Louis Rosenblum
EEE509 - ASU
06/16/2020

Project 2 Report

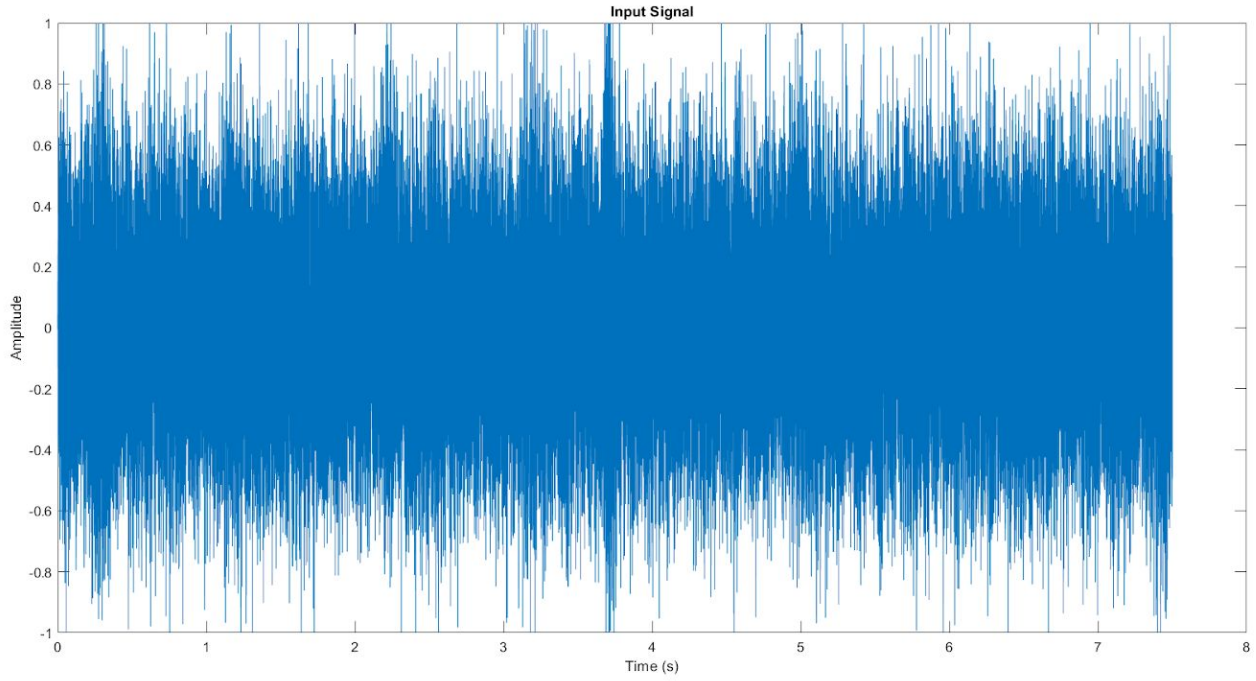
The Frequency Domain Adaptive Filter:

An adaptive noise-cancelling filter was developed for this project. This filter adapts its coefficients in the frequency domain based on data collected from the filter input. One main advantage to this method over time-domain adaptive filters is that the low computational complexity of the FFT can be exploited. In addition to this high order filters (long delays) aren't always necessary for FDAF.

(2006) Frequency-Domain Adaptive Filters. In: Acoustic MIMO Signal Processing. Signals and Communication Technology. Springer, Berlin, Heidelberg

Block Adaptive Filters and Frequency Domain Adaptive Filters . Tampere University of Technology, CS. http://www.cs.tut.fi/~tabus/course/ASP/SGN2206LectureNew6.pdf

The signal shown below was used as the input signal.



The signal to noise ratio of the input signal before filtering is measured at -21.13.

```
snr1 =

  -21.1297

fx >>
```

MATLAB Program:

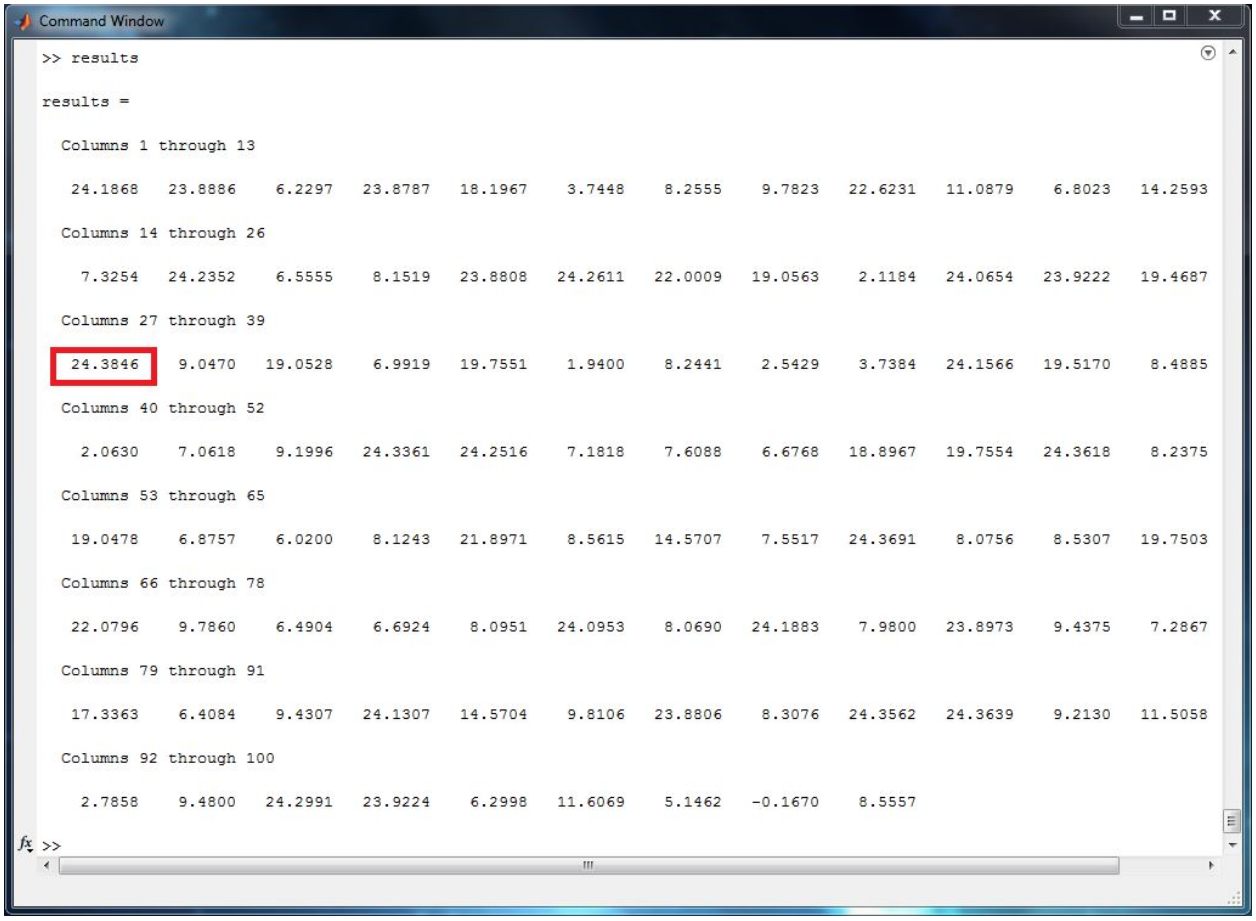
(See appendix for full code)

