal which have to match with the noise input ***n(n).***

The filter output, y(n) is given by equation (1).

$$y(n) = w^T(n) * x(n) \quad (1)$$

The output of adaptive filter subtracts from primary i/p to generate the adaptive filter error which is presented in equation (2)

$$e(n) = d(n) - y(n) \quad (2)$$

A typical performance criterion is to minimize error signal *e(n)* i.e. the mismatch between the filter output signal and a given desired signal. Based on metrics of the adaptive filter, to widely used algorithms applied to the noisy signals for speech enhancement are explained below.

2.2 Analysis of Adaptive Algorithms

The Adaptive noise canceller appeared in fig 1 comprises of Adaptive noise channel parameters d(n) is the essential sign, e(n) is the ideal sign or error-signal, y(n) is the yield of a Adaptive FIR channel, µ is the progression(step) size parameter, w(n) is the filter weight vector, x(n) is the suggested i/p signal vector & N is the filter-length utilized as parameters of channel. Here s(n) is the info vector of time deferred by input esteems,

$$s(n) \equiv [s(n), s(n-1), \ldots, s(n-N+1)]^T$$

Where the ***T*** represents the transpose of the given matrix.

$$w(n) = [w_0(n), w_1(n), \ldots, w_{N-1}(n)]^T$$

The vector represents the coefficients of the adaptive FIR filter tap weight vector at time ***n***. arrangement of an appropriate value for step size parameter(µ) is basic to the presentation of the adaptive algorithms, i.e., if µ is too little the time taken by adaptive filter to meet on the ideal arrangement will be excessively long and if µ is too huge than the adaptive filter gets temperamental and its output deviates. The adaptive filter algorithm require three particular computational strides in every emphasis as follows:

→ The output of the first stage adaptive FIR filter y(n) is estimated using equation

$$y(n) = \sum_{i=0}^{n-1} w(n)\, s(n-i) = w^T(n) s(n) \quad (3)$$

→ The value of the error function i.e., the difference between primary signal, derived adaptive filtered o/p signal is estimated using equation 4.

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

→ The tap weight vectors or filter coefficients of the first stage adaptive FIR were updated in a series of samples , for the next loop by using equation 5 – 9 for various adaptive algorithms.

i) ***LMS Algorithm***

$$w(n+1) = w(n) + \mu\, x(n) e(n) \quad (5)$$

To assure the steadiness of the algorithm, the step size is to be selected in the slab of,

$$0 < \mu < \frac{2}{NP_u} \quad (6)$$

The parameter- $\boldsymbol{P_u}$ **, *represents*** average power of the i/p signal ***s(n) &*** it is estimated as

$$P_u = x^T(n) x(n) \quad (7)$$

ii) ***NLMS Algorithm***

$$w(n+1) = w(n) + \mu(n)\, x(n) e(n) \quad (8)$$

The time varying step size parameter s***(n)*** is calculated :

$$\mu(n) = \frac{u}{\left(x^T(n)x(n)\right)} \quad (9)$$

2.3 Performance parameters of ANC

The clean speech signal can be removed from Primary(essential) signal. The presentation of simulation of ANC is thought about numerically by parameters MSE, RMSE, and SNR and abridged in Table 1.

Table 1: Performance parameters of ANC system

PARAMETERS	FORMULA
MSE	$\frac{1}{N}\sum_{k=0}^{K-1}[s(k)-e(k)]^2$
$RMSE$	$\sqrt{\frac{1}{N}\sum_{k=0}^{K-1}[s(k)-e(k)]^2}$
$SNR(After\ Filtering)$	$10\log_{10}\left(\frac{\sum_{k=0}^{K-1}[s(k)]^2}{\sum_{k=0}^{K-1}[e(k)-x(k)]^2}\right)$

