Multi Channel Acoustic Echo Cancellation Using ICA

A PROJECT REPORT

Submitted in partial fulfillment of the requirement for the award of the Degree of

BACHELOR OF TECHNOLOGY
IN
ELECTRONICS AND COMMUNICATION ENGINEERING

by

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CERTIFICATE

This is to certify that the Project work titled **"Multi Channel Acoustic Echo Cancellation Using ICA"** that is being submitted by ***Yashaswi Peta (14BEC1120), Sandeep Somu (14BEC1060)*** and ***Nithin Raj Anantha (14BEC1164)*** is in partial fulfillment of the requirements for the award of **Bachelor of Technology in Electronics and Communication Engineering**, is a record of bonafide work done under my guidance. The contents of this Project work, in full or in parts, have neither been taken from any other source nor have been submitted to any other Institute or University for award of any degree or diploma and the same is certified.

DR. K. MOHANAPRASAD
Guide

The thesis is satisfactory / unsatisfactory

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Approved by

DEAN
School of Electronics Engineering

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CHAPTER 1
INTRODUCTION

Echo is a reflection of sound that arrives at the listener with a delay after the direct sound. Acoustic echo arises when sound from a loud speaker is picked up by a microphone in the same room. For example, the sound from the speaker of a mobile handset can bounce off the walls and then get picked up by the handset's microphone. The presence of acoustic echo degrades the quality of the speech signal and this is a major problem in telephony and has to dealt with. An ICA based algorithm is proposed in this project to tackle the issue of undesired acoustic echo.

1.1 Objectives

The following are the objectives of this project:

- To develop an ICA based algorithm to cancel the acoustic echo in a multichannel setup.
- To extract the source signals. (Near end speakers)
- To calculate the Echo Return Loss Enhancement.
- To maintain an ERLE of 35dB to 40dB.

1.2 Background and Literature Survey

Acoustic Echo Cancellation (AEC) plays a vital role in the hands free communication environment. Generally, the echo can be suppressed by summing up the output of an adaptive filter (replica of the echoed signal) and the microphone signal. The coefficients of this adaptive filter are continuously updated by using various adaptive algorithms such as Least Mean Square (LMS), Normalized Least Mean Square (NLMS), Recursive Least Square (RLS) in order to keep the mean square error at the minimum. The adaptive filters fail to update the coefficients correctly in the presence of a double talk scenario, thus allowing some part of the echo to pass through. This made us develop an algorithm suitable to work in double talk situations.

A research paper titled *Wavelet based ICA using maximisation of non-Gaussianity for acoustic echo cancellation during double talk situation* published in the year 2015 which dealt in a single channel setup was our primary reference. This work made use of the measure of non-Gaussianity for acoustic echo cancellation. The measure of non-Gaussianity was obtained using kurtosis and other high order statistics like negentropy. The algorithm works even in double talk situations and the ERLE obtained was around 36dB which was very good.

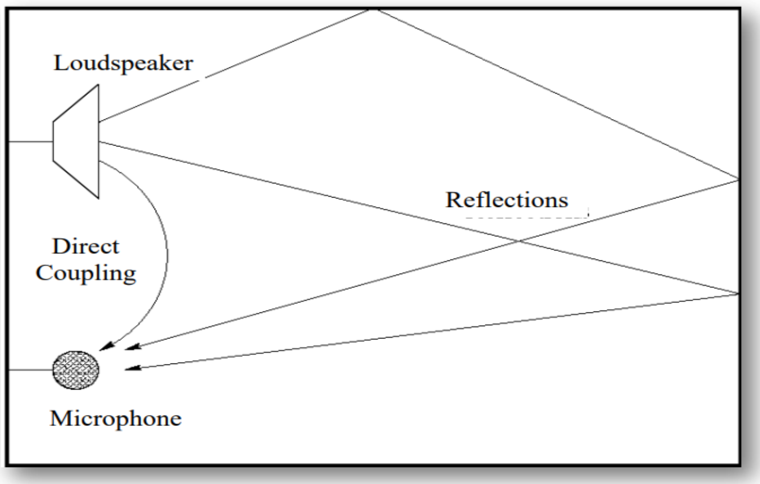
The ability of the algorithm to work in double talk situation with a good ERLE output made us further develop the algorithm for a multichannel environment.

Table 1.1 Literature Survey

Sl.No	**Name of article**	**Name of journal / Conference / Publisher**	**Information included**
1.	Wavelet based ICA using maximisation of non-Gaussianity for acoustic echo cancellation during double talk situation	Applied Acoustics 97 (2015) 37–45	ICA based algorithm for single channel environment.
2.	An Acoustic Front-End for Interactive TV Incorporating Multichannel Acoustic Echo Cancellation and Blind Signal Extraction.	Asilomar Conference 2010	Multichannel setup scenario 2
3.	Insight into a phase modulation technique for signal decorrelation in multi-channel acoustic echo cancellation.	ieeexplore.ieee.org/iel7/7465907/7471614/07471729	Multichannel setup scenario 1
4.	Multichannel acoustic echo cancellation exploiting effective fundamental frequency estimation.	Speech Communication 86 (2017) 97-106	Double talk detection
5.	Multichannel acoustic echo cancellation the wave domain with increased robustness to non-uniqueness	IEEE/ACM Transactions on audio, speech and language processing, VOL 24 March 2016	Discussion on the non-uniqueness problem.
6.	Independent Component Analysis	JOHN WILEY & SONS, INC.	Theoretical information on ICA.

1.3 Need for Acoustic Echo Cancellation

Acoustic Echo Cancellation (AEC) plays a vital role in the hands free communication environment. The presence of acoustic coupling between the far-end speech signal and near end speech signal produces an undesired acoustic echo, which degrades the quality of the speech signal. Acoustic echo cancellation is the answer to the above problem.



1.4 Organization of the Report

The remaining chapters of the project report are described as follows:

- Chapter 2 contains the methodology, fundamentals, kurtosis, ERLE, Restrictions in the project.
- Chapter 3 contains the proposed algorithms and the scenarios to which it was applied.
- Chapter 4 contains the cost analysis.
- Chapter 5 contains the results and discussion.
- Chapter 6 contains the conclusion and future work.

