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# Study of Echo Cancellation approach by using Least Mean Square (LMS) Algorithm



# Study of Echo Cancellation approach by using Least Mean Square (LMS) Algorithm

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**Abstract.** Sound is the origin of communication. We are using sound channel for interaction. The acoustic signal introduces echo signal properties which lead the original signal as error signal. In most of applications the adaptive filters implemented in time domain works quite efficiently. However, the complication of the adaptive filter increases as the impulsive reaction becomes quite large hence it cannot be implemented efficiently in time domain. Acoustic echo cancellation is one example where this can happen. In this paper we will explain about to acoustic cancellation and different methods that are adopted to overcome the interference of noise by using various levels of algorithms and apply it to an error signal and makes solution for the problem, verifies the performance, describe the best and suitable technique which gives most effective results.
**Keywords:** least mean square algorithm (LMS), Normalized least mean square (NLMS), Noise cancellation (NC), Loudspeaker-Enclosure-Microphone (LEM), Mean square error (MSE).

## 1. Introduction

Adaptive filter and its performance on echo cancellation by using algorithm. Wireless communication is the channel provider to inter connects two users. Different signals radiate to free space as it was the medium of wireless channel, where noise interference with the desired signal which misleads the packed data. Apart from that noise interference involves reducing the performance of communication where as we refer active filters to control, eliminate the interference of noise [1] the speed depends upon statistics of input signal, it have low convergence with coloured signal and high with acoustic signal so that noise mainly effects the acoustic region [2] using short decorrelation filters at both the sides of output and input signals, we provides noise enhancement and speed instead effected by noise [3], where the one-tap predictor is reassigned using the least mean square LMS scheme. The pseudo affine projection algorithm also applied to the pre-whitening technique, and its mean-square analysis was given in as we face stability problem in filtering stage like alpha, interference of noise was the main problem hence we go for LMS in order for best results in stability and noise elimination.

Algorithm one tap predictor came later to reduce the draw backs which come across LMS. Adaptive filters took a part in communications and in control systems where they are used in different areas like identifyingthe system, Active noise control of signal, echo cancellation, beam forming. One thing we came across that least mean square depends upon the input signal power. When comes to competition normalized least mean square have more advantages over least mean square. To make this successful and reduce the drawback different types of approaches have been introduced, they are sub-band adaptive filter, algorithm for affine projection andtransform domain adaptive filter. Prewhitening are those methods used in algorithm at input and output signals at short filter decorrelation which provides good convergence and signal speed with respective of noise. Using algorithms may suffer small stability problems when comes to speech and acoustic signals it was more decor-related parameters, decor-related normalized least mean square (DNLMS) was further extended to PDNLMS, the noise was more under these conditions such as audio processing, acoustic under water, power line broadband communications.

### 1.1. Algorithm

Here, we develop an algorithm on robust variable step size decorrelation least mean square by taking input signal and weight vectors

**Input Signal:**

For the case of multiple input

$$X_k = [x_{0k}\ \ x_{1k}\ \ \ldots\ldots\ \ X_{Lk}]^T \tag{1}$$

**Weight Vectors:**

Corresponding to weight vector input signal are

$$W_k = [w_{0k} w_{1k}\ \ \ldots\ldots\ \ W_{Lk}]^T \tag{2}$$

So, on multiplying equation (1) and (2), we get output signal as

$$Y_k = X_k^T\, W_k = W_k\, X_k^T \tag{3}$$

and since error signal is

$$e_k = d_k - y_k \tag{4}$$

Calculating error signal from equation (3)

$$e_k = d_k - X_k^T\, W = d_k - W\, X_k^T \tag{5}$$

On squaring equation (5), the instantaneous squared error is

$$e_k^2 = d_k^2 + W^T\, X_k\, X_k^T\, W - 2d_k\, X_k^T\, W \tag{6}$$

from equation (6) calculating the expected mean square value as mentioned in equation (7)

$$E\left[e_k^2\right] = E\left[d_k^2\right] + W^T E\left[X_k X_k^T\right] W - 2E\left[d_k X_k^T\right] W \tag{7}$$

