“ NOISE REDUCTION IN AUDIO SIGNAL USING LMS FILTER USING IN MATLAB ”

A Project Report of

Course Wireless Communication (18EC4111)

Submitted in the partial fulfilment of the requirements for the award of the degree of

Bachelor of Technology

in

Department of Electronics and Communication Engineering

By

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Under the Supervision of

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KONERU LAKSHMAIAH EDUCATION FOUNDATION

Department of Electronics and Communication Engineering

Declaration

The Wireless Communication Project Report entitled "**NOISE REDUCTION IN AUDIO SIGNAL USING LMS FILTER USING IN MATLAB** "is a record of bonafide work of **N.SATISH - ID NO :- 180040132** , submitted in partial fulfilment for the award of Bacheloor Technology in **Electronics And Communication Engineering**

I also declare that this report is of our own effort and it has not been submitted to any other university for the award of any degree.

N.SATISH - ID NO :- 180040132

KONERU LAKSHMAIAH EDUCATION FOUNDATION

Department of Electronics and Communication Engineering

Certificate

The Wireless Communication Project Report entitled “ **NOISE REDUCTION IN AUDIO SIGNAL USING LMS FILTER USING IN MATLAB** ” submitted by **N.SATISH - ID NO :- 180040132** in partial fulfilment for the award of Bachelor of Technology in **Electronics And Communication Engineering.**

Nageswara Rao sir

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ABSTRACT

In an audio speech signal, acoustic noise is a common problem while the speech is processed. Here, we are going to create color noise and add with an audio signal, after that a model are introduced to eliminate that noise.

This paper elaborates a new approach for noise cancellation in speech enhancement using an Adaptive LMS (Least Mean Square) filter and with the help of MATLAB Simulink we get the correct speech signal.

This filter is used to remove the acoustic noise due to its simplicity in computation & robust behavior when implemented in finite-precision hardware.
It provides better communication by suppressing the acoustic noise to a larger extent, since it provides a better balance between complexity & convergence speed.

In spite of various methods, the results obtained in this way of noise cancellation are optimistic.

The basic way of communication in humans to convey information or message is speech. This communication helps the humans to full fill their basic needs with a bandwidth of 4 KHz.

Noise is an unwanted signal is the major drawback affects the speech signal. There are various sources of noise which affects the speech such as electrical, acoustic, vibration or any other noise components.

Among that, acoustic noise plays a major role since it becomes more noticeable as the number of commercial equipment increases.

Acoustics is a branch related to physics. This noise introduces a masking problem to reduce the distinct nature of speech which affects the communication .

During the elimination of this acoustic noise, the change of signal characteristics becomes more common.

There are number of adaptive algorithms to remove the acoustic noise but we use Least Mean Square (LMS) filter to overcome this problem.
The reason behind this is when compared with traditional filter .

A noise is added and using adaptive LMS filter we suppress the acoustic noise leaving the speech signal unchanged.
This method improves the efficiency to a greater extent.

TABLE OF CONTENTS

1> OBJECTIVE OF THE PROJECT :-

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2> SOFTWARE TOOLS USED AND THEIR DESCRIPTION :-

- MATLAB (matrix laboratory)

Engineers and scientists **need** a programming language that lets them express matrix and array mathematics directly.

Linear algebra in **MATLAB** is intuitive and concise. The same is true for data analytics, signal and image processing, control design, and other applications.

Transmitted signal deals with signal processing . so , to study signal we requires signal processing tool box .

Signal Processing Toolbox™ provides functions and apps to analyze, preprocess, and extract features from uniformly and nonuniformly sampled **signals**. ... With the Filter Designer app you can design and analyze digital filters by choosing from a variety of algorithms and responses. Both apps generate **MATLAB**® code.

MATLAB allows you to test algorithms immediately without recompilation. You can type something at the command line or execute a section in the editor and immediately see the results, greatly facilitating algorithm development.

The ability to read in a wide variety of both common and domain-specific formats of a signal .
MATLAB and **signal processing** products help you analyze **signals** from a range of data sources.