CHAPTER 2

Acoustic Echo Cancellation

This Chapter contains the methodology, fundamentals, kurtosis, ERLE, Restrictions and summary.

2.1 Methodology

An ICA based algorithm is proposed to cancel the acoustic echo and to extract the source signal. MATLAB is used as the primary simulation setup. All the near end and far end speech signals are taken from the TIMIT database and sampled at 8 kHz. Male speech signal is used as the near end signal and female speech signal is used as the far end signal. It is made sure that both the near end and far end speech signals are active during the implementation, so that there is a double talk situation present.

For demonstration purposes, the microphone signal is derived from the random mixing of the far end and near end signals using a random matrix. The result of the simulation is analyzed using a parameter called Echo Return Loss Enhancment (ERLE), which is defined as the ratio of residual echo to that of the original echo.

2.2 ICA Fundamentals

2.2.1 ICA Model Basics

The main idea is to recover the unknown source signal (near end speech) from a group of mixed signals (mic input). The source signals are represented by s(t),i=1;2;...n, which are unknown to us and the mixed group of signals are represented by x(t),i=1;2;...n. Primitive block diagram of ICA model is shown in Fig. 1.

Mathematically the mixtures are represented as:

$$x_i(t) = \sum a_{ij} * s_j(t) \quad i = 1,2,3,\ldots,n \qquad (1)$$

Where a_{ij} is the coefficient of mixing of ith measurement and the jth source. The above equation in vector form is given as:

$$x = A * s$$

Where the unknown mixing matrix $[A_{ij}] = a_{ij}$. $x = [x_1(t),x_2(t),x_3(t),\ldots x_n(t)]^T$ is the linear combination of mixing matrix and $s = [s_1(t),s_2(t),s_3(t),\ldots s_n(t)]^T$ is the source signal which is assumed to be mutually independent. To separate the source signal,the separation matrix $W = A^{-1}$ is used in general, which is given by :

$$y = A^{-1} * x;$$

$$y = W * x;$$

Where W is the inversion matrix of A and cannot be directly determined because matrix A is unknown. It can be still be found by adaptively calculating the W vectors and putting up a cost function determined by the different ICA techniques.

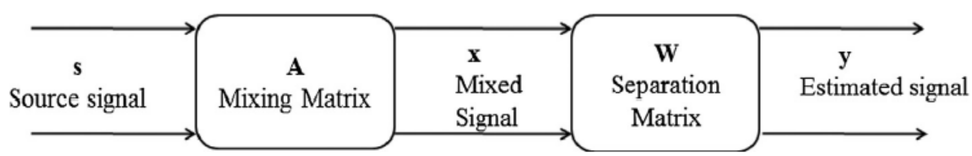


Fig. 1. Basic ICA model.

2.2.2 Pre-Processing in ICA

In order to make the process of ICA simpler and faster, pre-processing is vital. There can be many pre-processing steps involved but there are two most important steps:

1. Mixed signal data, x(t) i.e microphone signal (echo + near end speech signal) is centralized by subtracting the mean of the mixture from original mixture. The mean of the signal becomes zero when centering is done. This also simplifies the complexity of the entire algorithm.
2. Whitening is necessary to make the centered data uncorrelated, because independent signals are uncorrelated. An uncorrelated signal has a unit variance and its identity covariance matrix. i.e

$E[x\bar{x}] = I$

One of the most popular methods for whitening is Eigen Value Decomposition (EVD) which can represented mathematically as:

$V = E*D^{-1/2}*E^{T}$

Where E is the orthogonal matrix of Eigen vectors of $E[x\bar{x}]$.

D = diag{d1,d2,d3,….dn} is the diagonal matrix of its Eigen values. To implement whitening, the linearly transformed vector v is estimated to transform the original signal x into a whitened matrix Z which is given as

$Z = V * X$

The reduction in the number of parameters to be estimated, from n^2 to n(n-1)/2, is only because of whitening.

2.3 Maximization of non-gaussianity of ICA

The classical measure of Maximization of non-gaussianity is acquired using kurtosis.

2.3.1 Kurtosis

The fourth order cumulant of the whitened signal is known as kurtosis. It is expressed as

$kurt(y) = k4 = E\{y^4\} – 3*[E\{y^2\}]^2 – 4*E\{y^3\}*E\{y\} + 12*E\{y^2\}*E\{y^2\} – 6*E\{y^4\}$

Assuming that centered data has zero mean, the Eq is simplified as

$kurt(y) = k4 = E\{y^4\} – 3*[E\{y^2\}]^2$

Furthermore, the variance of a whitened signal is equal to 1, which implies $E\{y^2\} = 1$, then the Eq is reduced to

$kurt(y) = k4 = E\{y^4\} – 3$

From the above equation it is clearly understood that kurtosis is simply a normalised version of the fourth moment. Random mixtures of three or more sources follow gaussian distribution and in order to find one independent component from the mixture, gaussianity has to be changed to non gaussianity by maximizing the value of kurtosis. Kurtosis is zero for Gaussian random variable, positive for super-Gaussian random variable and negative for sub Gaussian random variable.

2.4 Echo Return Loss Enhancement

Echo Return Loss Enhancement (ERLE) can be elucidated as

$ERLE = 10 * \log_{10}[E\{p_s(n)\}2 / E\{e_r(n)\}2]$

Where $E\{p_s(n)^2\}$ is the power of the original echo and $E\{e_r(n)^2\}$ is the power of the residual echo. The optimum value of ERLE should be between 30 and 40 dB. Higher value of ERLE gives higher cancellation of echo. According to ITU-T recommendation G. 167, the value of ERLE should be 25dB for hands free telephony.

2.5 Restrictions in ICA

To make sure that the basic ICA model just given can be estimated, we have to make certain assumptions and restrictions.

1. The independent speech signals are assumed to be statistically *independent*.
2. The independent speech signals must have *non gaussian* distributions.
3. For simplicity purposes, the unknown mixing matrix is assumed to be a *square matrix.*

2.6 Summary

Thus the methodology, fundamentals, kurtosis, ERLE and the restrictions of the project were discussed in this chapter. The algorithm is presented in Chapter 3.

