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# New Insights of RLS For Noise Cancellation in Speech signal

[1] Vuyyuru Tejaswi , [2]Sharma.S, [3]Srikanta Nallapaneni, [4]Surendar.A

[1] [2] PG scholar School of Electronics, Vignan's University, Guntur, Andhra Pradesh.

[3] [4]Assistant Professor,School of Electronics, Vignan's University, Guntur, Andhra Pradesh.

[1]tejaswireddy.456@gmail.com, [2]Sharma_ece@vignanuniversity.org, [3]nsk_ece@vignanuniversity.org, [4]surendararavindhan@gmail.com

*Abstract*: ***Speech is the basic way through which information is exchanged among the people. Humans have an audible frequency range of 20 HZ to 20,000HZ and Human speech ranges from 300HZ to 3400 HZ. In this regard one of the most common way that speech is affected is due to background noises such as airport, car, babble, restaurant, street, train station, exhibition noises. As speech is a non-stationary signal which means its frequency or its spectral components changes with respect to time. When the speech and noise signals combines and change continuously to extract the desired speech from it there arises the need for adaptive filters. One such highly popular adaptive filter is Recursive Least square (RLS).Here in this paper a simulink model is designed and implemented for noise cancellation using RLS algorithm and tested for different background noises and the MSE(Mean Square Error) and Signal to Noise Ratio(SNR) are calculated. The speech signals have been taken from NOIZEUS database which contains different signals contaminated with noise. Finally , we can be observe that noise reduction has been achieved audibly.***

***Keywords: Adaptive filter, Recursive Least square (RLS) algorithm, NOIZEUS database***

## I. INTRODUCTION

In speech processing the most commonly occurring problem is due to noise. Noise is an unwanted signal which interferes with the normal operation of a device. This noise has been becoming a problem to the communication system performance and due to its random nature the design of system is also becoming difficult. Noise can be removed by using appropriate filtering techniques. The most commonly used filtering techniques are fixed and adaptive [1]. In Fixed filters the signal and noise must be known beforehand, so that a filter can be designed which passes the signal frequency and rejects the noise frequency. Adaptive filters on the other hand, have the ability of adjusting the filter coefficients to reduce the noise mixed with the signal. Adaptive filters requires no prior knowledge of both signal and noise characteristics.[3] Nowadays Adaptive filters are most commonly used for non-stationary signals. Adaptive filters are used in devices like mobile phones, medical equipment, and digital cameras.

### ADAPTIVE FILTERING

Adaptive filter is combination of digital filter with adaptive algorithm which is used to adjust the filter coefficients to minimize the error [4]. Adaptive filters are used in situations:

1. When the filter coefficients are to be changed
2. When there is an overlap between signal and noise spectrums.
3. The band occupied by noise is not known and it varies with time.

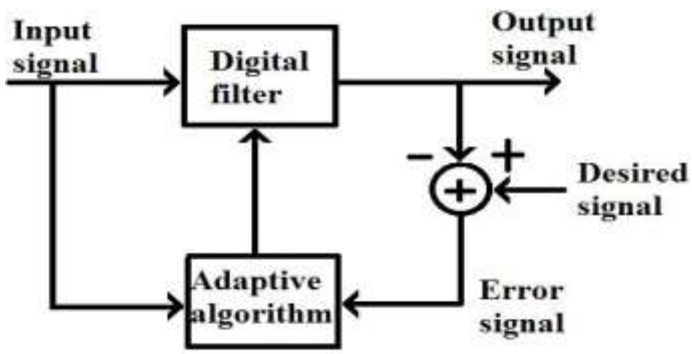


**Fig.1. Block diagram of Adaptive Filter**

In the diagram the input signal is given to a digital filter the output from the digital filter is compared with desired signal and if there occurs error, it is given to adaptive algorithm which adjusts the filter coefficients to minimize the error. The algorithm can be LMS, NLMS, and RLS. Here in this paper RLS is used because it has faster convergence rate and reduces error efficiently when compared to other algorithms.[7][8]

## RLS ALGORITHM

The RLS algorithm recursively calculates the filter weights to reduce the least square cost function relating to input signal.[6] In this the present filter coefficients updates by using the past filter coefficients The RLS algorithm provides faster convergence rate compared to LMS algorithm.

