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# ADAPTIVE NOISE CANCELLATION TECHNIQUE USING LMS,RLS AND WAVELET TRANSFORM

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## ABSTRACT

This paper describes the concept of adaptive noise cancelling (ADC), in alternative scheme of estimating signals corrupted by additive noise or interference. The main objective of the noise cancellation is to estimate the noise signal and to subtract from original input signal plus noise signal and to obtain the noise free signal. An alternative scheme called adaptive noise cancellation for estimating a speech signal corrupted by an additive noise or interference. This technique uses a primary input signal that contains the speech signal and a reference input containing noise. Its reference input is adaptively filtered and subtracted from the primary input signal to obtain the estimated signal. In this technique the desired signal corrupted by an additive noise can be recovered by an adaptive noise canceller using least mean square(LMS) algorithm and recursive least square(RLS) algorithm. This adaptive noise canceller is useful to improve the signal to noise ratio.

**KEYWORDS**: Noise cancellation, LMS algorithm, RLS, Adaptive filter. etc.

## INTRODUCTION

One of the most problems that affect almost all of our daily life is the background noise, Such as Transportation aircraft, vehicles, railroad stock, buses, trucks, automobiles, and motorcycles all producing excessive noise. These sources can be fall under some kind of noise; it may be white, colored or impulsive noise according to its power, and at what range of frequency it is occur.

The noise is unwanted and undesirable changes to our environment, in needed arise to get rid of noise, many attempts were held for this reason; one of the most effective methods is the adaptive noise cancellation.

An adaptive filter is a digital filter technique that can adjust its coefficients to give the best match to a given desired signal. The adaptive filter operates in a changeable atmosphere the filter coefficients can adapt in response to changes in the applied input signals source. In adaptive filters depend on recursive algorithms to update their coefficients and train them to near the optimum solution.

As compared to the conventional filter design techniques, adaptive filters do not consist of constant filter coefficients and previous in sequence is also unknown. Therefore the filter with adjustable parameters is known as adaptive filter. The adaptive filter self adjusts their coefficients to minimize the error signal, which can be accomplished as infinite impulse response (IIR), finite impulse response (LMS), lattice and transform domain filter. The finite impulse response (LMS) filter is the simplest one as it minimizes the instantaneous square error and simple gradient-based optimization method. Among all the other algorithms, the Least Mean Square (LMS) algorithm is most suitable for the adaptation of the filter coefficients schemes. Thus, the objective of operating under changing conditions and readjust itself continuously to minimize the error is fulfilled by the finite impulse response algorithm.

## NOISE CANCELLATION

Noise cancellation is a scheme of figuring out signals corrupted by additive noise or interference. From the figure 1 given below it is clear that this system is comprised of primary as well as reference input. Input to the primary one is signal source and indirectly noise source as well, Input to the reference one is Noise source.

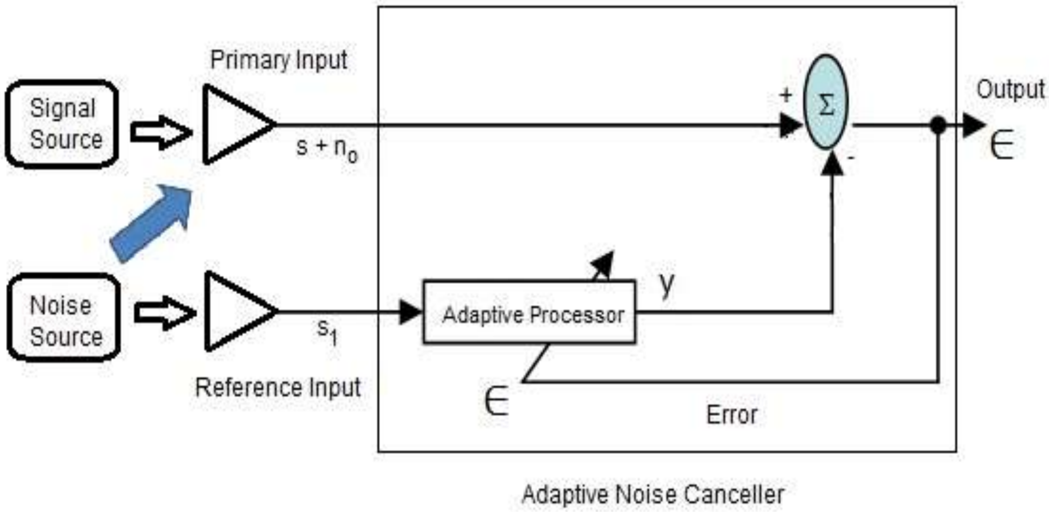


***Fig. 1 Adaptive noise cancellation system [2]***

From the output (z) can be analyzed which is a combination signal and noise with adaptive filter.