CHAPTER 3

PROPOSED ALGORITHM

This chapter contains the scenarios in which this algorithm was applied and also the proposed algorithm step by step.

3.1 Scenarios

There are three scenarios in total in which the proposed algorithm is applied.

3.1.1 Scenario 1 – Single Channel

Acoustic echo cancellation, realised in a single channel environment is depicted in Fig. There is a near end speaker sitting at the microphone and the far end speaker's output from the speaker bounces of the walls and enters into the microphone along with the near end speech. Now, the proposed algorithm should be able to cancel the acoustic echo (delayed far end speech) and provide the near end speech as the output.

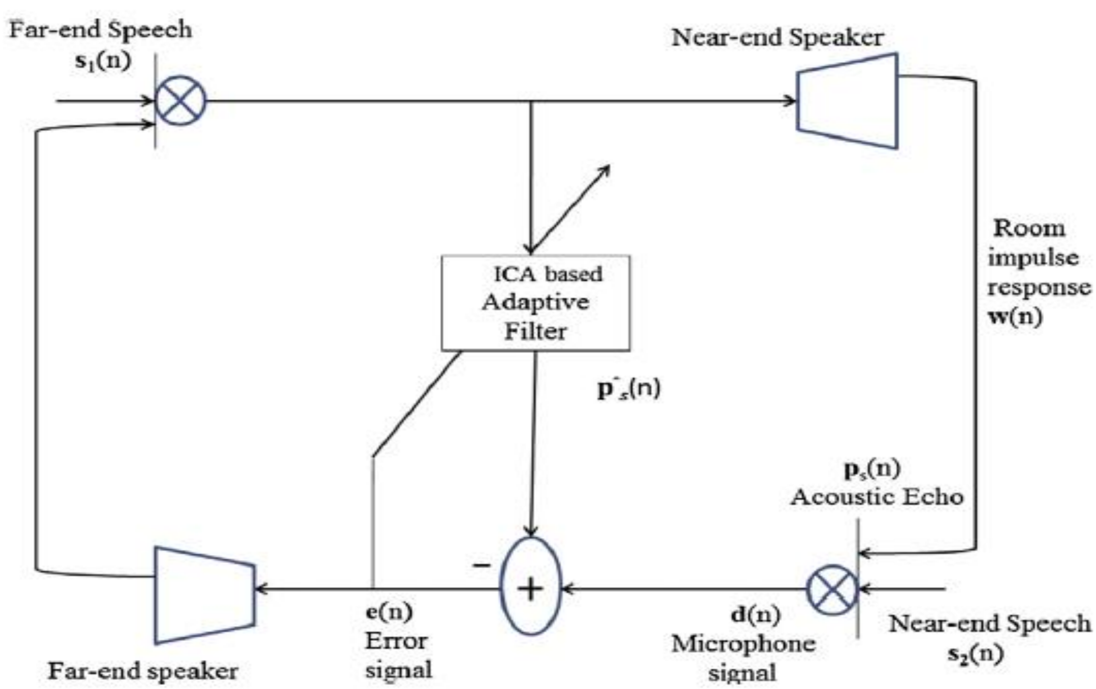
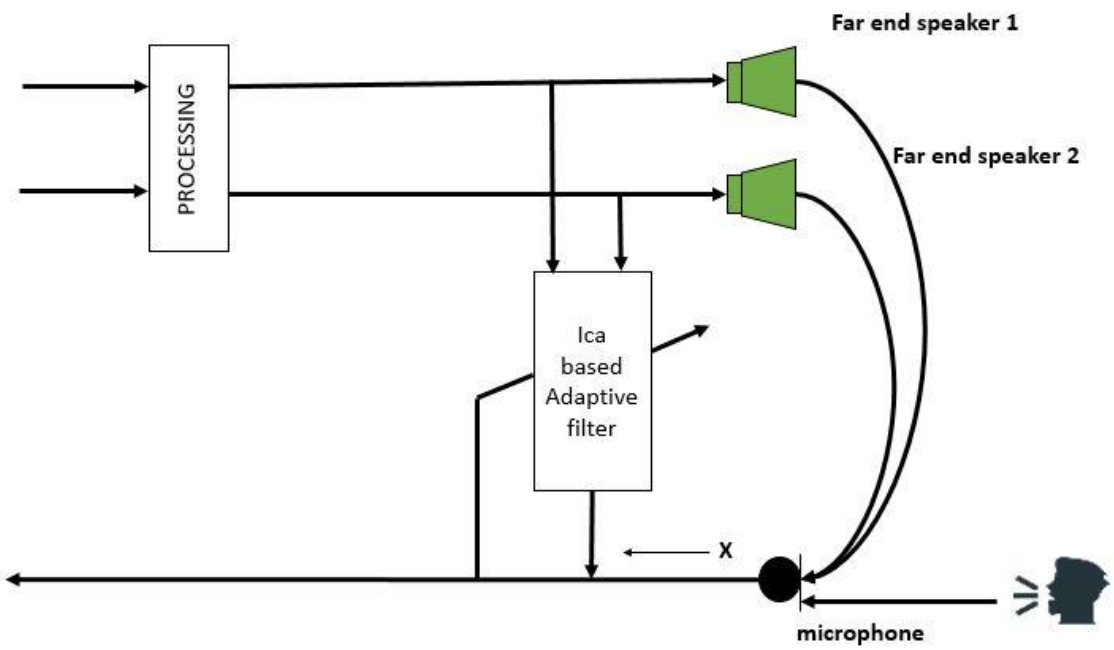


Fig. 2. Acoustic Echo Cancellation system.