The RLS algorithm performs the following operations to update the filter coefficients.

1. It computes the output signal y(n) .
2. It computes the error signal e(n) which is the difference between desired and output signal.

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

3.The filter coefficients are updated by using the equation

$$\vec{w}(n+1) = \vec{w}(n) + e(n) \cdot \vec{K}(n)$$

Where $\vec{w}(n)$ is the filter coefficients vector

$\vec{K}(n)$ is the gain vector.

$$\vec{K}(n) = \frac{P(n) \cdot \vec{u}(n)}{\lambda + \vec{u}^T(n) \cdot P(n) \cdot \vec{u}(n)}$$

Where, $\lambda$ is the forgetting factor

$P(n)$ is the inverse correlation matrix of the input signal.

$P(n)$ has the following initial value $P(0)$:

$$P(0) = \begin{bmatrix} \delta^{-1} & & & 0 \\ & \delta^{-1} & & \\ & & \ddots & \\ 0 & & & \delta^{-1} \end{bmatrix}$$

Where $\delta$ is the regularization factor. The equation for updating the inverse correlation matrix is :

$$P(n+1) = \lambda^{-1} P(n) - \lambda^{-1} \vec{K}(n) \cdot \vec{u}^T(n) \cdot P(n)$$

RLS algorithms compute $J(n)$ by the equation $J(n) = \frac{1}{N} \sum_{i=0}^{N-1} \lambda^i \; e^2(n-i)$

Here $N$ is the filter length and $\lambda$ is the forgetting factor.

The forgetting factor has a range of (0, 1].The $E[e^2(n)]$ is more accurate for RLS when compared to LMS.

## RLS SIMULINK MODEL

In this paper RLS algorithm [5] is used for background noise cancellation and it is performed in Matlab Simulink. The RLS block is available in DSP system toolbox in simulink. The Simulation Parameters:

Filter length=2

Forgetting factor=1

Initial Input Variance Estimate=0.1

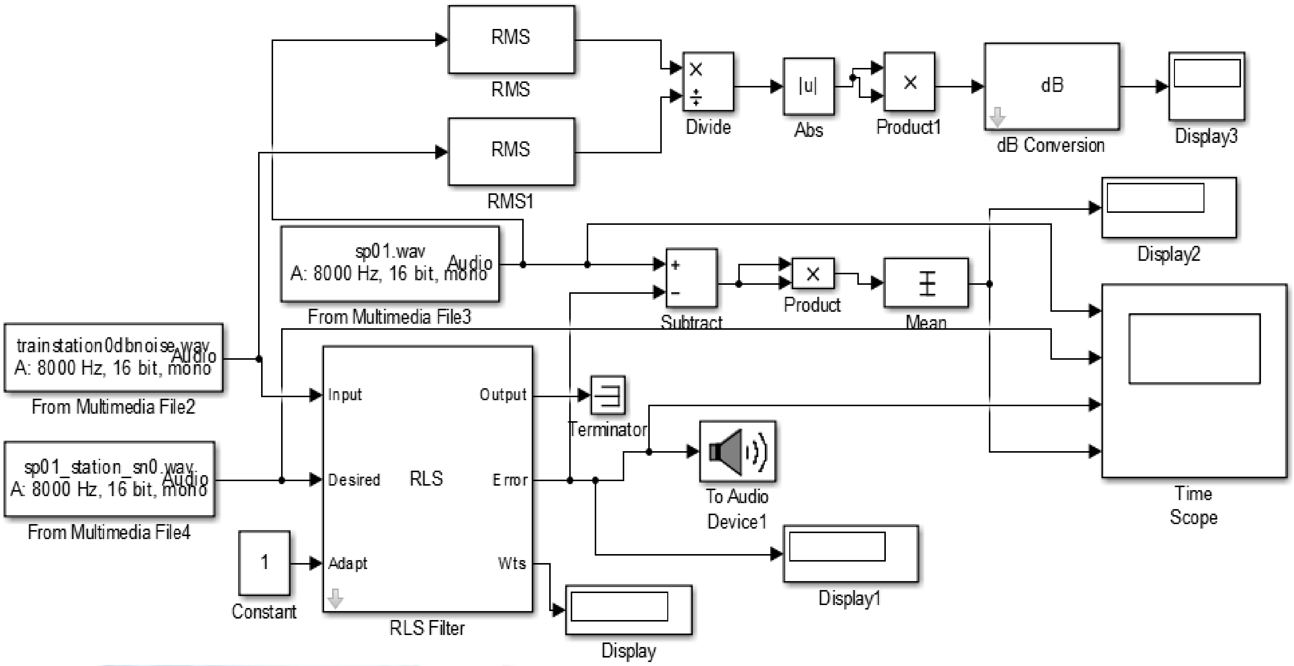


**Fig.2. Simulink model for noise cancellation using RLS block**

The important blocks are:

**From Multimedia File**: This is used to read the audio file.

**RLS block**: In this the input must be connected with a reference signal which is nothing but a noise signal. The desired signal must be a noisy speech that is signal containing both speech and noise.

If the Adapt port value is 1 then it continuously updates the filter weights. If it is 0 then filter weights remains constant.

Fig.2. is the simulink model for RLS and the noise used here is train station noise the speech signal is taken from NOIZEUS database [9][2]. Which contains clean speech signals and speech signal mixed with noise which is nothing but noisy speech signal. Here sp01 speech signal is taken and the sentence uttered is “The birch canoe slid on the smooth planks."The noise alone can be obtained by using the subtract block in simulink and subtracting clean speech from noisy speech. The algorithm is tested for different background noises like airport, babble, car, street, restaurant, exhibition noises by following the same procedure as in fig.2.

***Performance Measures used for simulation:***

1. Mean Square Error: It is defines as the difference between estimator and what is estimated.

2. Signal To Noise Ratio (SNR): It is defined as the ratio of signal power to noise power (or) Root mean square amplitude of signal to noise squared.

$$SNR=P_{Signal}/P_{Noise}=(A_{Signal}/A_{Noise})^2$$

## RESULTS

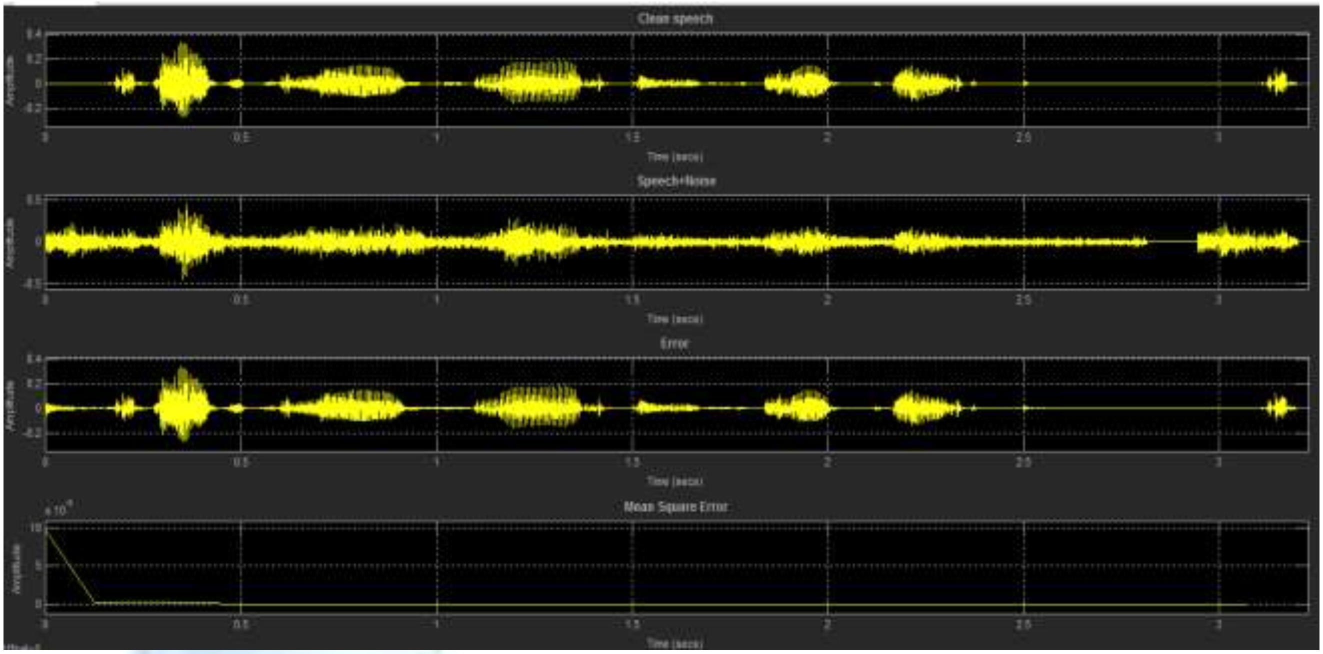


**Fig.3. Simulink output for train station noise**

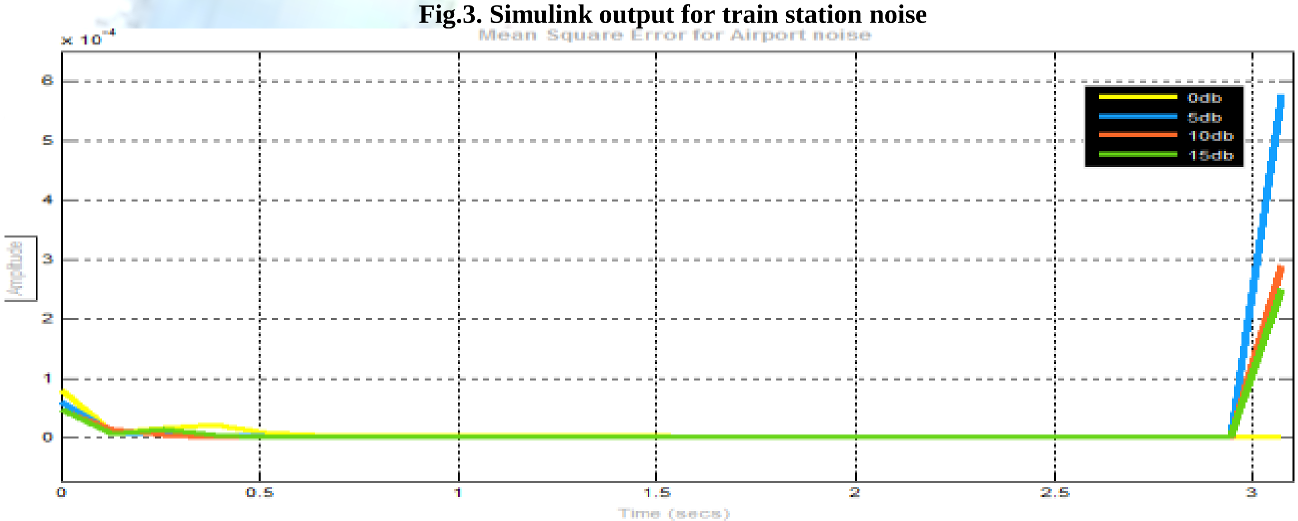