You can acquire, measure, transform, filter, and visualize **signals** without being an expert in **signal processing** theory.
You can apply **signal processing** tools to: Preprocess and filter **signals** prior to **analysis**.

3> INTRODUCTION/THEORY :-

In an audio speech signal, acoustic noise is a common problem while the speech is processed. Here, we are going to create color noise and add with an audio signal, after that a model are introduced to eliminate that noise.

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The reason behind this is when compared with traditional filter .

A noise is added and using adaptive LMS filter we suppress the acoustic noise leaving the speech signal unchanged.
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4> ABOUT FILTER :-

LMS filters are a class of adaptive filters that are able to "learn" an unknown transfer functions.

LMS filters use a gradient descent method in which the filter coefficients are updated based on the instantaneous error signal.

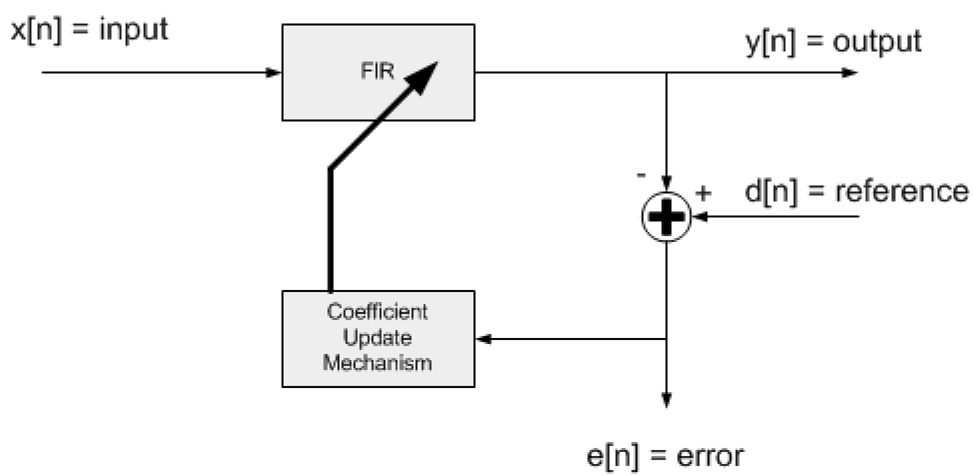
Adaptive filters are often used in communication systems, equalizers, and noise removal

An LMS filter consists of two components as shown below. The first component is a standard transversal or FIR filter. The second component is a coefficient update mechanism.

The LMS filter has two input signals. The "input" feeds the FIR filter while the "reference input" corresponds to the desired output of the FIR filter.

That is, the FIR filter coefficients are updated so that the output of the FIR filter matches the reference input. The filter coefficient update mechanism is based on the difference between the FIR filter output and the reference input. This "error signal" tends towards zero as the filter adapts.

The LMS processing functions accept the input and reference input signals and generate the filter output and error signal.



Internal structure of the Least Mean Square filter

The functions operate on blocks of data and each call to the function processes blockSize samples through the filter. pSrc points to input signal, pRef points to reference signal, pOut points to output signal and pErr points to error signal. All arrays contain blockSize values.

The functions operate on a block-by-block basis. Internally, the filter coefficients b[n] are updated on a sample-by-sample basis. The convergence of the LMS filter is slower compared to the normalized LMS algorithm.

The equation below is LMS algorithm for updating the tap weights of the adaptive filter for each iteration.

$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu e(n)\mathbf{x}^*(n)$

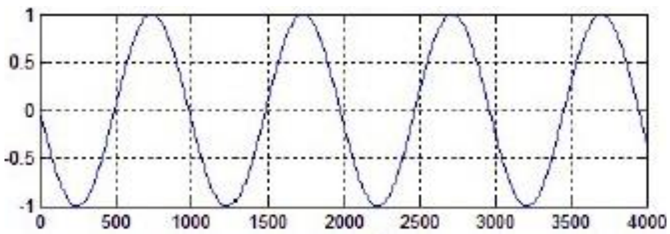
Where,

• **x**(n) : input vector of time delayed input values.