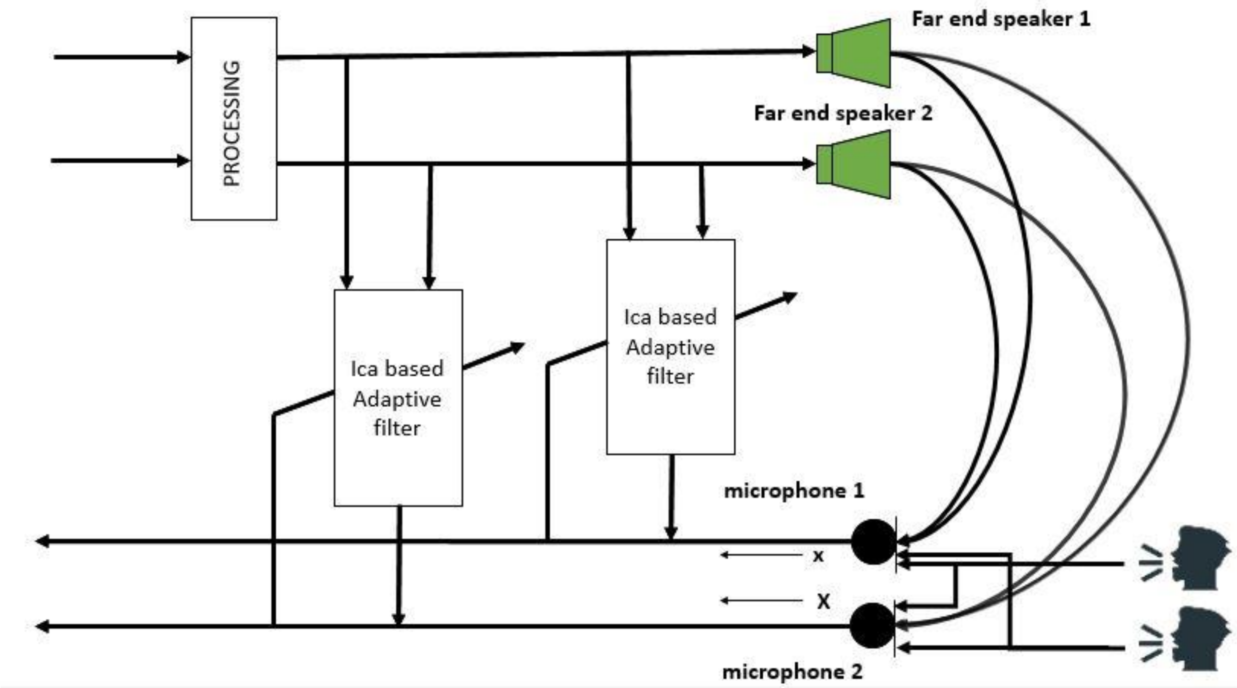
3.1.2 Scenario 2 – Multi Channel with Single Microphone

In this particular scenario, there are two far end speakers involved but only one near end speaker i.e only one microphone but two speakers. This is clearly depicted in Fig. So, the combination of two far end speeches is the echo in this scenario. This echo passes through the microphone along with the one near end speech while mixing randomly. Now, the proposed algorithm should be able to separate the three signals and cancel the acoustic echo (two far end speeches) and provide the near end speech as output.



3.1.3 Scenario 3 – Multi Channel with Dual Microphones

Two far end speakers and two near end speakers are involved in this scenario. So, there are two inputs and two outputs. For microphone 1, the two far end speeches and the second near end speech combined becomes the echo. This echo is randomly mixed with the first near end speech and passed as microphone 1 input. Similarly, the two far end speeches and the first near end speech combined become the echo for the second microphone. This echo is then randomly mixed with the respective near end speech (second near end speech) and passed as microphone 2 input. The proposed algorithm should be able to cancel the acoustic echo and provide the two near end speeches as output.



3.2 Kurtosis Algorithm Using Gradient Approach

1. The mean is made zero by centering the data.
2. The observed data **x** is whitened using **z = v * x** (with **x** as zero mean), where $\mathbf{v} = \mathbf{D}^{-1/2} * \mathbf{E}^{T}$ is a whitening matrix. D & E are Eigen value & Eigen vector matrix of covariance of **x** respectively.
3. Take an initial mixing matrix W and normalize it.
4. The direction in which the absolute value of kurtosis of $y = W^{T} * Z$ increases is computed.
5. The initial random matrix taken is W = 0 of norm 1 with L = 1.
6. Random matrix is updated using $W(L) = W(L-1) + \alpha(L) * \text{sign}(\text{kurt}(W(L-1)^{T} * Z)) * E[Z * (W(L-1)^{T} * Z)^{3}]$.
7. W(L) is normalized every time using W(L) = W(L) / ||W(L).
8. $|W(L)^{T} * W(L-1)|$ should be close enough to 1, if not, then L = L + 1 and go back to step 6. If it is close enough to 1 then output is the vector W(L) i.e move the vector W in that direction.

Before applying step 1 of the above algorithm the audio files taken as inputs should be converted to matrix form. After step 8 the Echo Return Loss Enhancement (ERLE) is calculated using the below formula:

ERLE = $10*\log_{10}[E\{p_s(n)\}2 / E\{e_r(n)\}2]$.

The output of the algorithm is a matrix, so the matrix has to be converted back into an audio format so that we can clearly judge the working of the proposed algorithm.

3.3 Summary

This chapter describes how the proposed algorithm can be implemented step by step and also to what scenarios this particular algorithm was applied to. There are three scenarios to which this proposed algorithm was applied:

1. Single Channel environment.
2. Multichannel with one microphone.
3. Multichannel with dual microphones.

CHAPTER 4

4.1 Cost Analysis

The primary aim of this project is to cancel the acoustic echo in a teleconferencing system. So, the cost also includes teleconferencing equipment like microphones and speakers.