**Fig.4. Mean square error for 0, 5, 10, 15 db AIRPORT noise**

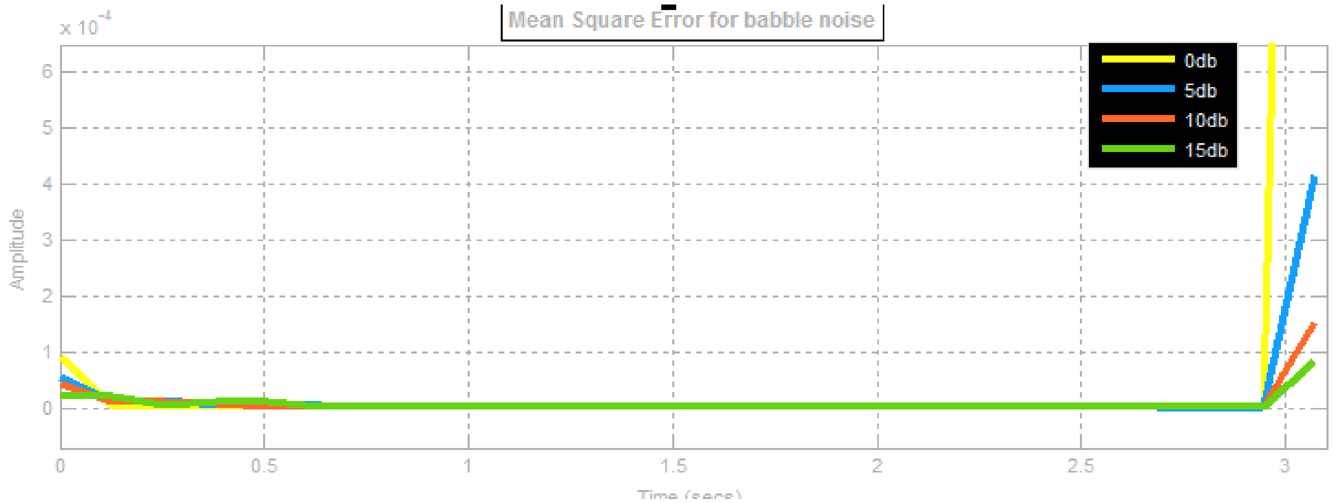


Fig.5. Mean square error for 0,5, 10, 15 db BABBLE noise

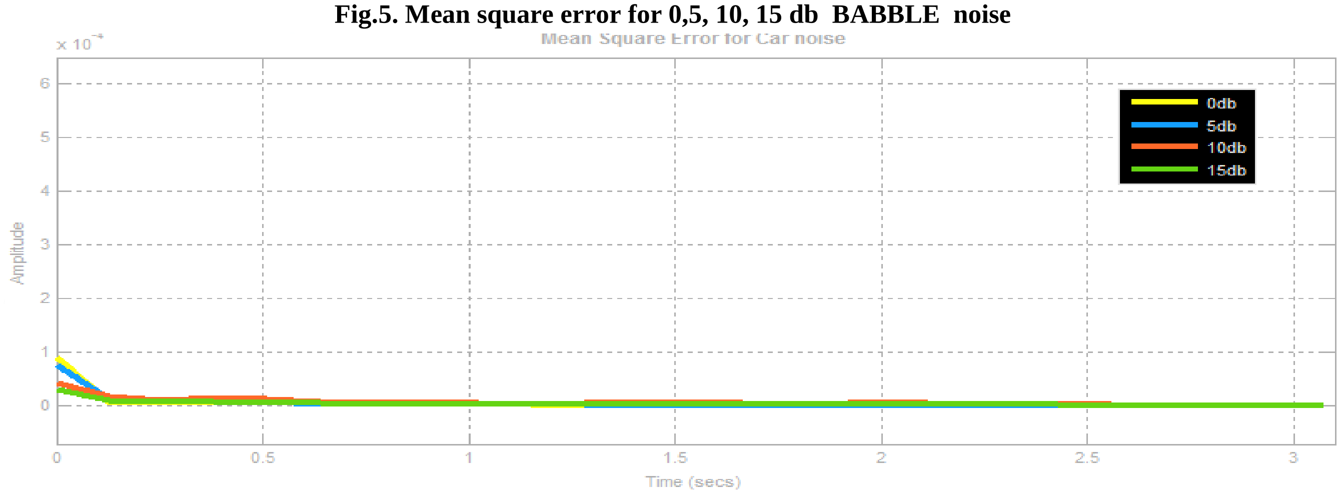


Fig.6. Mean square error for 0, 5, 10, 15 db CAR noise

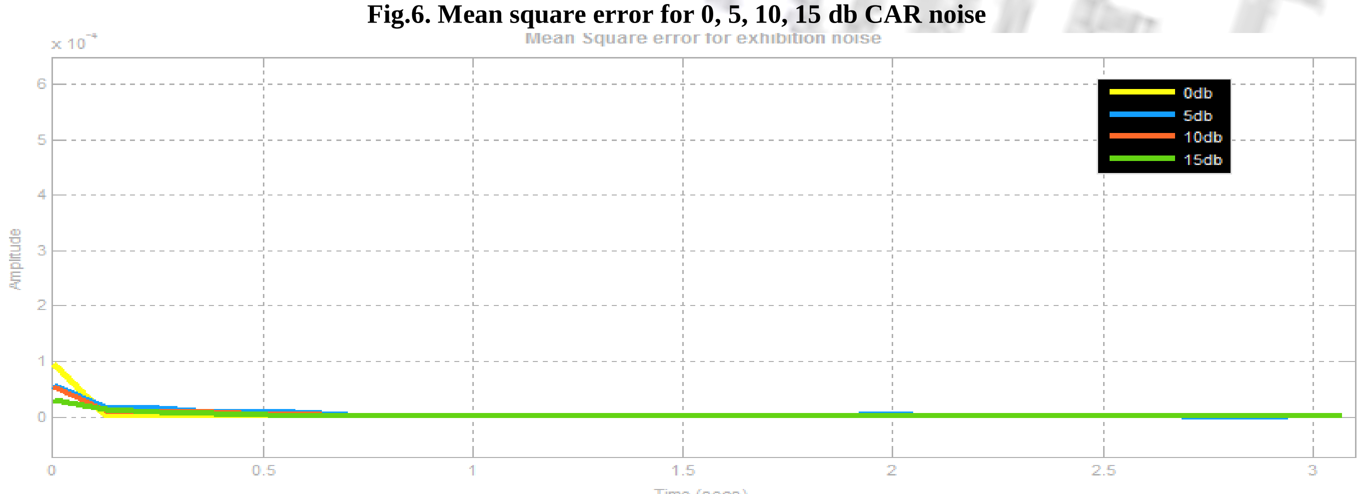


Fig.7. Mean square error for 0, 5, 10, 15 db EXHIBITION noise

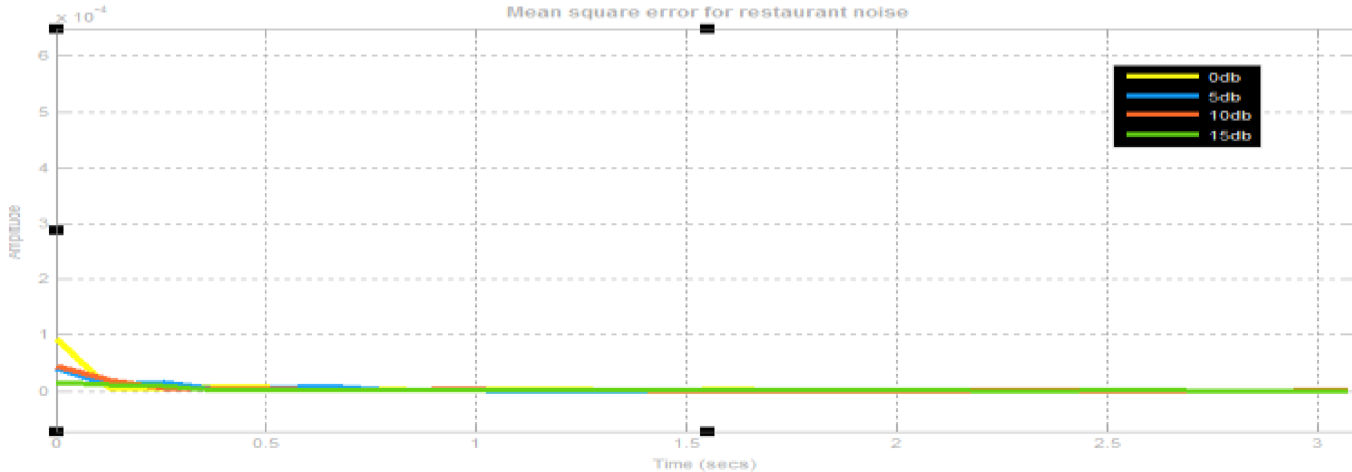


Fig.8. Mean square error for 0, 5, 10, 15 db RESTAURANT noise

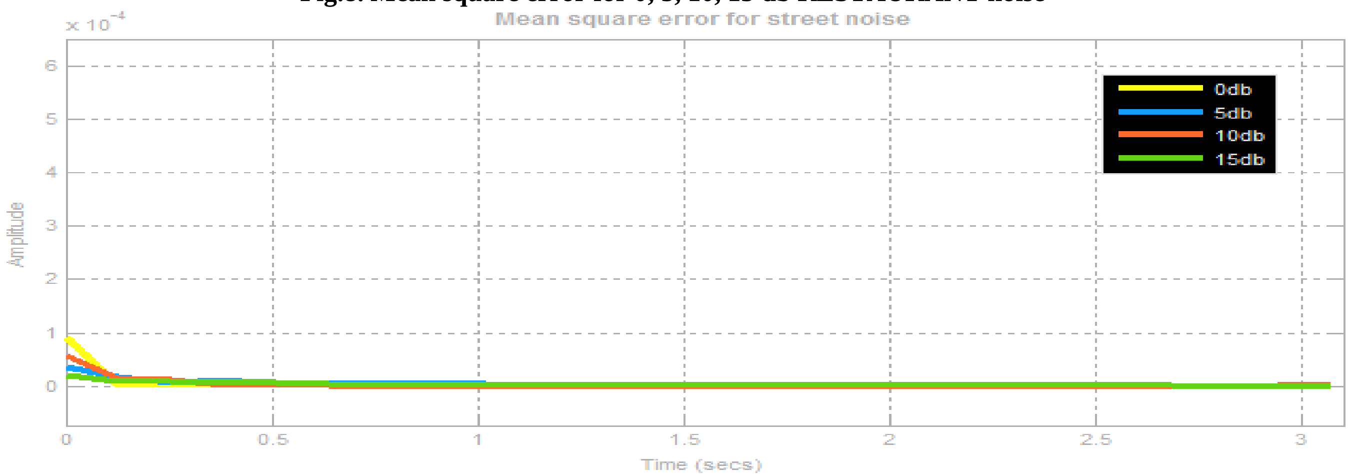


Fig.9. Mean square error for 0, 5, 10, 15 db street noise

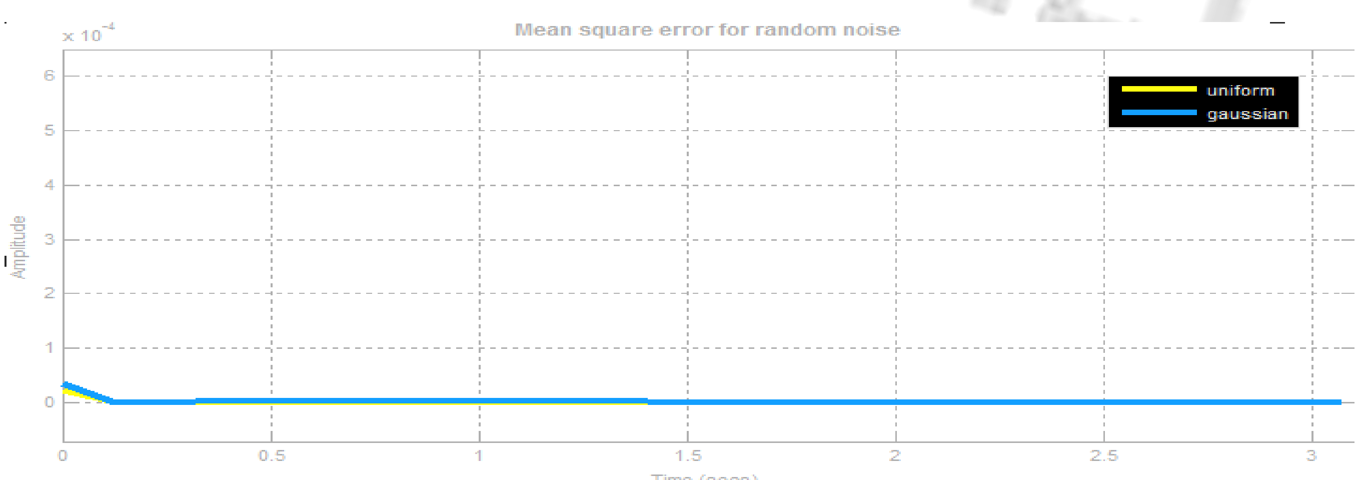


Fig.10. Mean square error for RANDOM noise

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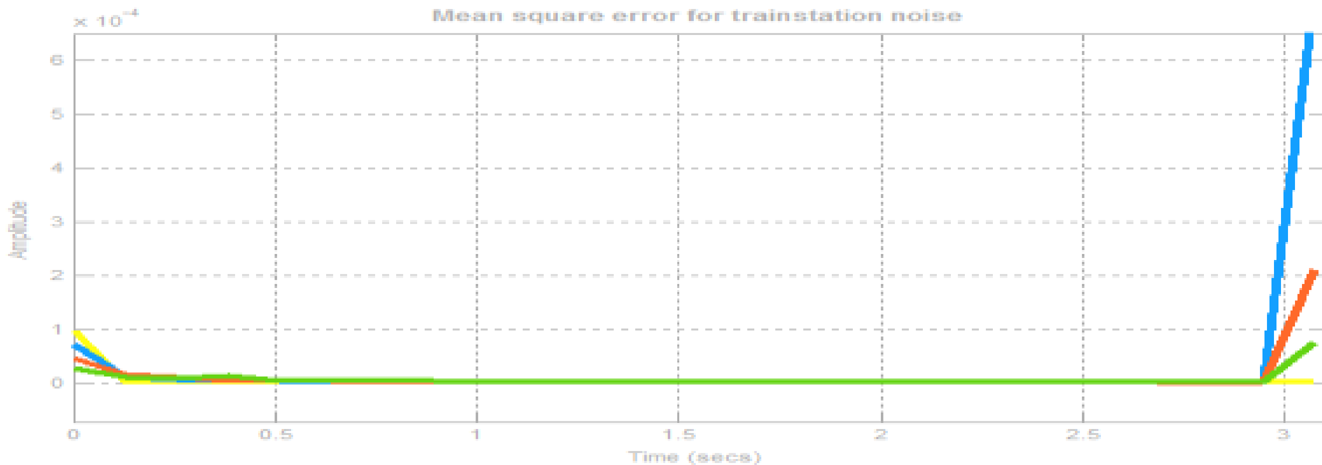


