DIGITAL COMMUNICATION
PROJECT REPORT

PROJECT NAME:NOISE CANCELLATION USING LMS AND RLS ALGORITHM

SUBMITTED TO :

Prof.Rajesh Birok,
ECE Dept,
DTU

SUBMITTED BY:
Group Number:23
R.Prashant(2k18/EC/124)
RaghavGoel(2k18/EC/126)

Abstract

The concept of noise cancellation has recently gained much attention and has been identified as a vital method to eliminate noise contained in useful signals . The application of this technique can be found in various industrial and communication appliances, such as machineries, hands-free phones and transformers . Additionally, noise cancellation has also been implemented in the field of image processing, biomedical signal, speech enhancement and echo cancellation . As the noise from the surrounding environment severely reduces the quality of speech and audio signals it is quite necessary to suppress noise and enhance speech and audio signal quality, hence the acoustics applications of noise cancellation has become the thrust area of research. The basic concept of Adaptive Noise Canceller (ANC) which removes or suppresses noise from a signal using adaptive filters was first introduced by Widrow . Due to long impulse responses, the computational requirements of adaptive filters are very high especially during implementation on digital signal processors. Where as in case of non-stationary environments and colored background noise convergence becomes very slow if the adaptive filter receives a signal with high spectral dynamic range. To overcome this problem numerous approaches have been proposed in the last few decades. For example, the Kalman filter and the Wiener filter, Recursive-Least-Square (RLS) algorithm, were proposed to achieve the optimum performance of adaptive filters . Amongst these the Least Mean Square (LMS) algorithm is most frequently used because of its simplicity and robustness. Though, the LMS lacks from substantial performance degradation with colored interference signals . Other algorithms, such as the Affine Projection algorithm (APA), became alternative approaches but its computational complexity increases with the projection order, restricting its use in acoustical environments . Noise from the surroundings automatically gets added to the signal in the process of transmission of information from the source to receiver side. The usage of adaptive filters is one of the most popular proposed solutions to reduce the signal corruption caused by predictable and unpredictable noise. Adaptive filters have been used in a broad range of application for nearly five decades.

The objective is to filter the input signal, x(n), with an adaptive filter in such a manner that it matches the desired signal, d(n). In order to generate an error signal the desired signal, d(n), is subtracted from the filtered signal, y(n). An adaptive algorithm is driven by the error signal which generates the filter coefficients in a manner that minimizes the error signal. Unlike from the fixed filter design, here

the filter coefficients are tunable, are adjusted in dependency of the environment that the filter is operated in, and can therefore track any potential changes in this environment. Using this concept, adaptive filters can be tailored to the environment set by these signals. However, if the environment changes filter through a new set of factors, adjusts for new features . The adaptive filter constitutes a vital part of the statistical signal processing. The application of an adaptive filter offers a smart solution to the problem wherever there is a need to process signals that result from operation in an environment of unknown statistics, as it typically provides a significant enhancement in performance over the use of a fixed filter designed by conventional methods . The aim of this paper is to review the existing noise cancellation techniques for enhancing speech and audio signal quality and to provide the understanding of suitability of various developed models. Prior to this, a brief review of the adaptive noise cancellation methods and its application is presented in the next section. Finally, a The concept of noise cancellation has recently gained much attention and has been identified as an important method to eliminate noise contained in useful signals . The application of this system are often found in various industrial and communication appliances, like machineries, hands-free phones and transformers . Additionally, noise cancellation has also been implemented within the field of image processing, biomedical signal, speech enhancement and echo cancellation . As the noise from the encompassing environment severely reduces the standard of speech and audio signals it's quite necessary to suppress noise and enhance speech and audio signal quality, hence the acoustics applications of noise cancellation has become the thrust area of research. The basic concept of Adaptive Noise Canceller (ANC) which removes or suppresses noise from a sign using adaptive filters was first introduced by Widrow . Due to long impulse responses, the computational requirements of adaptive filters are very high especially during implementation on digital signal processors. Where as just in case of non-stationary environments and colored ground noise convergence becomes very slow if the adaptive filter receives a sign with high spectral dynamic range. To overcome this problem numerous approaches have been proposed in the last few decades. For example, the Kalman filter and therefore the Wiener filter, Recursive-Least-Square (RLS) algorithm, were proposed to realize the optimum performance of adaptive filters . Amongst these the Least Mean Square (LMS) algorithm is most frequently used because of its simplicity and robustness. Though, the LMS lacks from substantial performance degradation with colored interference signals . Other algorithms, such as the Affine Projection algorithm (APA), became alternative approaches but its computational complexity increases with the projection order, restricting its use in acoustical environments . Noise from the environment automatically gets added to the signal within the process of transmission of data from the source to receiver side. The usage of adaptive filters is one among the foremost popular proposed solutions to scale back the signal corruption caused by predictable and unpredictable noise.