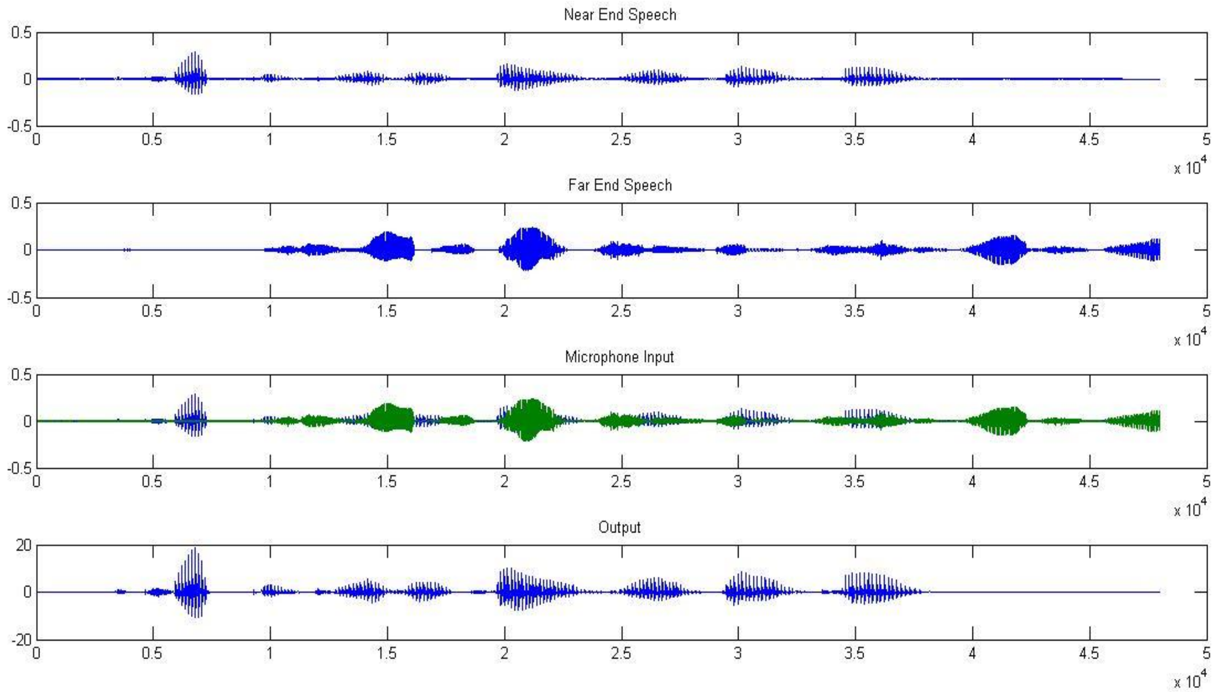
COMPONENT	**COST**
Microphone (2 No.s)	Rs 500
Speaker (4 No.s)	Rs 600
DSP Processor Kit	Rs 28000
TOTAL	Rs 29100

Since the DSP Processor Kit is used, the cost is very high but the entire kit can be miniaturised into one single chip if commercially implemented, then the cost will drop drastically. So, if this project is implemented in a large scale then it is very economical for all the users.

CHAPTER 5

5.1 Results

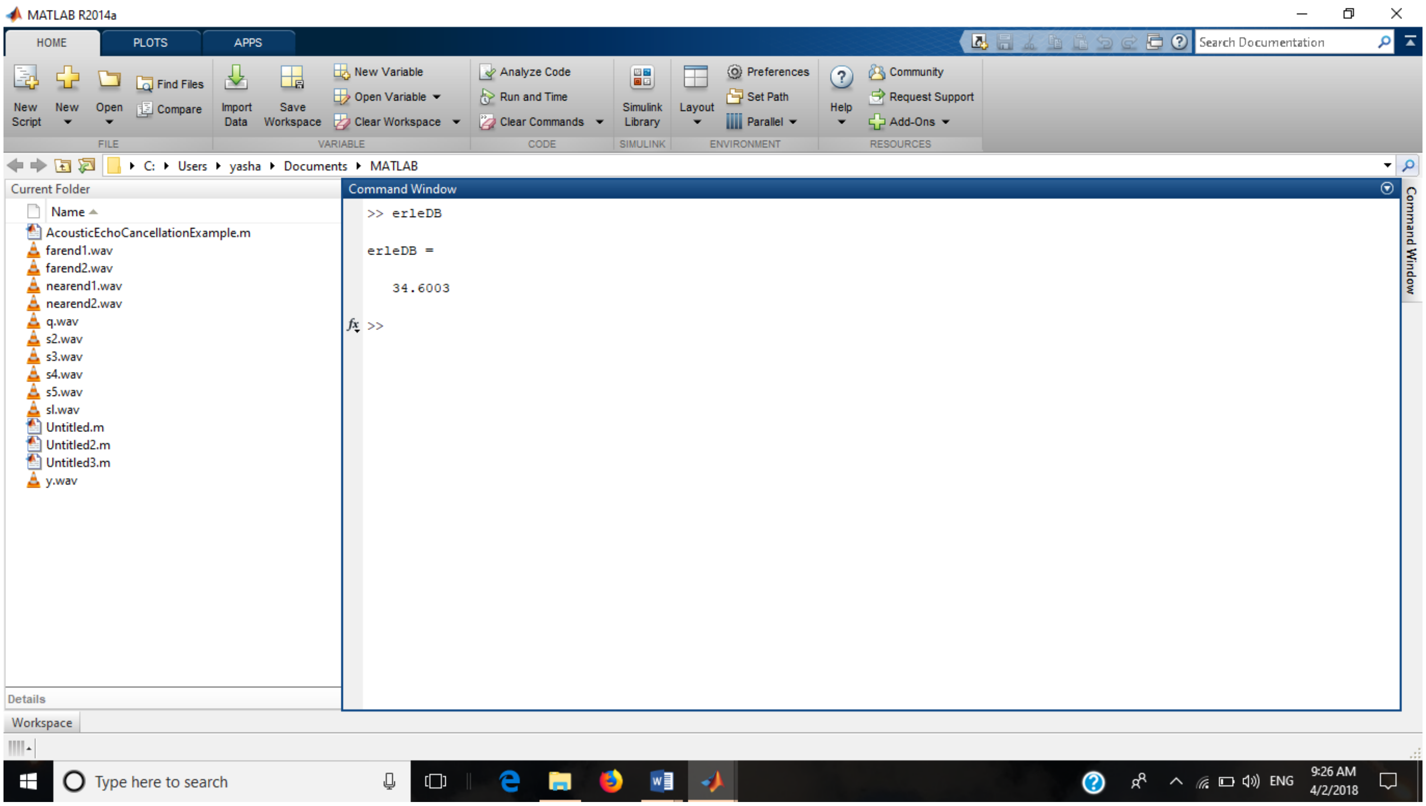
5.1.1 Single Channel



The above Fig consists of the near end speech, far end speech, mic input and also the output of the proposed algorithm.