A MATLAB program was developed to simulate the adaptive digital filter. The program contains an initialization state, file I/O, signal deconstruction into frames, parallel processing, the filter instantiation, and plots/results.

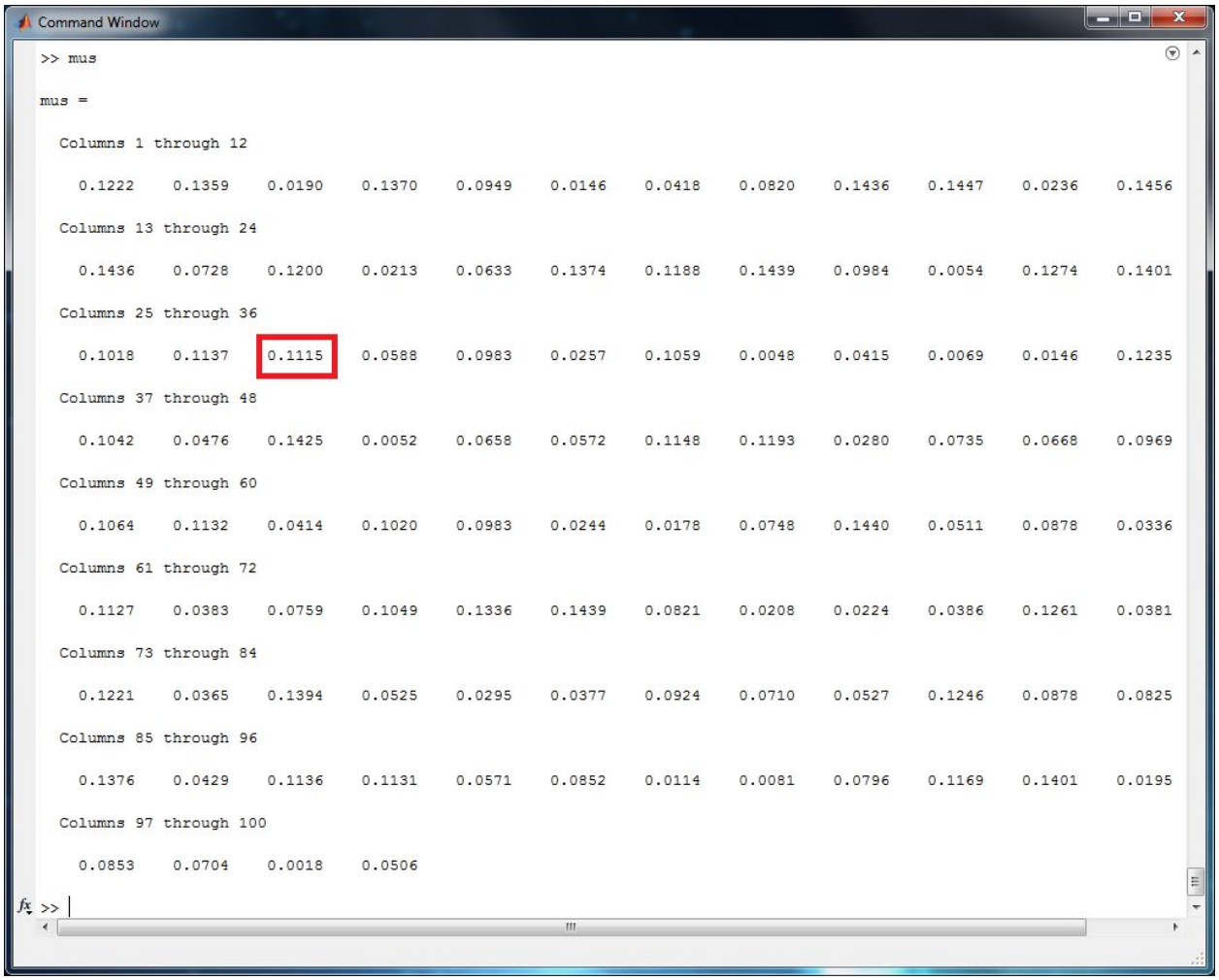
Another script was also created in order to determine the optimal mu value for each N-point FFT.