**Fig.11. Mean square error for 0, 5, 10, 15 db train STATION noise**

**TABLE: Shows the MSE and SNR for all Noises**

| SL.NO. | Noise Type | | MSE | SNR |
|---|---|---|---|---|
| 1. | Colored Noise | Pink | 1.597exp-07 | -35.65 |
| | | white | 2.696exp-07 | -26.75 |
| | | brown | 1.5904exp-07 | -63.19 |
| | | blue | 2.0436exp-07 | -23.44 |
| | | purple | **6.7482exp-09** | -30.92 |
| | | custom | 1.5972exp-07 | -35.67 |
| 2. | Random Noise | Uniform | **4.0001exp-08** | -29.73 |
| | | Gaussian | 4.4547exp-07 | -29.34 |
| 3. | Airport | 0db | 1.8719exp-03 | -2.545 |
| | | 5db | 5.7754exp-04 | 4.5587 |
| | | 10db | 2.9025exp-04 | 9.5213 |
| | | 15db | **2.5076exp-04** | 13.5548 |
| 4. | babble | 0db | 3.3537exp-03 | -6.085 |
| | | 5db | 4.1484exp-04 | 4.6140 |
| | | 10db | 1.5358exp-04 | 9.5771 |
| | | 15db | **8.4269exp-05** | 14.5868 |
| 5. | Car | 0db | **7.3402exp-08** | -4.907 |
| | | 5db | 1.0098exp-07 | 4.4489 |

| | | | | |
|---|---|---|---|---|
| | | 10db | 4.7719exp-07 | 9.6175 |
| | | 15db | 3.4568exp-07 | 14.5030 |
| 6. | Exhibition | 0db | 5.5359exp-08 | -6.336 |
| | | 5db | **3.8918exp-08** | 5.098 |
| | | 10db | 1.601exp-07 | 6.533 |
| | | 15db | 3.6606exp-07 | 11.2 |
| 7. | Restaurant | 0db | 9.1954exp-07 | -7.119 |
| | | 5db | **1.4072exp-08** | 7.436 |
| | | 10db | 1.0959exp-06 | 2.281 |
| | | 15db | 2.0094exp-07 | 12.39 |
| 8. | Street | 0db | 2.4340exp-07 | -2.924 |
| | | 5db | **8.3330exp-08** | 6.181 |
| | | 10db | 1.7256exp-06 | 1.728 |
| | | 15db | 1.9900exp-07 | 13.3 |
| 9. | Train | 0db | **6.2757exp-08** | -3.566 |
| | | 5db | 7.1779exp-04 | 1.398 |
| | | 10db | 2.1084exp-04 | 6.612 |
| | | 15db | 7.4603exp-05 | 10.73 |

## CONCLUSION

Through performing the simulation it is observed that if the forgetting factor increases then MSE decreases and SNR increases. The simulation is performed for all kinds of noise and their corresponding MSE and SNR are calculated. Hence the RLS algorithm provides faster convergence rate and minimizes the error more efficiently when compared to other algorithms like LMS, NLMS. The future work is to implement this on hardware to get effective results. The hardware can be of any DSP boards or on FPGA.

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