$$z = s + n0 - y \tag{1}$$

Squaring both the side we get

$$z^2 = s^2 + (n0 - y)^2 + 2s\,(n0 - y) \tag{2}$$

$$E[z^2] = E[s^2] + E[(n0 - y)^2] + 2E[s(n0 - y)]$$
$$= E\,[s^2] + E\,[(n0 - y)^2]$$

The signal power E [s2] will be unaffected as the filter is adjusted to minimize E [z2]. Accordingly, to the minimum output power is eq. 3

$$\min E[z^2] = E[s^2] + \min E[(n_0 - y)^2] \tag{3}$$

Thus When the filter is adjusted so that E $[z^2]$ is minimized, E $[(n_0\text{-}y)^2]$ is, then, also minimized. The filter output y is then a best least squares estimate of the primary noise no. Moreover, when E $[(n_0\text{-}y)^2]$ is minimized, $[(z\ \text{-}s)^2]$ is also minimized,

Therefore

$$(z - s) = (n0 - y). \tag{4}$$

Adjusting or adapting filter to minimize the total output power system is thus tantamount to causing the output z to be a best least squares estimate method of the signal s for the given structure and adjustability of the adaptive filter and for the given reference input signal. The output z will contain the signal s plus noise. The output noise is given by (n0 -y). Since the minimizing E $[z^2]$ and minimizes E $[(no - y)^2]$ minimizing the total output power minimizes the output noise power. As the signal in the output remains constant, and minimizing the total output power maximizes the output signal to noise ratio (SNR).

The noise cancellation can be achieved by various other techniques, as we have reviewed that step size variation plays an important role for noise cancellation process. In other parameters are also associated which can be adjusted in order to achieve noise cancellation. Next we will discuss proposed methods for the same.

## LEAST MEAN SQUARE (LMS) ADAPTIVE FILTER

The LMS adaptive filter is used worldwide because of its easy computation and flexibility. In this algorithm is a member of stochastic gradient algorithm, and because of its robustness and low computational complexity it is used worldwide. The algorithm using the steepest distance is as given below.

$$P\,(n+1) = w\,(n) - \mu\,\nabla J\,(P\,(n)) \tag{5}$$

Where

$$\nabla J\,(P\,(n)) = -2p_{dx} + 2R_x w\,(n) \tag{6}$$

$R_x$ and $p_{dx}$ are the instantaneous estimates and are defined as

$$\mathbf{Rx} \cong \mathbf{x}(n)\mathbf{x}^T(n),\ \mathbf{p_{dx}} \cong \mathbf{d}(n)\mathbf{x}(n)$$

Substituting equation (1) and (2) in the above equation we get

$$P(n+1) = p(n) + 2\mu e(n)x(n) \quad (7)$$

Where

$$Y(n) = P^T(n)x(n) \quad (8)$$

$$E(n) = d(n) - y(n) \quad (9)$$

The above two equations are required output of least mean square algorithm where y (n) is the filter output and e (n) is the error. Figure below shows the block diagram of adaptive filter,