Determining mu:

The optimal mu was determined used MATLAB's random number generator with an upper limit of mu = 0.15. Values reaching not far above 0.15 would cause instability in the filter. 100 different mus between 0 and 0.15 were tested for each N-point fft case. Below the results from N = 64 can be seen.



SNR Difference between output and input at each mu

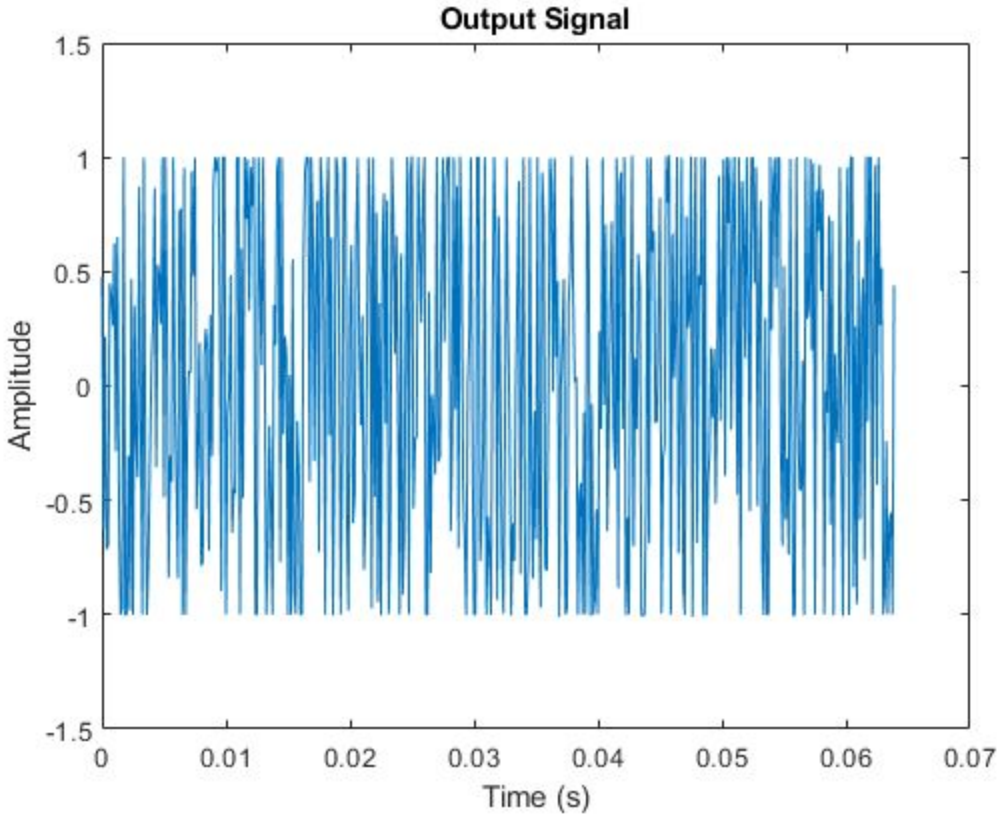


Test vector of mu values

Results

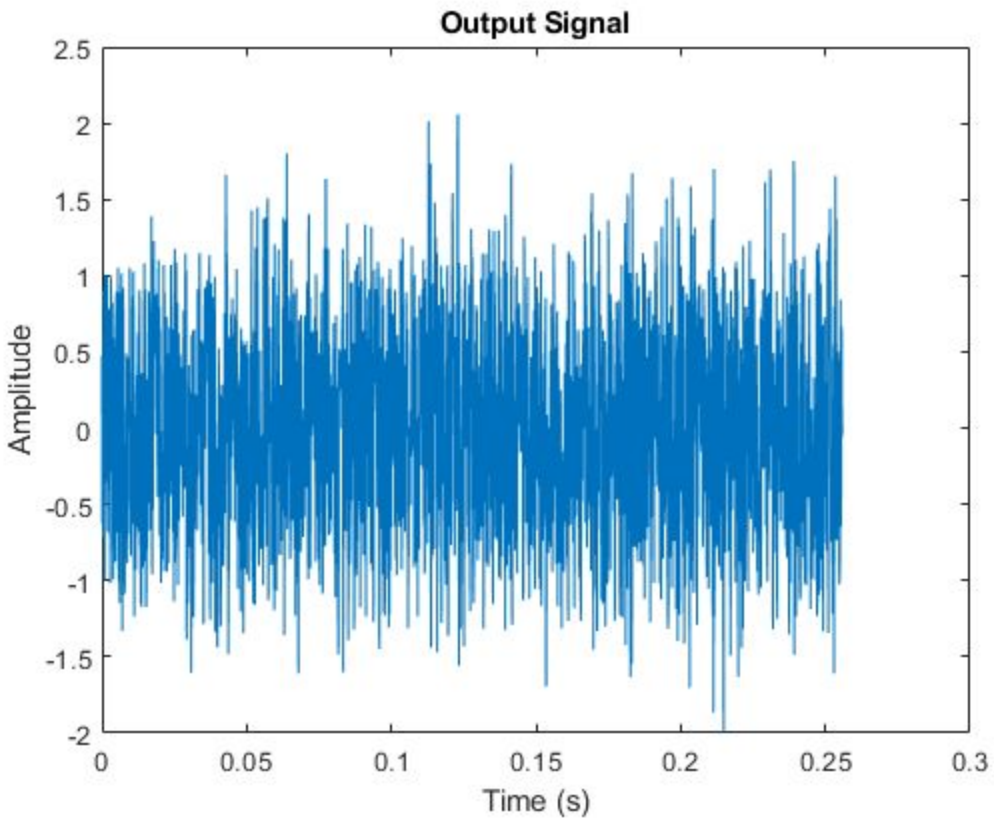
For each N-point FFT's optimal mu value the output signal and the difference in SNR between the input signal and output signal can be seen for each of the cases below:

N = 4



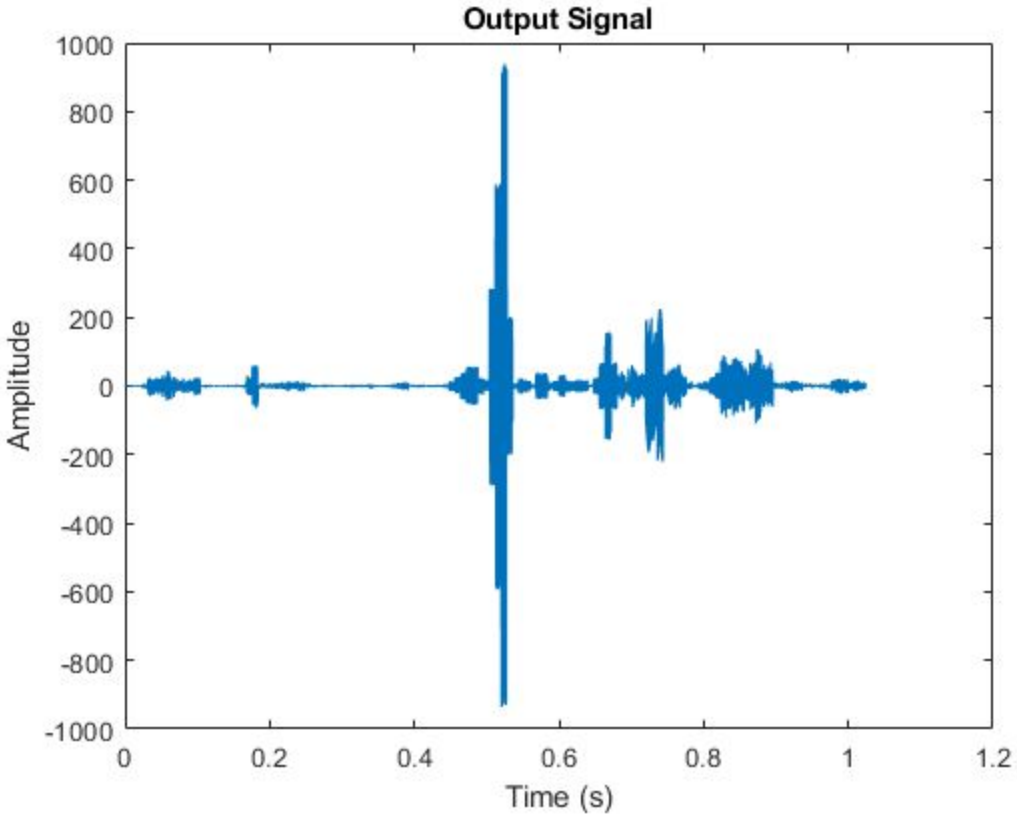

```
snr1 =

  -21.1297


snr2 =

   -9.6178

>> diff

diff =

   11.5120
```

N = 16



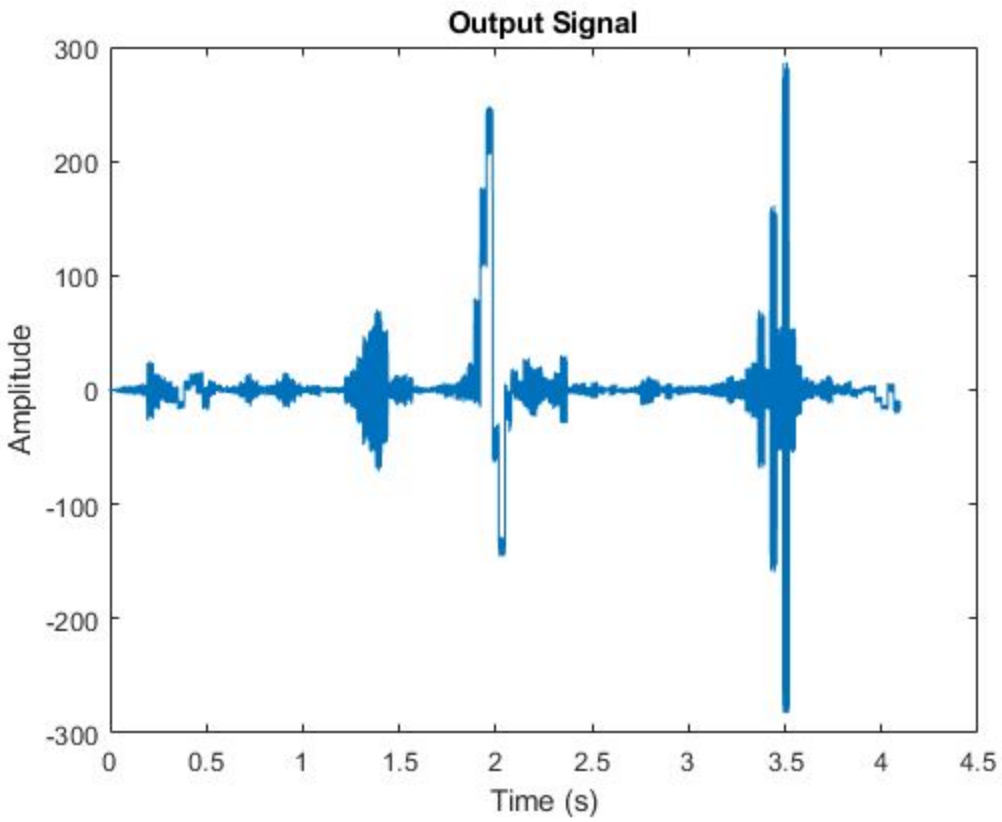

```
snr1 =

  -21.1297


snr2 =

  -11.9589


diff =

    9.1709
```