Adaptive filters have been used in a broad range of application for nearly five decades.

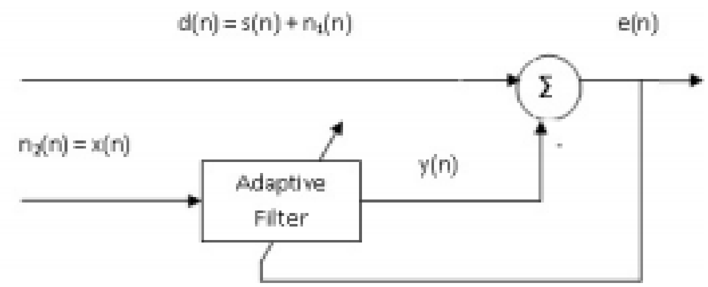
The objective is to filter the input , x(n), with an adaptive filter in such a fashion that it matches the specified signal, d(n). In order to get a mistake signal the specified signal, d(n), is subtracted from the filtered signal, y(n). An adaptive algorithm is driven by the error signal which generates the filter coefficients in a manner that minimizes the error signal. Unlike from the fixed filter design, here the filter coefficients are tunable, are adjusted in dependency of the environment that the filter is operated in, and may therefore track any potential changes in this environment. Using this idea , adaptive filters are often tailored to the environment set by these signals. However, if the environment changes filter through a replacement set of things , adjusts for brand spanking new features . The adaptive filter constitutes an important a part of the statistical signal processing. The application of an adaptive filter offers a sensible solution to the matter wherever there's a requirement to process signals that result from operation in an environment of unknown statistics, because it typically provides a big enhancement in performance over the utilization of a hard and fast filter designed by conventional methods . The aim of this paper is to review the prevailing noise cancellation techniques for enhancing speech and audio signal quality and to supply the understanding of suitability of varied developed models. Prior to this, a quick review of the adaptive noise cancellation methods and its application is presented within the next section. Finally, a perception on upcoming research is usually recommended for further consideration.perception on upcoming research is suggested for further consideration.

INTRODUCTION:

ADAPTIVE NOISE CANCELLATION

Acoustic noise cancellation is indispensable from the health point of view as extensive exposures to high level of noise may cause serious health hazards to human being. The conventional noise cancellation method uses a reference input signal (correlated noise signal) which is passed through the adaptive filter to make it equal to the noise that is added to original information bearing signal. Subsequently this filtered signal is subtracted from noise corrupted information signal. This makes the corrupted signal a noise free signal. The fundamental concept of noise cancellation is to produce a signal that is equal to a disturbance signal in amplitude and frequency but has opposite phase. These two signals results in the cancellation of noise signal. The original Adaptive noise cancellation (ANC) uses two sensors to receive the noise signal and target signal separately.

The relationship between the noise reference x(n) and the component of this noise that is contained in the measured signal d(n) may be determined by Adaptive noise cancellation



Adaptive Filters

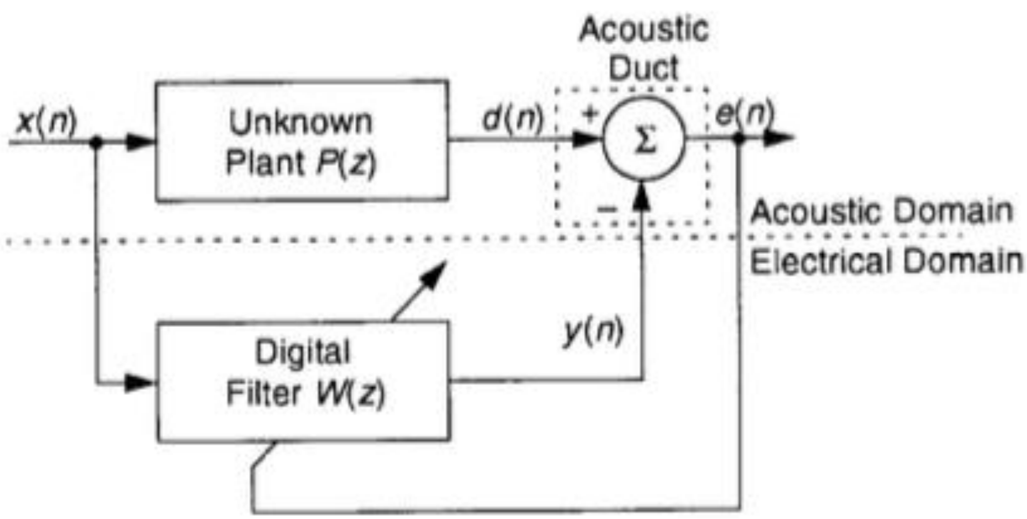
An adaptive filter **may be a** system with a linear filter which consists of transfer function restrained by variable parameters and **a way to regulate** those parameters **consistent with** an optimization algorithm. Adaptive linear filters are linear dynamical system with variable or adaptive structure and parameters and have the property to modify the values of their parameters, i.e. their transfer function, during the processing of the input signal, in order to generate signal at the output which is without undesired components, noise, and degradation and also interference signals.