2.4. Complexity of Algorithm

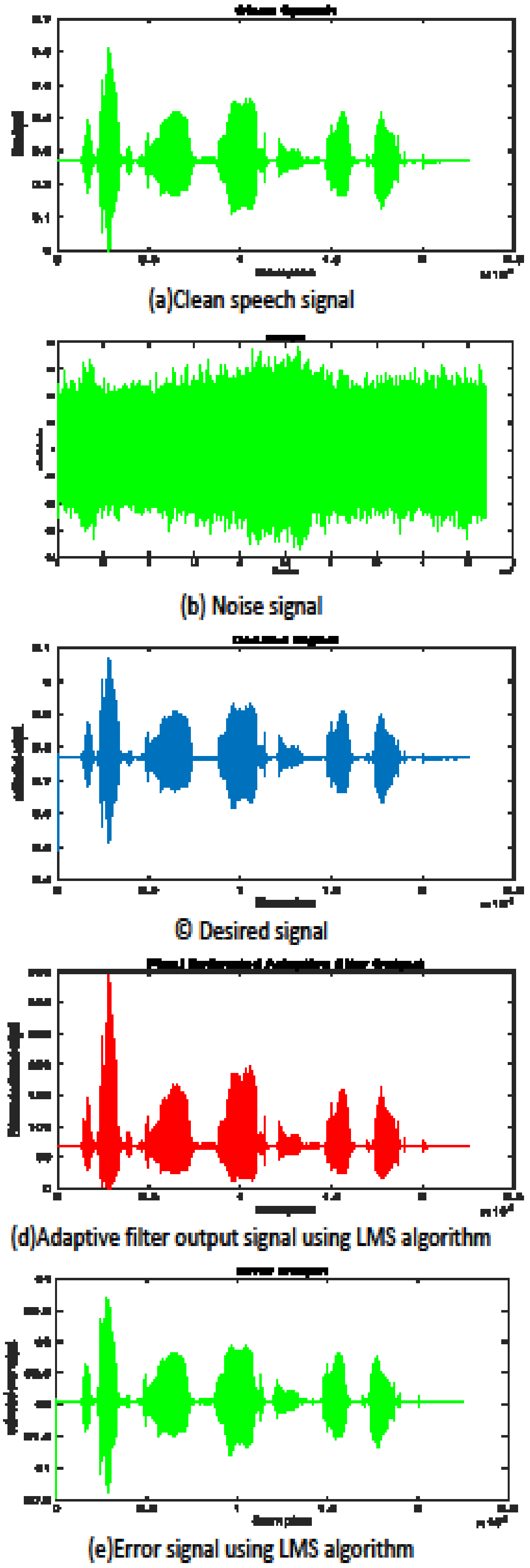
The computational complexity representation diagrams are essential to estimate LMS for all, as projected above are abridged in Table 2.

Table 2: Complexity of Algorithm

ALGORITHMS	MULTIPLIERS REQUIRED	ADDERS REQUIRED
LMS	2N+1	2N
NLMS	3N+1	2N

III. SIMULATION RESULTS AND DISCUSSION

Experiments were performed on noisy data which was prepared by adding noise signal to clean speech samples at 0dB, 5dB and 10dB SNR levels. These noise signals are taken from the NOIZEUS, TIMITT data bases for different noise environments such as machinegun, Factory, vehicle and Traffic noises. The time domain plot of the clean speech signal, noise signal, desired signal, estimated signal, and Error output signals are shown in Fig.2.



(a)Clean speech signal

(b) Noise signal

© Desired signal

(d)Adaptive filter output signal using LMS algorithm

(e)Error signal using LMS algorithm

Figure.2: Simulation results using LMS algorithm

Tables 3-5 shows the performance anaysis of adaptive noise cancellation (ANC) of LMS and NLMS algorithms in the parameters of SNR, MSE and RMSE. From these simulation results, it is observed that the NLMS algorithm was found to be a better optimal adaptive noise canceller for speech signal.

Table.3: SNR Values (in dB)

Type of Noise		Factory Noise	Machine Gun Noise	Vehicle Noise	Traffic Noise
LMS	0 dB	14.5196	12.7374	12.9859	13.0464
	5 dB	17.3156	15.1356	18.2136	17.4657
	10 dB	32.9263	32.9217	31.8696	31.869
NLMS	0 dB	15.8948	13.4497	6.3696	14.3503
	5 dB	17.7177	14.9669	15.3534	18.4473
	10 dB	40.7499	40.7342	38.5022	37.4031

Table.4 MSE values

Type of Noise		Factory Noise	Machine Gun Noise	Vehicle Noise	Traffic Noise
LMS	0 dB	0.0000022	0.00000201	0.00000204	0.0000024
	5 dB	0.000023	0.0000109	0.0000207	0.0000202
	10 dB	0.000104	0.000105	0.000177	0.000177
NLMS	0 dB	0.0000266	0.0000376	0.0000359	0.0000355
	5 dB	0.0000728	0.00000489	0.00000521	0.0000521
	10 dB	0.0000328	0.0000328	0.00000421	0.0000421

Table.5RMSE Values

Type of Noise		Factory Noise	Machine Gun Noise	Vehicle Noise	Traffic Noise
LMS	0 dB	0.0015	0.0014	0.0014	0.0017
	5 dB	0.0023	0.0021	0.017	0.0019
	10 dB	0.0102	0.0102	0.0133	0.0133
NLMS	0 dB	0.000516	0.000613	0.000599	0.000596
	5 dB	0.0027	0.0022	0.0023	0.0023
	10 dB	0.0057	0.0057	0.0065	0.0065

IV. CONCLUSION

In this paper, to analyse the speech enhancement performance with different noise reduction algorithms using adaptive filters like LMS, NLMS. Simulation were performed on noisy data which was prepared by adding machinegun, Factory, vehicle and Traffic noise at 0dB, 5dB and 10dB SNR levels to clean speech samples. The performance comparison of adaptive noise cancellation (ANC) system using LMS and NLMS algorithms was done in terms of parameters SNR, MSE and RMSE. Based on performance analysis, the NLMS algorithm was found to be a better optimal adaptive noise canceller for speech signal.

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