N = 64




```
snr1 =

  -21.1297


snr2 =

    3.2545


diff =

   24.3842
```

N = 256



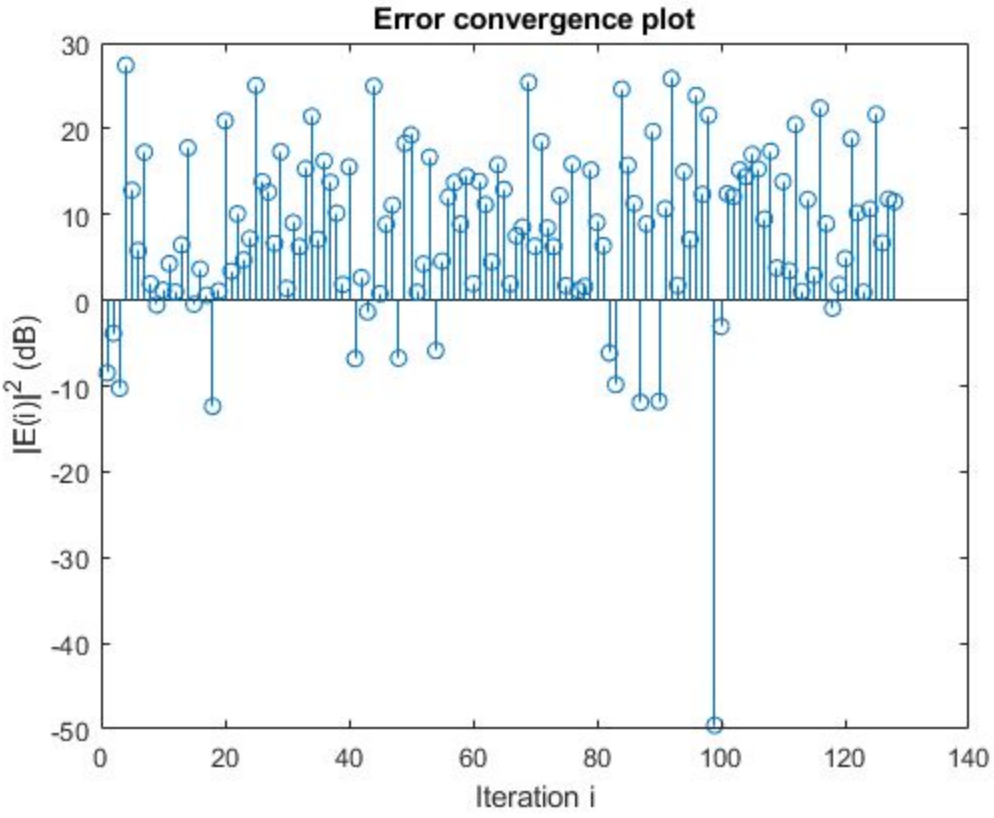

```
snr1 =

  -21.1297


snr2 =

    8.3085


diff =

   29.4383
```

Full simulation results:

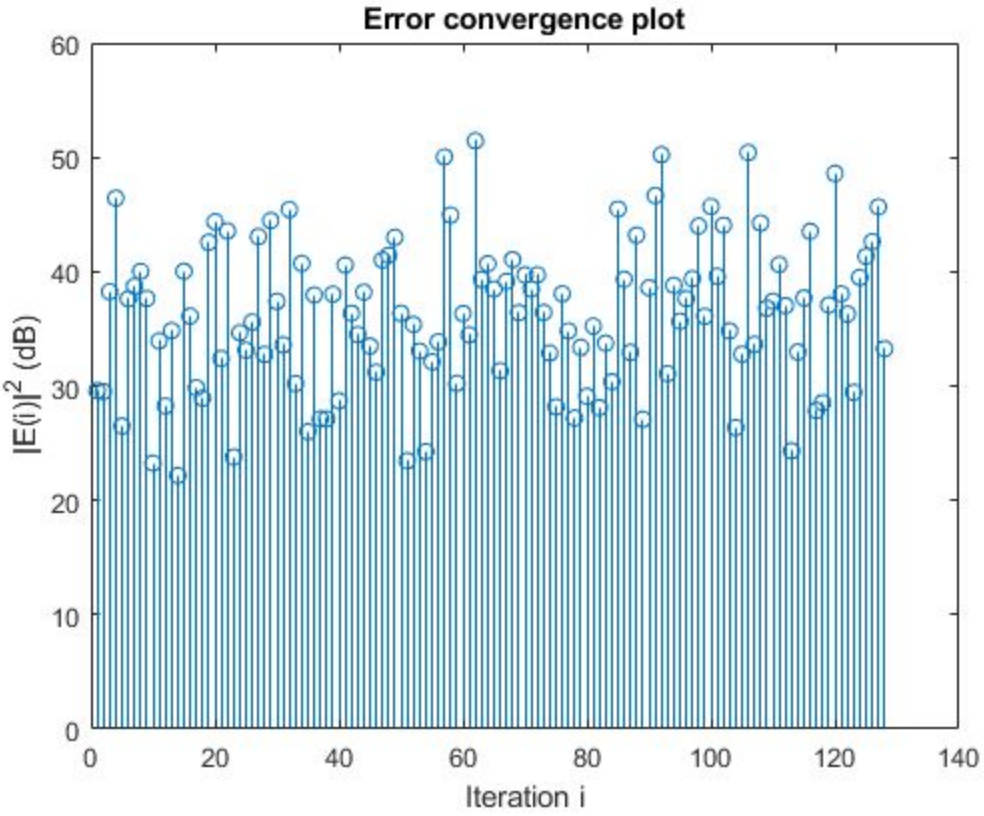
N-point FFT	SNR Improvement (dB)	mu	nFrames
4	11.51	0.0007	128
16	9.17	0.0636	128
64	24.38	0.1115	128
256	29.44	0.0235	128