Figure shows the basic concept of an adaptive filter whose primary objective is to filter the input signal, x(n), with an adaptive filter in such a manner that it matches the desired signal, d(n). The desired signal, d(n), is subtracted from the filtered signal, y(n), to produce an error signal which in turn drives an adaptive algorithm that generates the filter coefficients in a manner that minimizes the error signal. The adaptation adjusts the characteristics of the filter through an interaction with the environment in order to reach the desired values. Contrary to **the traditional** filter design techniques, adaptive filters **don't** have constant filter coefficients and no priori information **is understood** , such a filters with adjustable parameters are called an adaptive filter. Adaptive filter adjust their coefficients to minimize an error signal and may be termed as finite impulse response (FIR) , infinite impulse response (IIR) , lattice and transform domain filter. Generally adaptive digital filters **contains** two separate units: the digital filter, with a structure determined **to realize** desired processing (which **is understood** with an accuracy to the unknown parameter vector) **and therefore the** adaptive algorithm for the update of filter parameters, with an aim **to ensure** fastest possible convergence to the optimum parameters from **the**

purpose of view of the adopted criterion. Majority of adaptive algorithms signify modifications of **the quality** iterative procedures for **the answer** of **the matter** of minimization of criterion function in real time. The most common form of adaptive filters are the transversal filter using least mean square (LMS) algorithm and recursive least square (RLS) algorithm**Adaptive Algorithms**

Adaptive algorithms have been extensively studied in the past few decades and the most popular adaptive algorithms are the least mean square (LMS) algorithm and the recursive least square (RLS) algorithm. Attaining the best performance of an adaptive filter requires usage of the best adaptive algorithm with low computational complexity and a fast convergence rate.

Least-Mean-Square Algorithm (LMS)



A very straightforward approach in noise cancelling is that the use of LMS algorithm which was developed by Windrow and Hoff . This algorithm uses a gradient descent to estimate a time varying signal. The gradient descent method finds a minimum, if it exists, by taking steps within the direction negative of the gradient and it does so by adjusting the filter coefficients so as to attenuate the error. The gradient is that the del-operator and is applied to seek out the divergence of a function, which is that the error with reference to the nth coefficient during this case. The LMS algorithm has been accepted by several researchers for hardware implementation due to its simple structure. so as to implement it, modifications need to be made to the first LMS algorithm because the recursive loop in its filter update formula prevents it from being pipelined.
The following equation shows the detail of LMS algorithm,

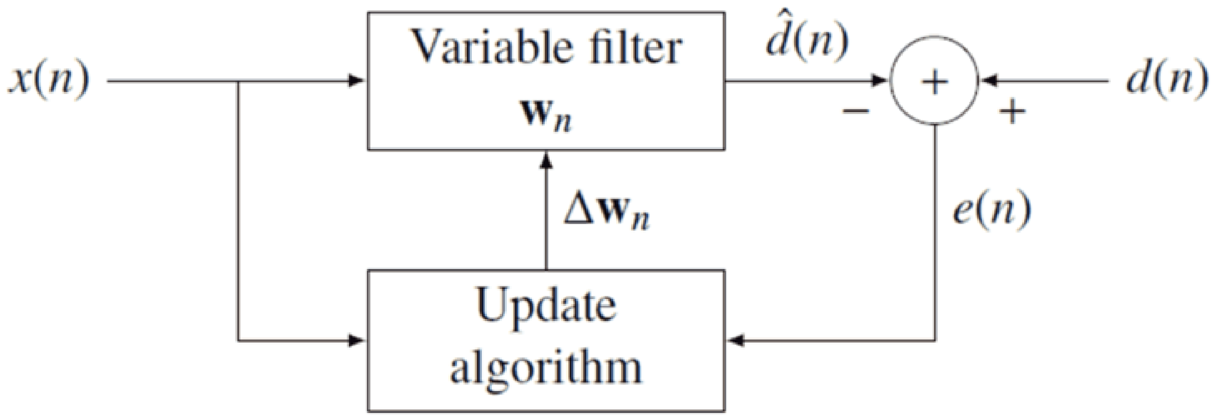
$$e(n) = d(n) - \boldsymbol{w}^T(n)\boldsymbol{x}(n)$$