$X_kX_k^T$is the autocorrelation matrix of the input signal and it is represented by term R as given in the equation (8)

$$R = E\left[X_kX_k^T\right] = E\begin{bmatrix} x_{0k}^2 & x_{0k}x_{1k}\cdots & x_{0k}x_{1k} \\ \vdots & \ddots & \vdots \\ x_{Lk}x_{0k} & x_{Lk}x_{1k}\cdots & x_{Lk}^2 \end{bmatrix} \tag{8}$$

Cross terms are cross correlation among the input components and diagonal terms are mean square of input components. Next term $E[d_kX_k]$ is the cross correlation between the input signal and desired signal and it is expressed by P and represented in equation (9)

$$P = E[d_kX_k] = E[d_kx_{0k}d_kx_{1k} \dots\dots\ d_kx_{Lk}]^T \tag{9}$$

Now calculating mean square error from equation (7)

$$MSE \triangleq \zeta = E\left[e_k^2\right] = E\left[d_k^2\right] + W^TRW - 2P^TW \tag{10}$$

Gradient and Mean square error:

$$\nabla \triangleq\ \partial\zeta / \partial W\ = [\partial\zeta / \partial w_0\quad \partial\zeta / \partial w_1\ \ \partial\zeta / \partial w_L\ ]^T \tag{11}$$

$$=2RW–2P \tag{12}$$

To obtain the minimum mean-square error the weight vector W is set at its optimal value W*, where the gradient is zero.

$$\nabla = 0 = 2RW*2P \tag{13}$$

W* is wiener weight vector

$$W* = R^{-1}P \tag{14}$$

1 Mean square error is obtained by substituting W* from equation (14) for W in equation (10)

$$\zeta min = E\left[d_k^2\right]+W *^T\ RW * -2P^TW * \tag{15}$$

$$= E\left[d_k^2\right]+ [R^{-1}P]^TRR^{-1}P - 2P^TR^{-1}P \tag{16}$$

On simplifying equation (16) we get

$$\zeta min = E\left[d_k^2\right] – P^TR^{-1}P = E\left[d_k^2\right]–P^TW* \tag{17}$$

### 1.2. Acoustic Echo:

Acoustic echo is a local audio loop back when the microphone picks up the audio signal from the speaker and directs it to the original contributor. In this case the original contributor will hear their own voice as they speak. Acoustic echo is more prominent when sensitive microphone is used as well as in the case where the speaker volume is at a high level and also when the microphone and speaker are placed near each other. This type of echo is called as reflective acoustic echo.

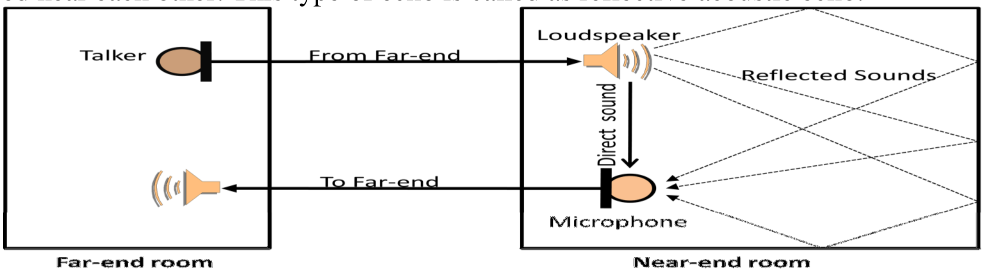


**Fig1: A teleconference system with echo paths of room**

In this paper investigation of LMS, NLMS, Adaptive filters is carried out. The brief overview of all the methods of noise cancellation was explained in below section. They explore the comparison between various filters and algorithms which are used in real life applications, finally we designed an algorithm and concluded the session.