Error Convergence Charts:

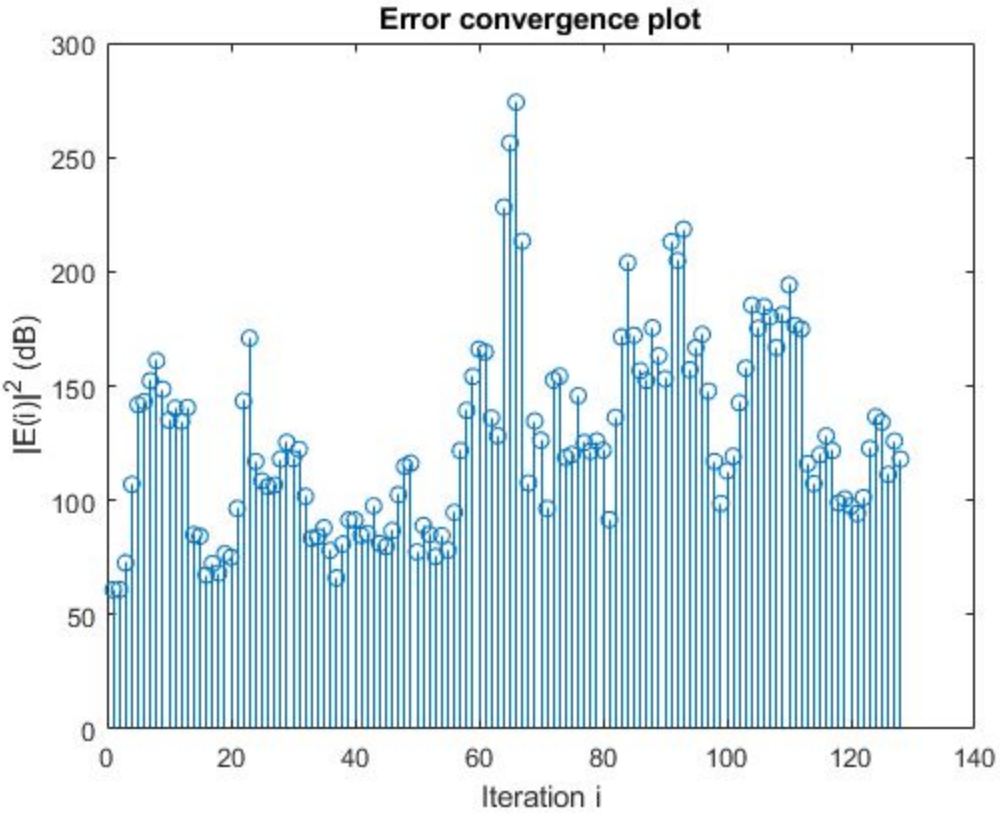
N = 4



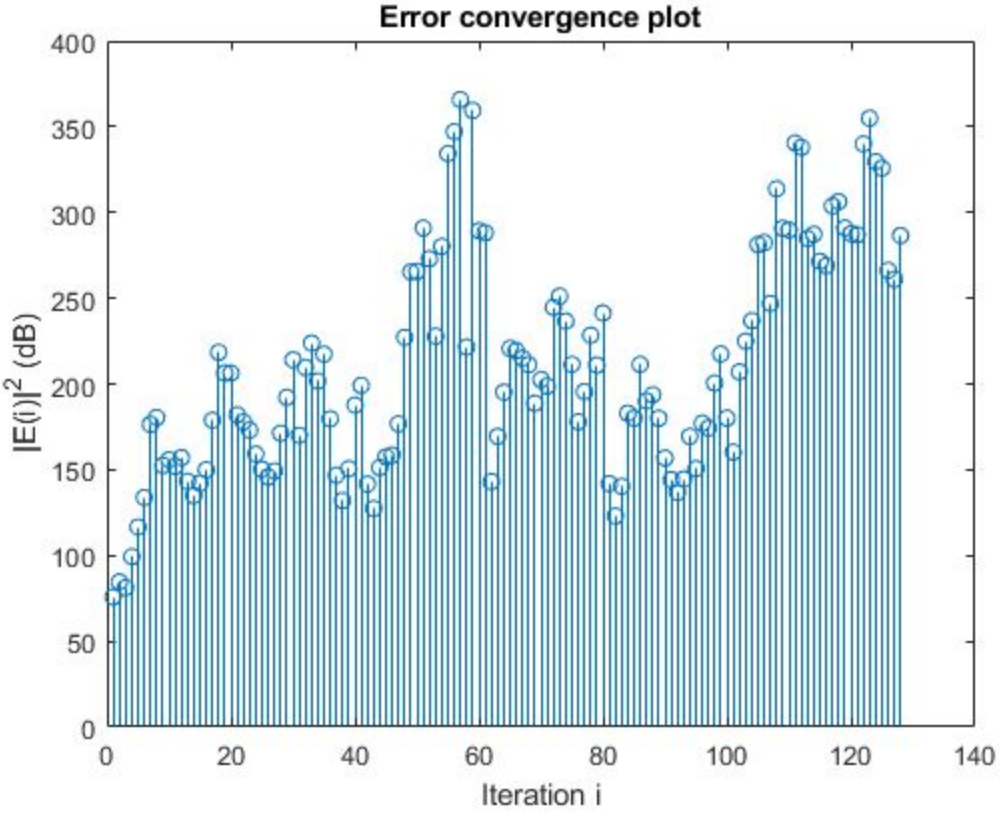
N = 16:


Error convergence plot
60
50
40
30
20
10
0
|E(i)|^2 (dB)
0
20
40
60
80
100
120
140
Iteration i