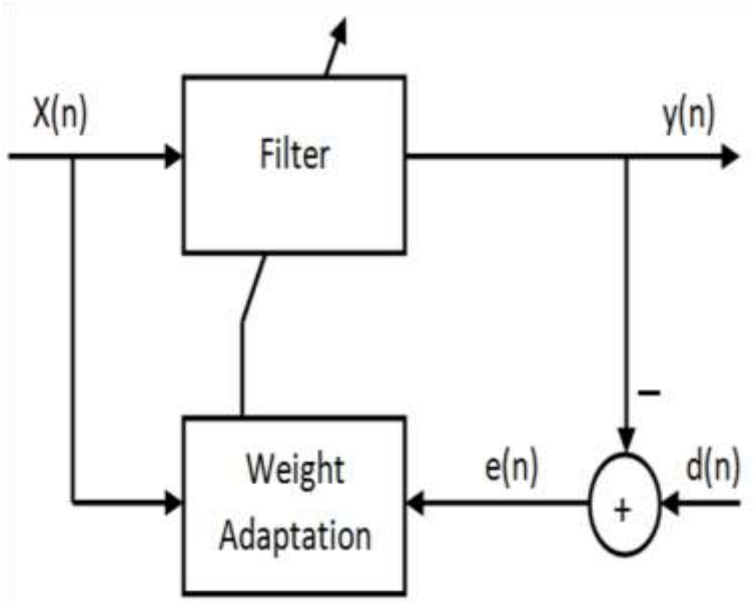


***Fig. 2 A Typical Adaptive filter block diagram [1]***

If the values of d (n) and y (n) will become equal we will get zero error. Filter could be used in combination of different other applications discussed above but our main focus is on noise cancellation system. There are number of parameters related to least mean square adaptive filter, which could differently play an important role in order to reduce the error. In the various applications are also there, which are already discussed above can also be analyzed using least mean square (LMS) filter. In this paper noise cancellation is achieved by variation in least mean square algorithm, i.e. step size. The least mean square adaptive filter also deals with the problem of Adaptive Noise Cancellation (ANC) method for signal corrupted with an additive white Gaussian noise.

## RECURSIVE LEAST SQUARES (RLS) ALGORITHMS

The standard Recursive least squares (RLS) algorithm performs the following operations to update the coefficients of an adaptive filter. The calculates the error signal e (n) by using the following equation:

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

Updates the filter coefficients in using the following equation:

$$\vec{w}(n+1) = \vec{w}(n) + e(n).\vec{K(n)} \quad (10)$$

Where $\vec{w}(n)$ is the filter coefficients vector and $\vec{K}(n)$ is the gain vector $\vec{K}(n)$ Is defined by the following equation:

$$\vec{K}(n) = \frac{P(n).\vec{u(n)}}{\lambda + \vec{u^T(n)P(n)}.\vec{u(n)}} \quad (11)$$

Where λ is the forgetting factor and P (n) is the inverse correlation matrix of the input signal. P (n) has the following initial value P (0):

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

Where δ is the regularization factor. The standard RLS algorithm uses the following equation to update this inverse correlation matrix.

$$p(n+1) = \square^{-1} p(n) - \square^{-1} \underset{K}{\rightarrow}(n) \overrightarrow{u^T(n)p(n)} \qquad (12)$$

RLS algorithms calculate J (n) by using the following equation

$$\mathbf{J(n)} = \frac{1}{N} \sum_{i=0}^{N\ 1} \square^i e^2(\mathbf{n} - \mathbf{i}) \qquad (13)$$

Where N is the filter length and λ is the forgetting factor. In this algorithm calculates not only the instantaneous value $e^2$ (n) but also the past values, such as $e^2$ (n–1), $e^2$ (n–2)... $e^2$ (n–N+1). The value range of the forgetting factor is (0, 1]. The forgetting factor is less than 1, this factor specifies that this algorithm places a larger weight on the current value and a smaller weight on the past values. In resulting E [$e^2$ (n)] of the Recursive least squares (RLS) algorithms are more accurate than that of the least mean square (LMS) algorithms.

## WAVELET TRANSFORM

In this paper, preprocessing is carried out using wavelet transform. When compared to other transforms like Fourier transform we prefer wavelet transform due to the following reasons. Fourier transform represents the signal as the sum of sine and cosine functions which have infinite duration in time. Whereas in wavelet transform the signals are represented as the sum of basic functions which are localized in time leads to more compact representation and also provides better insight into the properties of the signal. In wavelet analysis, signals are represented as a set of basis functions which are derived from a single prototype function called mother wavelet. Basis functions are formed by translating and dilating the mother wavelet.

$$f(t) = \sum_{k=-\infty}^{\infty} \sum_{l=-\infty}^{\infty} d(k, 1) 2^{-\frac{k}{2}} \psi(2^{-k} t - 1) \qquad (14)$$

Where d(k,l) represents the discrete wavelet transform of f(t) and k, l are referred to as integers.
While dealing with Wavelet analysis, two types of transforms can be used.