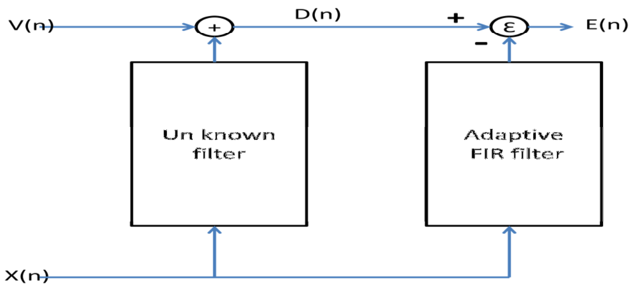


**Fig2: System identification model based on adaptive filter**

## 2.Literature:

The acoustic echo signal cause interference and have a chance of reducing the quality of the communication and effects the data at the end receiver. The acoustic coupling disturbance that takes place between the microphone and loudspeaker in an enclosed place like say Hall, is called a Loudspeaker-Enclosure-Microphone (LEM) system. In the Loudspeaker-Enclosure-Microphone model the signalsproduced by the loudspeaker arrives at the microphone directly, as well as the reflected signals from the surrounding objects and walls. Different types of methods are mentioned for acoustic noise cancellation which depending upon the method. We use filters and algorithms of various types like adaptive filter X algorithm here it helps in reduction of noise and provides an linear signal with respective to input variations [5] Variable step-size algorithm came into image in order to provide the best results for echo cancellation by designing suitable algorithm consists adaptive filters makes this algorithm to resist from echo. [6] Echo signal is undesired signal which makes delay in speech signal where echo interference is the major problem that comes across communication channel which disturb operators. To suppress the echo, we use adaptive filtering techniques which results in improvement in signal quality using TMS320C6748 DSK [7]Volterra adaptive filter is ahalf-duplex method which suppress the echo signal. Echo is the main problem when the equipment used was not standard. Non linearity occurs during this case, still we can eliminate them using linearity filters by using linear and nonlinear algorithm like NLMS, variable step size LMS and second order Volterra we eliminate echo [8] sub band filtering is another method to decrease the influence of noise based on mean square error MSE, controlling the parameters of MSE eliminates the acoustic echo using of algorithm by this process would be more easer. [9] Split Kernel Adaptive Filtering Architecture is a method which reduce the performance of acoustic noise based on frame network and kernel based adaptive filter can be work effectively. The structure can estimate the path echo [10] by using large scale digital signal processors. Thus, the acoustic echo problem was solved. However, such system is expensive and should be applied before hand with audio devices, For Voice Quality Enhancement we use DSP Processor in this method for eliminating the background noise and echo cancellation. By using this method improves voice intelligibility and reduce listening effort. Echo cancellation was poor when connecting two devices like microphone and speaker in hand free devices this can be achieved by algorithms.[11] Stereophonic Acoustic Echo Cancelation is a type which comes across in teleconference systems. The main reason beingthat the two input audio channels have a strong cross correlation.The effect of correlation can be reduced by introducing nonlinearities. This is a good linear method to de-correlate the two-channel input signals andevade the unrequired nonlinear disturbances. Introducing two channel low complication adaptive filter algorithms into image which results higher change than LMS algorithm and elimination of noise became easier. [12]