Echo Return Loss Enhancement (ERLE) is the primary parameter in judging the performance of the proposed algorithm. For this particular scenario consisting one far end speaker and one near end speaker the ERLE in dB is found to be 34.6003 which is very good. Any value of ERLE above 25 dB is fine according to the ITU standard.

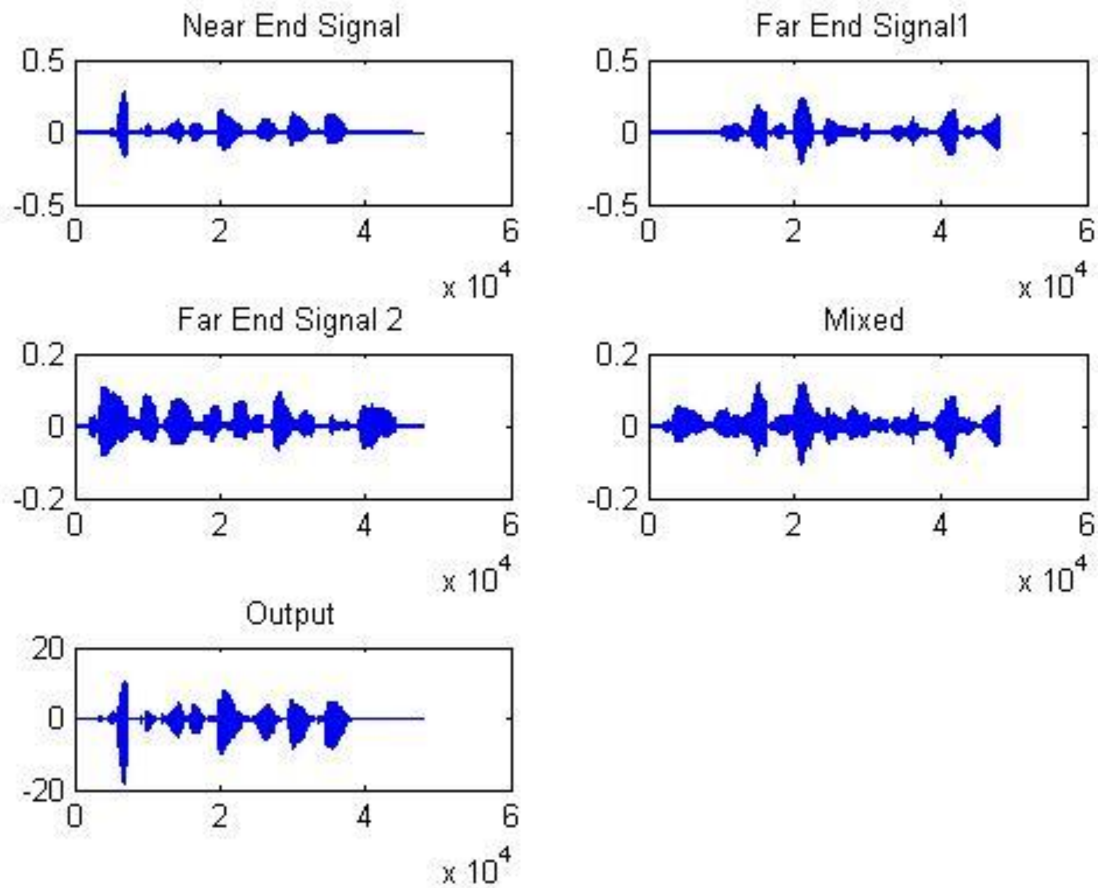
The below Fig shows the ERLE value of this particular scenario in dB.

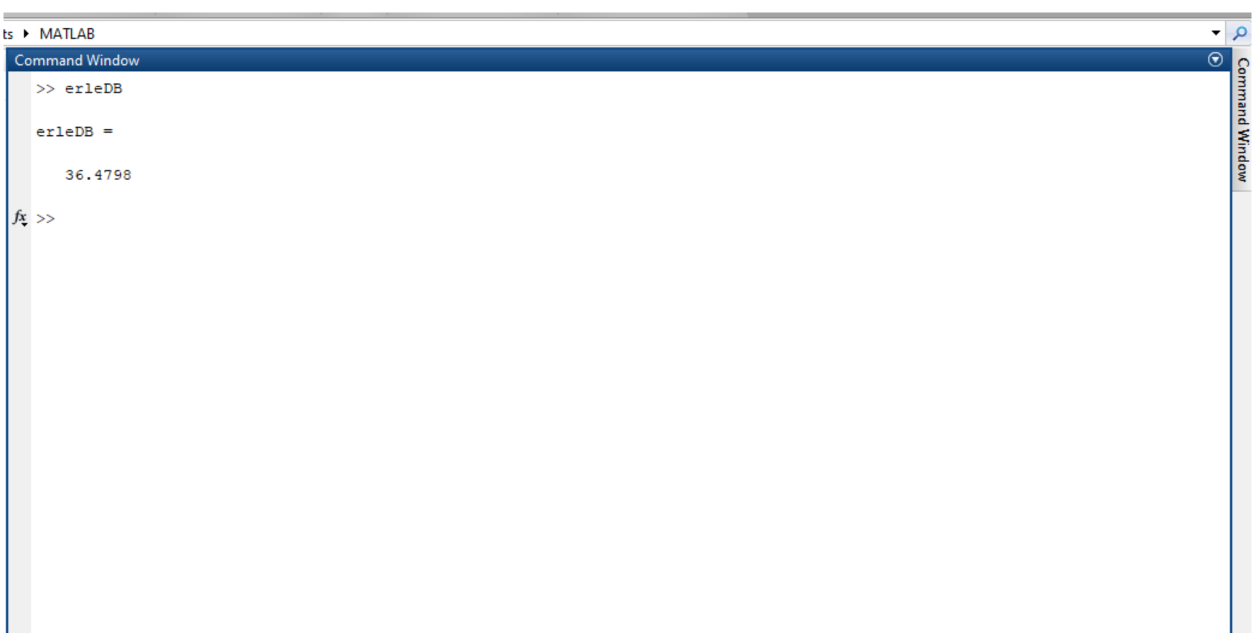


5.1.2 Scenario 2 – Multi Channel with Single Microphone

In in particular scenario there are two far end speakers and one near end speaker. The combination of the two far end speeches is the acoustic echo which needs to be cancelled. The acoustic echo then randomly mixes with the near end speech making that mixed signal as the microphone input.