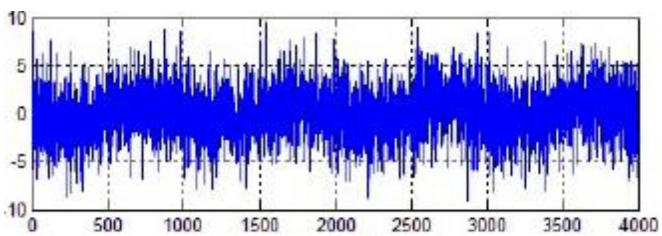
• **w**(n) : weight vector at time n .

μ is a step-size parameter and it controls the immediate change of the updating factor. It shows a great impact on the performance of the LMS algorithm in order to change its value. If the value of μ is so small then the adaptive filter takes long time to converge on the optimal solution and in case of large value the adaptive filter will be diverge and become unstable.

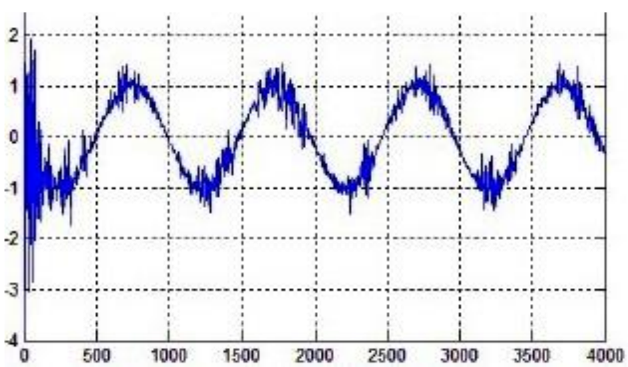
INPUT SIGNAL :-



NOISE + INPUT SIGNAL :-

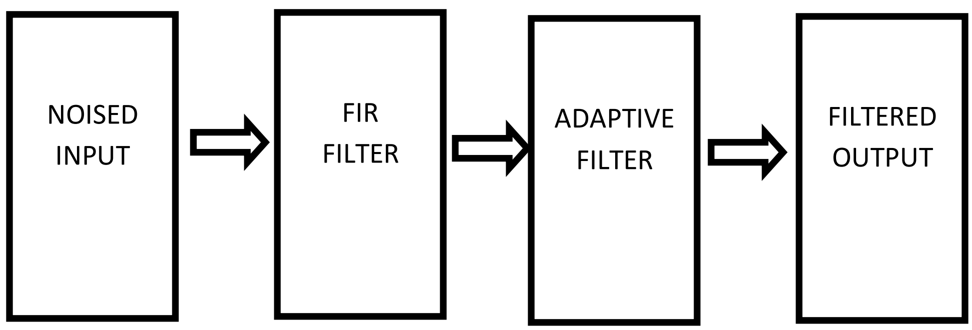


FILTERED OUTPUT :-



5> **BLOCK DIAGRAM :-**

The generated noise is added to the original audio signal which is in .Wav format file. The noise added signal is filtered [2] by fir filter in order to eliminate thewhite noise which is presented in noise signal that is generated from the random noise generator and to get the remaining colored noise from the signal. This noise is further filtered by using an adaptive LMS filter that we designed and created in order to cancel the acoustic noise completely.