- Continuous Wavelet transforms.
- Discrete Wavelet transforms.

***Continuous wavelet transforms (CWT)***

This transform works when we use a continuous wavelet function to find the detailed coefficient of a continuous signal. We define a mother wavelet function $\psi$ (t)$\in$ $L^2$(R), which is limited in time domain. That is,$\psi$ (t) has values in a certain range and zeros elsewhere. Another property of mother wavelet is zero-mean. The other property is that the mother wavelet is normalized. Mathematically, they are

$$\int_{-\infty}^{\infty} \psi(t) dt = 0 \qquad (15)$$

$$||\psi(t)||2 = \int_{-\infty}^{\infty} \psi(t) \psi^*(t) dt = 1 \qquad (16)$$

***Discrete wavelet transforms (DWT)***

Discrete Wavelet Transform", transforms discrete signalfrom time domain into time-frequency domain. Thetransformation product is set of coefficients organized inthe way that enables not only spectrum analyses of thesignal, but also spectral behavior of the signal in time.This is achieved by decomposing signal, breaking it intotwo components, each caring information about sourcesignal. Filters from the filter bank used for decompositioncome in pairs: low pass and high pass. The filtering issucceeded by down sampling (obtained filtering result is"re-sampled" so that every second coefficient is kept).Low pass filtered signal contains information about slowchanging component of the signal, looking very similar tothe original signal, only two times shorter in term

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ofnumber of samples. High pass filtered signal containsinformation about fast changing component of the signal.In most cases high pass component is not so rich with dataoffering good property for compression [6].

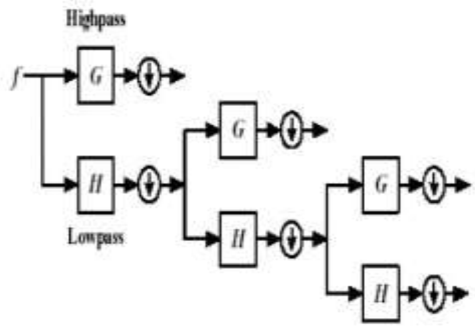


***Fig.3 2-D for discrete wavelet transforms [9]***

## RESULTS AND DISCUSSION

The performance of adaptive noise cancellation (ANC) we compare the LMS, MSE and RLS algorithms as noise canceller. The algorithms are implemented according to the different steps. And its simulation and results are as follows