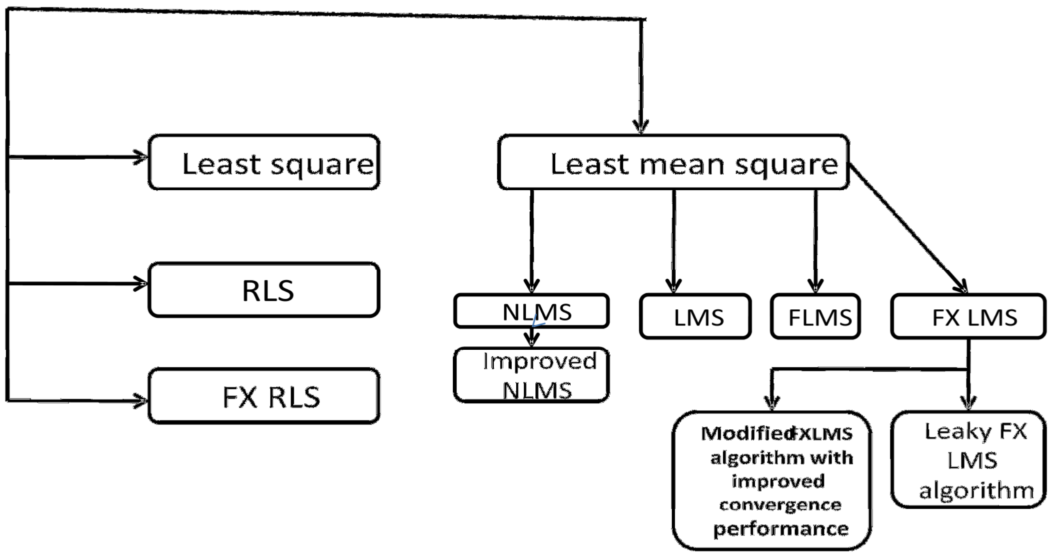


**Fig3: Existing algorithm**

Affine Projection Algorithm for Acoustic Echo Elimination is a method which is used widely for noise cancellation whereas here it offers high convergence rates and tracingabilities [13]. Long-time echo paths with respective time implies environment of acoustic signal. Background noise and double talk are low misadjusted. Different attractive versions of acoustic echo cancellation however step-size parameter have the performance criteria. APA represents most reliable solution. [14, 15]

## 3. **Methodology:**

Benefits of Adaptive filters and Algorithms

- Computed in real time application
- Minimization in mean square estimation error

Here we define robust variable step-size decorrelation by using NLMS algorithm. This process explains the way how the noise was eliminated by passing signals throw adaptive filters and by creating algorithms, taking different references papers we provide the best way to noise free interaction. Providing ‘n’ no of methods make easy for acoustic cancellation with desired conditions as we have wide range of users interference of noise became easy to disturb the channel of interaction but here we provide solution in order to retain we introduced various level filters such can divide the source data form noise interference, below mentioned are the various stages which noise can interfere. In below figure we can see clearly that noise interference occurs in channel.
Where it was caused due to using of unstable system and its performance on input signal in a feedback amplifier, a portion of the output is deducted from the input and then this error signal is gained up by a large amount to produce the output. The output is not instantaneous with respect to a change in input

there is some delay. As the frequency is increased this delay from input of the amplifier to the output results in a phase shift.

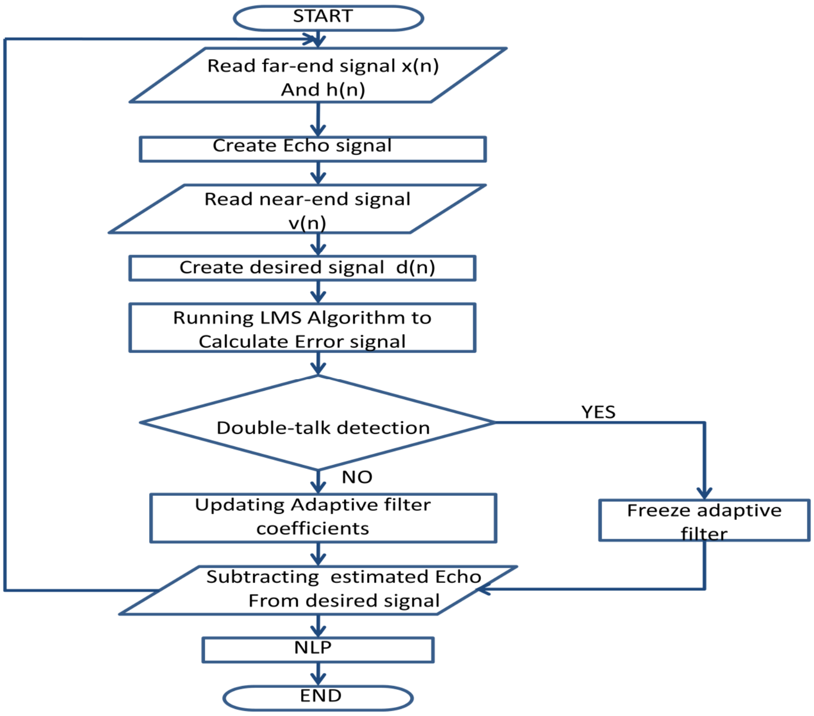


**Fig4: Flowchart of Acoustic Echo Cancellation algorithm**

Eventually the phase shift is sufficient to invert the feedback. So, the negative feedback is turned into positive feedback. If the amplifier still has gain greater than unity, then the signal will continue to increase in amplitude and oscillate. Prior to the point of oscillation, the gain could be “close to unity” but still less than unity. In this case, the signal will tend to peak in the frequency response amplitude due to this positive feedback.When we consider the below block there are different registers, here in register-1 consists data input, clock, enable input and reset input. The same inputs are allowed to register-2 but the difference between them was here we provide desired signal as input signal instead of data input.With parallel to itwe have a multiplierblock (M) where the output of register-1 was available as input feed to both M2, M3. R2 data was processed through subtract unit and then multiplied through M1, M2blocks and then processed with an adder block which combines the data and it results the error signal, and the same signal was taken as a reference signal with step input to M1. So that make easy to differentiate the error signal.