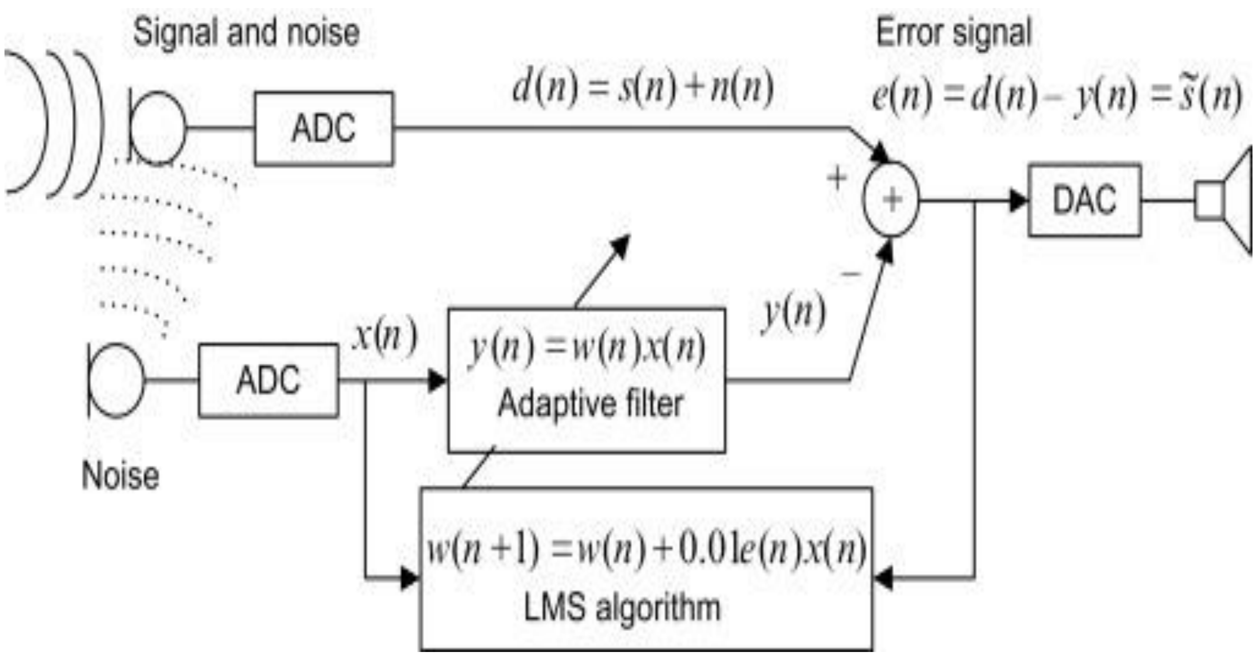


6> **METHODOLOGY :-**

Here, we use a method called Least Mean Square algorithm which is used to suppress the acoustic noise by

using Simulink in MATLAB11a software. In MATLAB 11a Simulink has a Data Acquisition Toolbox helps to cancel the acoustic noise from the original signal.

For suppressing the acoustic noise from the original signal, we are using the Block LMS filter.