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) - \mu\boldsymbol{x}(n)e(n)$$

Where Myu is the learning rate.

Recurrent Least Square(RLS) ALGORITHM

RLS algorithm is another potential alternative to overcome slow convergence in colored environments , which uses the least squares method to develop a recursive algorithm for the adaptive transversal filter. The RLS recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. RLS tracks the time variation of the method to the optimal filter coefficient with relatively in no time convergence speed; though it's increased computational complexity and stability problems as compared to LMS-based algorithms . The RLS algorithm has established itself because the "ultimate" adaptive filtering algorithm within the sense that it's the adaptive filter exhibiting the simplest convergence behavior. Unfortunately, practical applications of the algorithms are often associated with high computational complexity and poor numerical properties.
RLS uses the least squares method to develop a recursive algorithm for the adaptive transversal filter. The RLS recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals.



Suppose that a signal d(n) is transmitted over an echoey, noisy channel that causes it to be received as

$$x(n) = \sum_{k=0}^{q} b_n(k) d(n-k) + v(n)$$

$$d(n) \approx \sum_{k=0}^{p} w(k) x(n-k) = \mathbf{w}^T \mathbf{x}_n$$

$$\hat{d}(n) = \sum_{k=0}^{p} w_n(k) x(n-k) = \mathbf{w}_n^T \mathbf{x}_n$$

where v(n) represents additive noise. The intent of the RLS filter is to recover the desired signal d(n) by use of a p+1-tap FIR filter, w.

EXPERIMENTATION:

The following algorithm was coded and simulated in SCILAB. The code has been attached forthwith:

LMS Algorithm:

```
// set len of signal mp
mp=500
// set time vector n
n=(1:1:mp ) ;
// set len of filter hmm and hmp
nfilter =20 ;
// set noise vector
R= .2 // variance Gaussian
av =0 // mean Gaussian
sd =sqrt(R) // std Gaussian
v = grand(1,mp,'nor',av,sd) ;// generate white gaussian
hmm = zeros(1,nfilter) ;
hmp = zeros(1,nfilter) ;

// est vector is dest
dest=zeros(1, mp);

// create input signal x in theta
dtheta =2*%pi/mp ;
x=dtheta*n ;
// create desired vector d as sin(x)
d =sin(x) ;
figure(0) ;
plot (x ,d) ;
// set xmm init vector
i=1 ;
x1=x; // we are using x as d + noise so we store x as x1
// now create input vector x with noise

x= d+v ;
deltn =.01 // step size
// begin computation for i statement
for i=1: mp-nfilter ;
i ;
```

```
xmm = x(1:1, i: (nfilter+i -1) ) ;
in= i+1 ;

// set next input vector
//xmp = x(1:1, in: (nfilter+in -1) ) ;

// compute thedhat value from xmm ' *hmm
// dhat is set to zero vector first

dhat = xmm*hmm' ;
// update est vector
dest(i) = dhat ;
// compute last error
elast= d(i) - dhat ;
elast ;
// update hmm vector
hmp= hmm + xmm*elast*deltn ;
hmm =hmp ;
// next step
end ;
d ;
dest ;
figure(1);
subplot(221) ;
title (' true signal ');
plot(x1, d);
subplot(222) ;
title (' signal + noise ');
plot(x1, x);
subplot(224)
title (' filtered signal ');
plot(x1, dest);
subplot(223)
title (' noise ');
plot(x1, v);
```