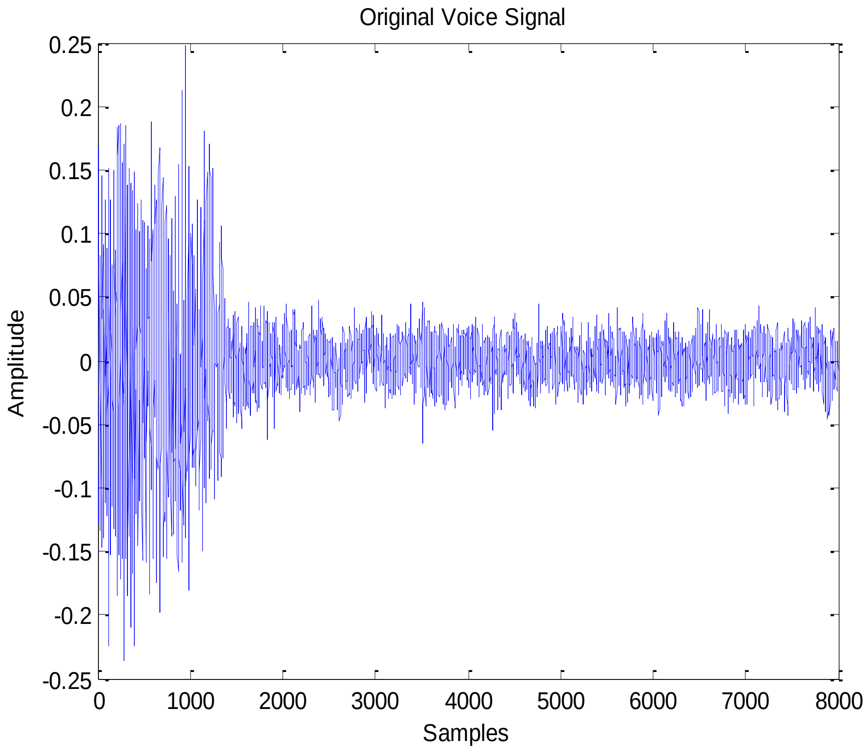


***Fig.4 Original voice signal to amplitude Vs Samples***

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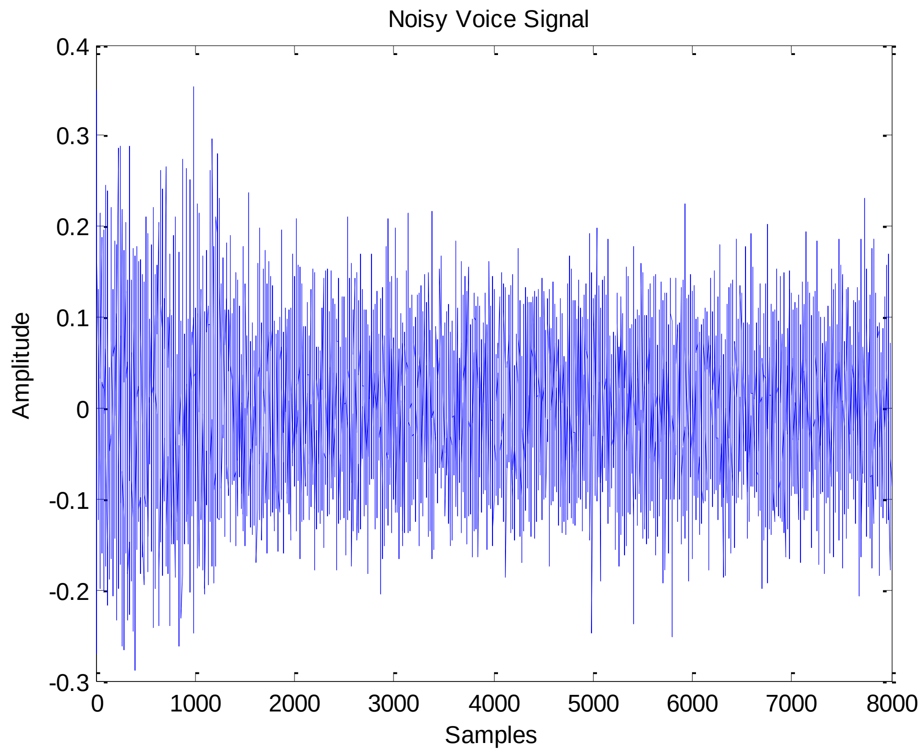


***Fig.5 Performance of amplitude Vs sample***

***Table 1. Performance Parameter Without Wavelet***

| S.No. | Parameter | Value | De-noised Value |
|---|---|---|---|
| 1 | SNR | 5.000000 | 5.000000 |
| 2 | RMSE | 0.063084 | 0.046047 |
| 3 | MAE | 0.050170 | 0.036489 |