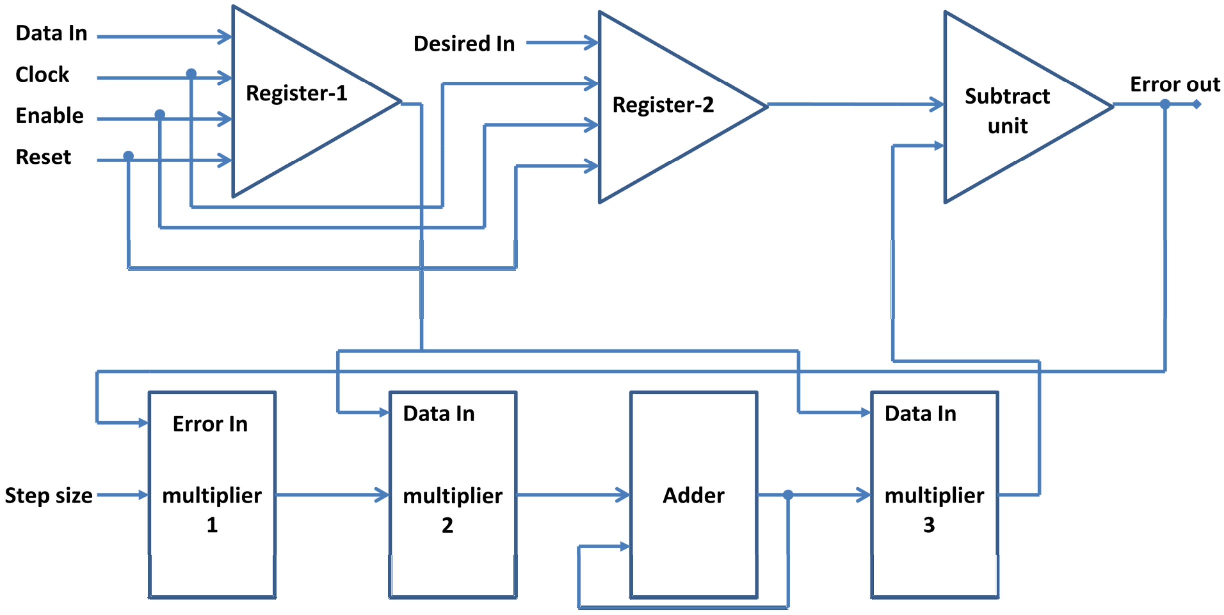


**Fig5:Adaptive filter using LMS algorithm**

## 4.Result

We have mentioned accuracy improvement in acoustic cancellation using various techniques and proposed the best way which overcomes the problem. Here below are the algorithms proposed and their respective mathematical equations, in order of successive performance, we consider every mathematical ideology the RLS is best of all for SNR rejection but we have a drawback here that is required complex operations, Speech characteristics is the performance for function. Furthermore, the algorithms may be influenced by the noise power level and hencefurtherresearch is required.

| **Algorithm** | **Multiplication** | **Addition** | **Square root** | **Comparison** |
|---|---|---|---|---|
| RVSSNLMS | *2L+8* | 2L+1 | 2 | 2 |
| Pseudo-AP | $2L+1+KL+\varphi(K)$ | 2L+KL+φ(K) | 0 | 0 |
| NLMS | *2L* | 2L | 2 | 0 |

**Applications:**

- System Identification
- Modelling Inverse System
- Signals Prediction
- Interference Cancelation

## 5. Conclusion:

In this review paper we have proposed an NLMS algorithm with the existing algorithms, according to that we have updated its weight vectors decor relatedposterior error signal with the help of norm optimization. Applying analysis theoretically shows that the defined algorithm is stable and converge the LMS with a real system.

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