RLS Algorithm:

```
// set len of signal mp
```

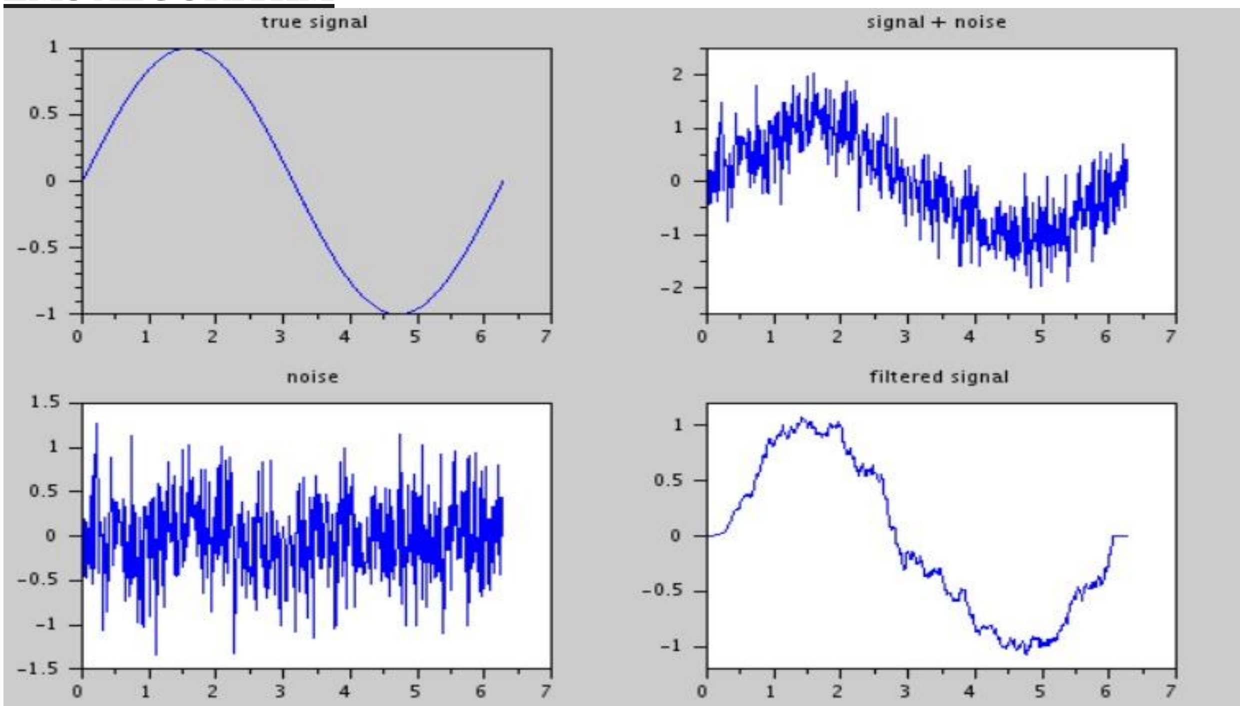
```
mp=500
// set time vector n
n=(1:1:mp ) ;
// set len of filter hmm and hmp
nfilter =20 ;
// set noise vector
R= .2 // variance Gaussian
av =0 // mean Gaussian
sd =sqrt(R) // std Gaussian
v = grand(1,mp,'nor',av,sd) ;// generate white gaussian
hmm = zeros(1,nfilter) ;
hmp = zeros(1,nfilter) ;
// set weight w bet 0 and 1
w=.9
// est vector is dest
dest=zeros(1, mp);

// set pmm and pmp matrices
pmm=eye(nfilter,nfilter) ;
pmp=eye(nfilter,nfilter) ;

// create input signal x in theta
dtheta =2*%pi/mp ;
x=dtheta*n ;
// create desired vector d as sin(x)
d =sin(x) ;
figure(0) ;
plot (x ,d) ;
// set xmm init vector
i=1 ;
x1=x; // we are using x as d + noise so we store x as x1
// now create input vector x with noise

x= d+v ;

emm = zeros (1,nfilter) ;
emp= zeros(1,nfilter) ;
// begin computation for i statement
for i=1: mp-nfilter ;
i ;
```

```
xmm = x(1:1, i: (nfilter+i -1) ) ;
in= i+1 ;

// set next input vector
xmp = x(1:1, in: (nfilter+in -1) ) ;

// compute thedhat value from xmm ' *hmm
// dhat is set to zero vector first

dhat = xmm*hmm' ;
// udate est vector
dest(i) = dhat ;
// compute last error
elast= d(i) - dhat ;
elast ;
// compute error vector emm
emm = d(1: 1 , i: (nfilter+i -1) ) -dest(1:1 , i: (nfilter+i -1) ) ;

// compute kalman gain
kmg = w + xmm*pmm*xmm' ;

// compute kalman gain vector
kalgv =pmm*xmm' /kmg ;

// update pmm
pm = kalgv*xmm*pmm ;
pmp= pmm-pm ;
pmm = pmp*(1/w) ;

// update hmm vector
hmp= hmm + kalgv'*elast ;
hmm =hmp ;
// next step
end ;
d ;
dest ;
figure(1);
subplot(221) ;
title (' true signal ');
plot(x1, d);
```