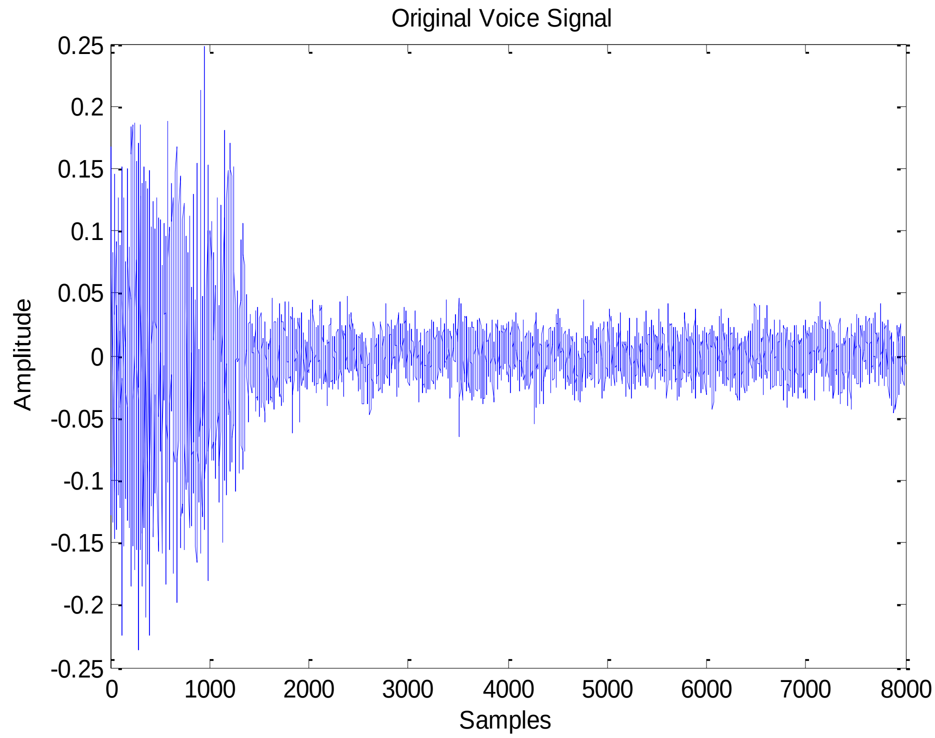


***Fig.6 Original voice signal to amplitude Vs Samples***

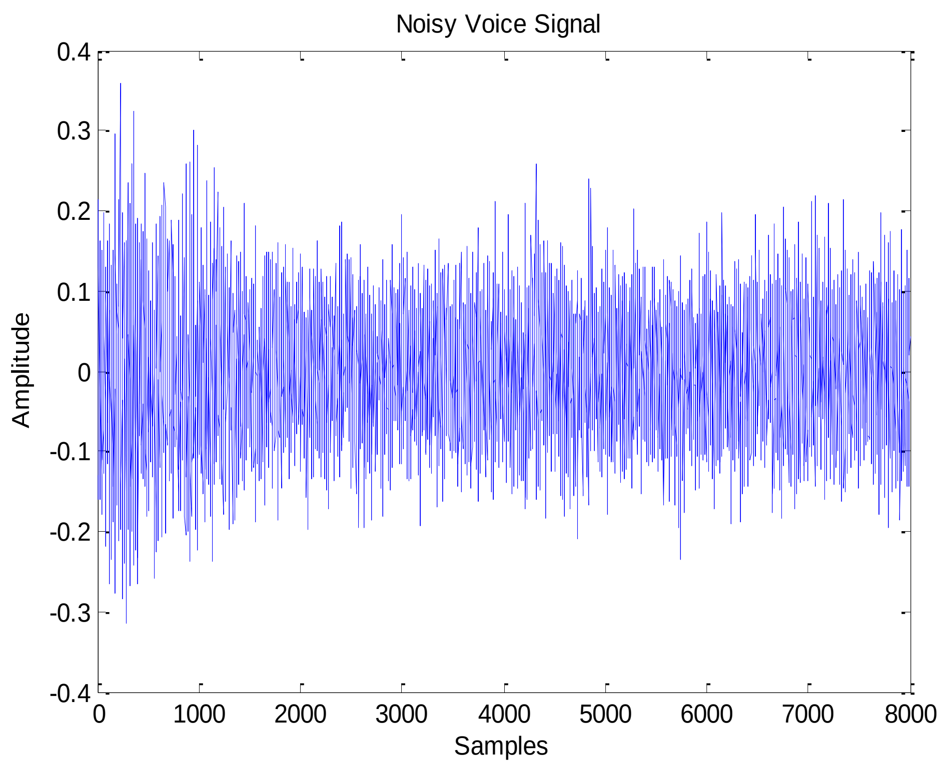


***Fig.7 Performance of amplitude Vs sample***

*Table 2. Performance Parameter With Haar Wavelet Transform*

| S.No. | Parameter | Value | De-noised Value |
|---|---|---|---|
| 1 | SNR | 5.000000 | 5.000000 |
| 2 | RMSE | 0.019813 | 0.017597 |
| 3 | MAE | 0.015873 | 0.013238 |

## CONCLUSION

The performance of adaptive noise cancellation technique with different algorithm. A comparison of LMS and RLS adaptive algorithms with Wavelet Transform for the case of denoising a speech signal. Form the analysis of the paper, we can conclude that for real time denoising operation of voice signal DWT is preferable but, for non-real time case RLS is preferable for precise denoising. Wavelet transforms are suitable for denoising of a signal but in case of recovery of fading signals of wireless communication, adaptive algorithms are the best.

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