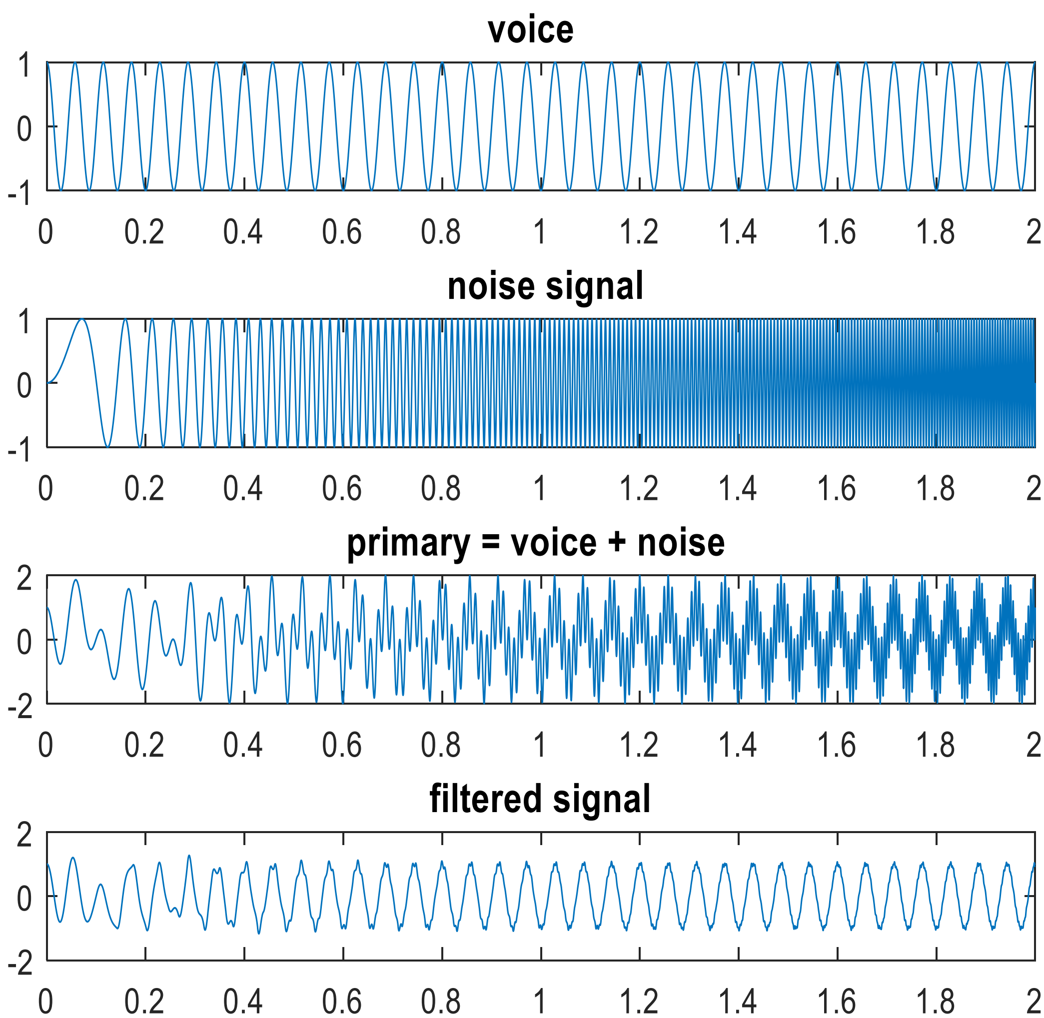
7> MATLAB CODE :-

```
%noise cancellation
order=2;
size=2;                    %time duration of inputs
```

```
fs=8192;                         %digital sampling frequency
t=[0:1/fs:size];
N=fs*size;                  %size of inputs
f1=35/2;                         %frequency of voice
f2=99/2;                         %frequency of noise

voice=cos(2*pi*f1*t);
subplot(4,1,1)
plot(t,voice);
title('voice  ')

noise=sin(2*pi*f2*t.^2);                    %increasy frequency noise
primary=voice+noise;
subplot(4,1,2)
plot(t,noise)
title(' noise signal')
subplot(4,1,3)
plot(t,noise+voice)
title('primary = voice + noise ')

%ref=noise+.25*rand;                             %noisy noise
%subplot(4,1,3)
%plot(t,ref)
%title('reference  (noisy noise)  ');

w=zeros(order,1);
mu=.006;
for i=1:N-order
   buffer = noise(i:i+order-1);                         %current 32 points of reference
   desired(i) = primary(i)-buffer*w;              %dot product reference and coeffs
   w=w+(buffer.*mu*desired(i)/norm(buffer))';%update coeffs
```

```
end

subplot(4,1,4)
plot(t(order+1:N),desired)
title('filtered signal')
```

8> OUTPUT :-



9> INFERENCE AND ANALYSIS :-

we design the acoustic noise cancellation using an adaptive filter based on LMS. The noised audio signal and the filtered signal mean without an acoustic noise signal can hear using through the audio player which is designed at simulation work space.

In our proposed method we created the digital filter module which updates its coefficients in order to remove the noise from the given noised input signal.

so , from our LMS (least mean square) algorithm we almost got the expected and similar output by considering the input signal .

10> RESULT :-

So , after applying LMS filter to the noised input signal we almost got the similar output as same as the input signal .

11> **CONCLUSION :-**

We designed and created a model in order to show the adaptive LMS filter works perfectly on the acoustic noise cancellation process. The results we obtained clearly shows that the ability of the adaptive LMS filters in noise cancellation and achieves the noise suppressed signal. The resulted output is almost similar to the original input audio signal.