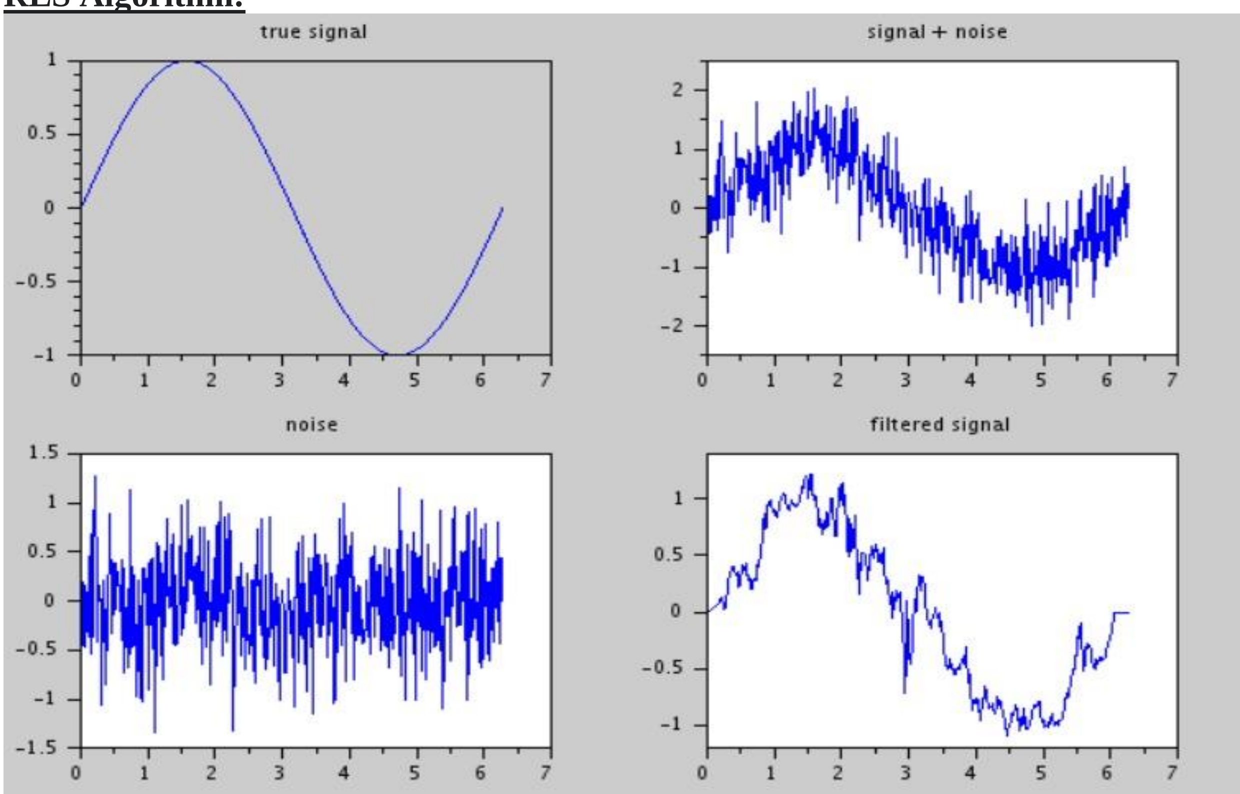
```
subplot(222) ;
title (' signal + noise ');
plot(x1, x);
subplot(224)
title (' filtered signal ');
plot(x1, dest);
subplot(223)
title (' noise ');
plot(x1, v);
```

Result:

LMS ALGORITHM



RLS Algorithm:



CONCLUSION:

A comprehensive review has been carried out to identify the existing literature related to adaptive filtering in noise reduction using LMS adaptive algorithms in particular. LMS is preferred over RLS algorithms for various noise cancellation purposes as RLS has increased computational complexity and stability problems as compared to LMS-based algorithms which are robust and reliable.

The LMS algorithm is relatively simple to implement and is powerful enough to evaluate the practical benefits that may result from the application of adaptivity to the problem at hand. Moreover, it provides a practical frame of reference for assessing any further improvement that may be attained through the use of more sophisticated adaptive filtering algorithms.