The below Fig shows all the speeches of this scenario in a graphical format. The ERLE in dB which is the primary parameter to judge this algorithm comes out to be 36.4 dB which is very good. According to the standards anything above 25 dB is fine so, 36.4 dB is considered very good. The snapshot of the ERLE value in dB is also provided below.

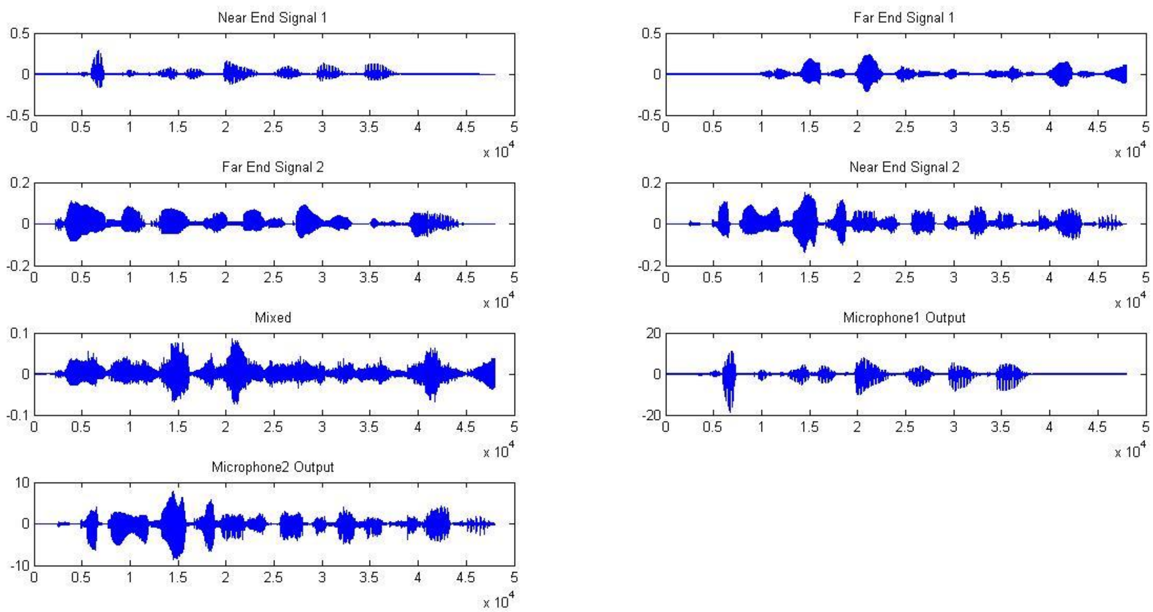

Near End Signal
Far End Signal1
Far End Signal 2
Mixed
Output
x 10^4


MATLAB
Command Window
>> erleDB
erleDB =
36.4798
>>

5.1.3 Scenario 3 – Multi Channel with Dual Microphones

There are two far end speakers and two near speakers in this scenario. So, effectively there are two microphone inputs and two microphone outputs. The combination of the two far end speeches along with one of the near end speech is the acoustic echo in this case. This acoustic echo then randomly mixes with the other near end speech making that mixed signal as the microphone input.

The below Fig clearly depicts all the aspects related to inputs and outputs. The ERLE in this particular case was to found to be 38.2 dB which is very good. Anything near 40 dB is considered very good but according to the standards it only needs to be above 25 dB.



5.2 Summary

Through this chapter the working of the proposed algorithm has been conveyed in the form of results. The ERLE values in dB and the input & output signals for each and every scenario has been provided clearly. Compared to the ITU standards, the acquired ERLE values in dB are better.

CHAPTER 6

6.1 Conclusion

The primary aim of this project was to cancel the acoustic echo in a multichannel environment by developing an ICA based technique, specifically using maximisation of non-gaussianity. The proposed algorithm was first tested in a single channel environment and when it was successful, it was applied to a multichannel environment. The multichannel environment was split into two scenarios with the first scenario having only microphone and the second scenario having dual microphones. The proposed algorithm was applied to both the scenarios and the results were satisfactory with both the scenarios producing ERLE values of above 35 dB. With the ERLE value above 35 dB, the secondary aim, which was to maintain a good ERLE, was also achieved.

6.2 Future Work

For any algorithm the execution time is very important, like in this project there is no use if the cancellation of the acoustic echo takes too much time since it is a teleconferencing application. The user of the teleconferencing system cannot wait that long. The best way to reduce the wait time for this algorithm is to improve the pre-processing. The only pre-processing done in this proposed algorithm is centering and whitening, which helps in reducing the wait time but not quite. With the help of wavelet based techniques along with centering and whitening the execution time can be reduced. The addition of the wavelet based pre-processing is a good way to go in the future.

APPENDIX

SINGLE CHANNEL

```
clc
clear all
[s1,Fs] = audioread('nearend.wav');
s1 = [s1;zeros(1612,1)];
n1=1:length(s1);


[s2,Fs] = audioread('farend1.wav');
samples = [1,3*Fs];
[s2,Fs] = audioread('farend1.wav',samples);
n2=1:length(s2);

s1 = s1 - mean(s1);
s2 = s2 - mean(s2);
M = rand(2,2);

s = [s1,s2];
x = M * s';
c = cov(x');
[E, D] = eig(c);
v = E * D^(-0.5) * E';
z = v*x;
w = rand(1,2);

 L = 100;

eta=2e-2;
for i=1:L
    % Get estimated source signal, y.
    y = w*z;

    % Get estimated kurtosis.
    K = mean(y.^4)-3;

    y3=y.^3;
    yy3 = y3';
    yy3 = repmat(yy3,1,2);
    g = mean( yy3.*z' );

    % Update w to increase K ...
    w = w + eta*g;

    % Set length of w to unity ...
```

```
    w = w/norm(w);

end;
y = w*z;




% Listen to extracted signal ...
    soundsc(y,Fs);


% p = rms(s4).^2;
p = (norm(s2)^2)/length(s2);
e = x - repmat(y,2,1);
ee = e - repmat(s1',2,1);
% ee = ee;
% pp = rms(ee).^2;
pp = (norm(ee)^2)/length(ee);
erle = p./pp;
erleDB = -10*log10(erle);
```

MULTICHANNEL SCENARIO 1

```
clc
clear all
[s1,Fs] = audioread('nearend1.wav');
s1 = [s1;zeros(1612,1)];
n1=1:length(s1);



[s2,Fs] = audioread('farend1.wav');
samples = [1,3*Fs];
[s2,Fs] = audioread('farend1.wav',samples);
n2=1:length(s2);

[s3,Fs] = audioread('farend2.wav');
sample = [1,3*Fs];
[s3,Fs] = audioread('farend2.wav',sample);
n3 = 1:length(s3);

s1 = s1 - mean(s1);
s2 = s2 - mean(s2);
s3 = s3 - mean(s3);

M = randn(2,2);
s4 = [s2';s3'];
s4 = mean(s4);
```

```
s = [s1';s4];
x = M * s;
c = cov(x');
[E, D] = eig(c);
v = E * D^(-0.5) * E';
z = v*x;
w = randn(2,1);
w = w/norm(w);

 L = 100;

eta=2e-2;
for i=1:L
    % Get estimated source signal, y.
    y = w'*z;

    % Get estimated kurtosis.
    K = mean(y.^4)-3;



    y3=y.^3;
    yy3 = y3';
    yy3 = repmat(yy3,1,2);
    g = mean( yy3.*z' );



    % Update w to increase K ...
    w = w + eta*g';

    % Set length of w to unity ...
    w = w/norm(w);


end;
y = w'*z;




% Listen to extracted signal ...
soundsc(y,Fs);

p = (norm(s4)^2)/length(s4);
% e = x - repmat(y,2,1);
% ee = y - s1';
%e = x - repmat(y,2,1);
e = x - repmat(s1',2,1);
% ee = e - repmat(s1',2,1);
ee = y - s1';
```

```
pp = (norm(ee)^2)/length(ee);
erle = p./pp;
erleDB = -10*log10(erle);
```

MULTICHANNEL SCENARIO 2

```
clc
clear all
[s1,Fs] = audioread('nearend1.wav');
s1 = [s1;zeros(1612,1)];
n1=1:length(s1);



[s2,Fs] = audioread('farend1.wav');
samples = [1,3*Fs];
[s2,Fs] = audioread('farend1.wav',samples);
n2=1:length(s2);

[s3,Fs] = audioread('farend2.wav');
sample = [1,3*Fs];
[s3,Fs] = audioread('farend2.wav',sample);
n3 = 1:length(s3);

[s5,Fs] = audioread('nearend2.wav');
sample = [1,3*Fs];
[s5,Fs] = audioread('nearend2.wav',sample);
n5 = 1:length(s5);

s1 = s1 - mean(s1);
s2 = s2 - mean(s2);
s3 = s3 - mean(s3);
s5 = s5 - mean(s5);

M = randn(2,2);
s4 = [s2';s3';s5'];
S4 = [s2';s3';s1'];
S4 = mean(S4);
s4 = mean(s4);
s = [s1';s4];
x = M * s;
c = cov(x');
[E, D] = eig(c);
v = E * D^(-0.5) * E';
z = v*x;
w = randn(2,1);
w = w/norm(w);

S = [s5';S4];
```

```
X = M * S;
C = cov(X');
[Ei, Di] = eig(C);
V = Ei * Di^(-0.5) * Ei';
Z = V * X;

 L = 100;

eta=2e-2;
for i=1:L
    % Get estimated source signal, y.
    y = w'*z;

    % Get estimated kurtosis.
    K = mean(y.^4)-3;

        y3=y.^3;
    yy3 = y3';
    yy3 = repmat(yy3,1,2);
    g = mean( yy3.*z' );



    % Update w to increase K ...
    w = w + eta*g';

    % Set length of w to unity ...
    w = w/norm(w);

end;
y = w'*z;




for i=1:L
    % Get estimated source signal, y.
    Y = w'*Z;

    % Get estimated kurtosis.
    Ku = mean(Y.^4)-3;

    % Find gradient @K/@w ...
    Y3=Y.^3;
    YY3 = Y3';
    YY3 = repmat(YY3,1,2);
    G = mean( YY3.*Z' );
```

```
% Update w to increase K ...
    w = w + eta*G';

    % Set length of w to unity ...
    w = w/norm(w);
    % Record h and angle between wopt and gradient ...

end;
Y = w'*Z;




% Listen to extracted signal ...
soundsc(y,Fs);
soundsc(Y,Fs);

p = (norm(s4)^2)/length(s4);
% e = x - repmat(y,2,1);
% ee = y - s1';
%e = x - repmat(y,2,1);
e = x - repmat(s1',2,1);
% ee = e - repmat(s1',2,1);
ee = y - s1';
pp = (norm(ee)^2)/length(ee);
erle = p./pp;
erleDB = -10*log10(erle);




P = (norm(S4)^2)/length(S4);
% e = x - repmat(y,2,1);
% ee = y - s1';
%e = x - repmat(y,2,1);
E = X - repmat(s5',2,1);
% ee = e - repmat(s1',2,1);
EE = Y - s5';
PP = (norm(EE)^2)/length(EE);
ERLE = P./PP;
ERLEDB = -10*log10(ERLE);
```

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