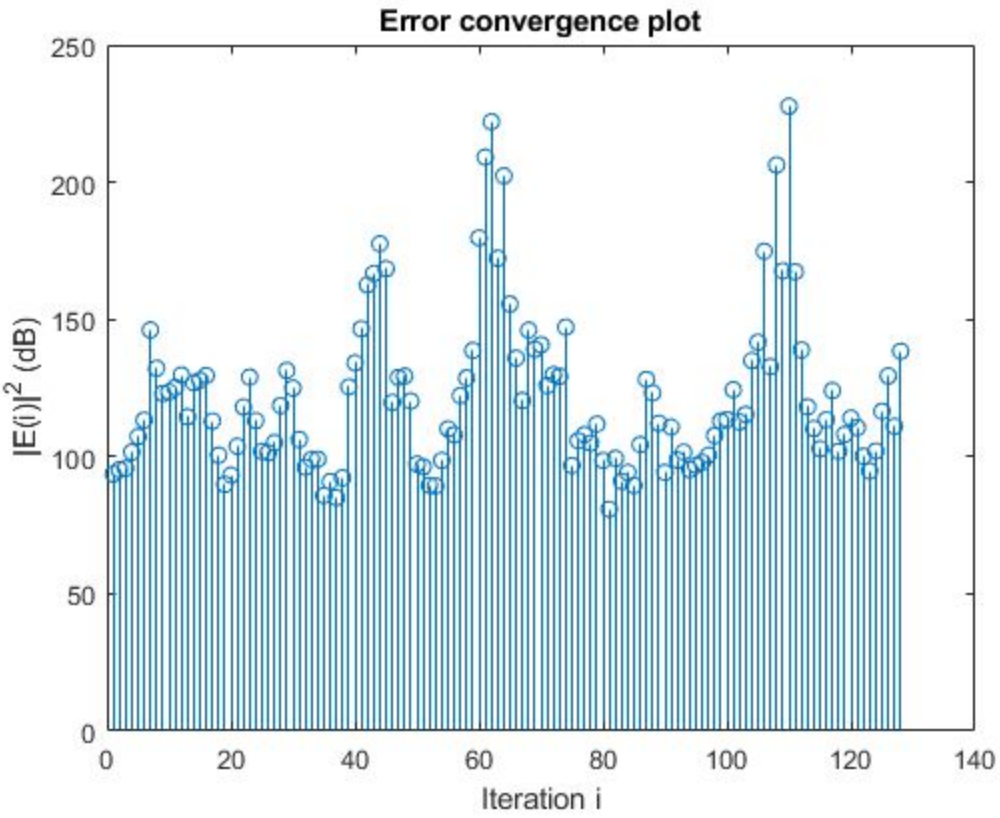
N = 64:


Error convergence plot
300
250
200
150
100
50
0
|E(i)|^2 (dB)
0
20
40
60
80
100
120
140
Iteration i

N = 128:

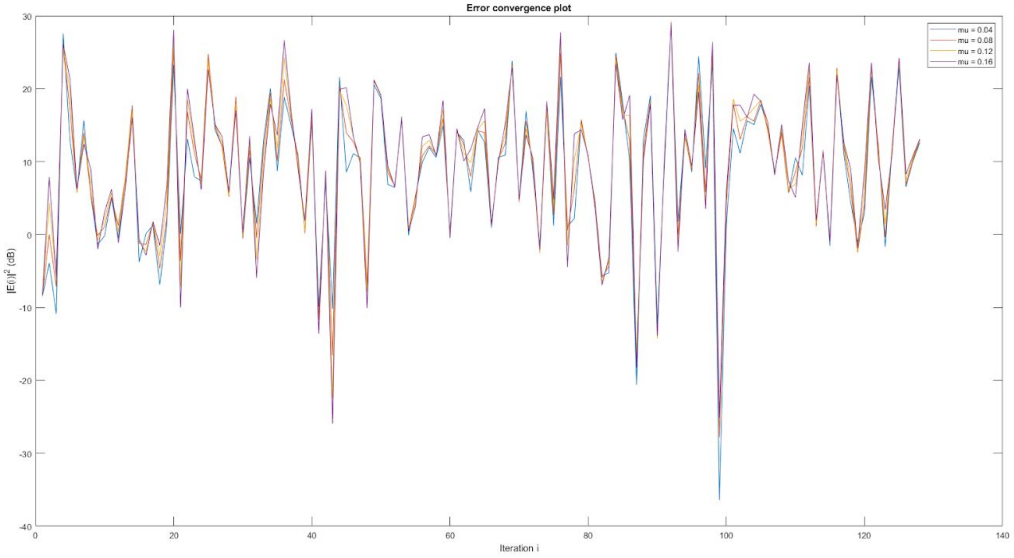

Error convergence plot
400
350
300
250
200
150
100
50
0
|E(i)|^2 (dB)
0
20
40
60
80
100
120
140
Iteration i

N = 256:

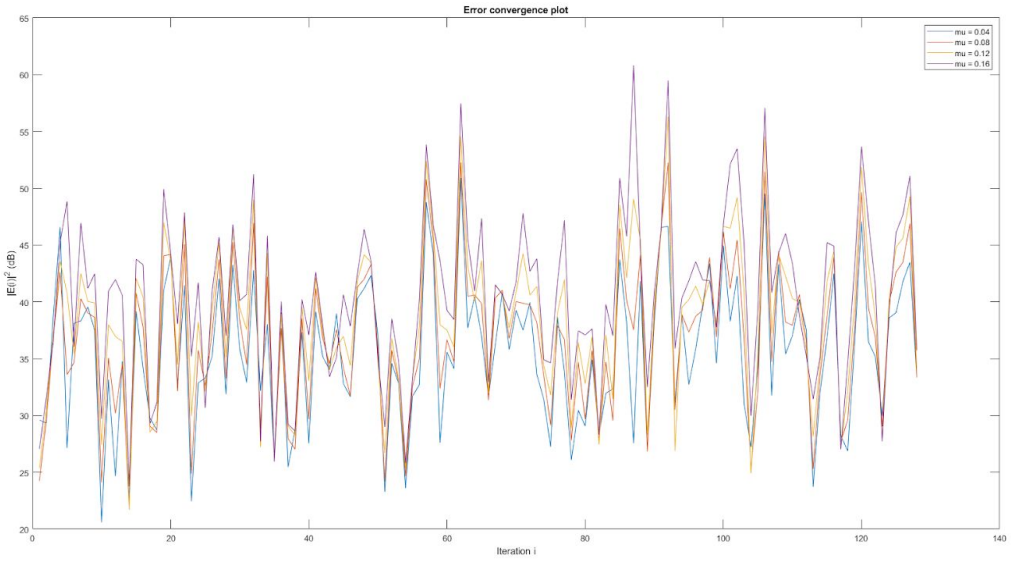


The following error convergence plots were generated at different mu values for each N-point FFT case. At N = 256 the filter becomes unstable by the time mu reaches a value of 0.16.

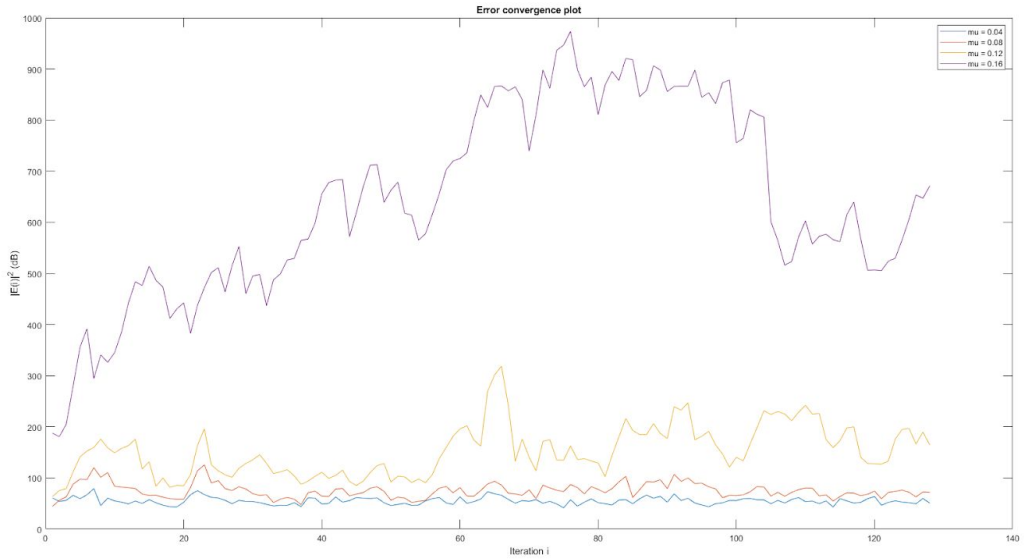
N = 4:



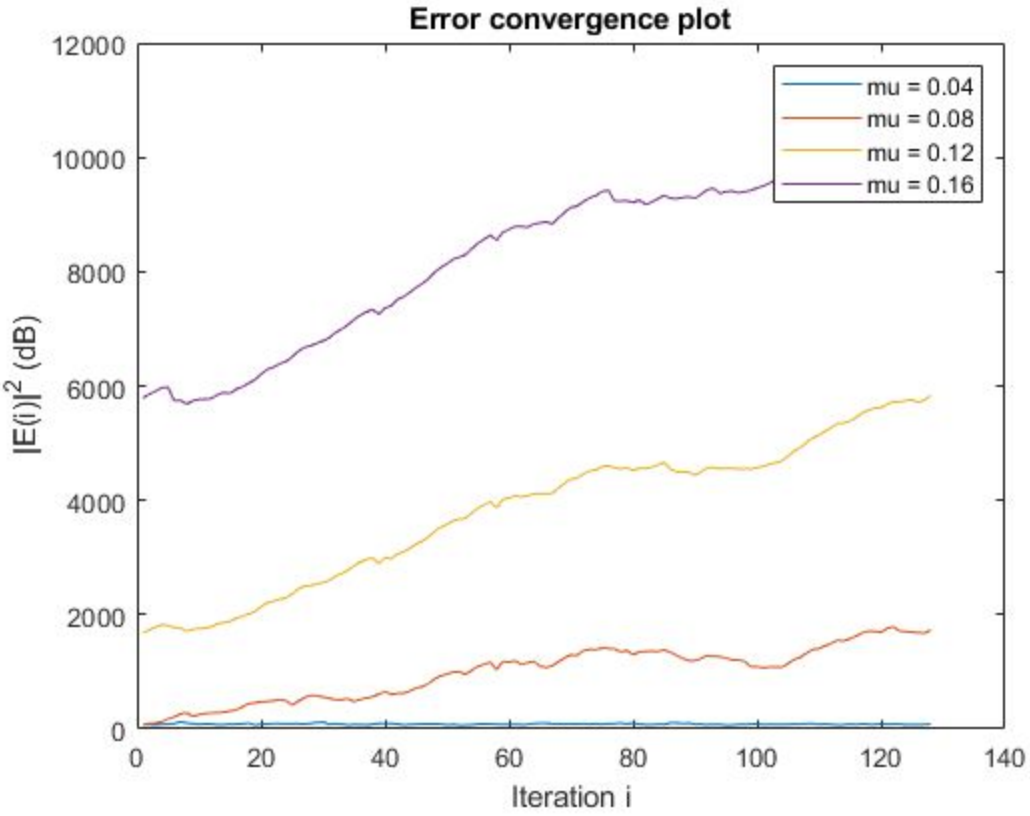
N = 16:


Error convergence plot
mu = 0.04
mu = 0.08
mu = 0.12
mu = 0.16
|E(i)|^2 (dB)
Iteration i

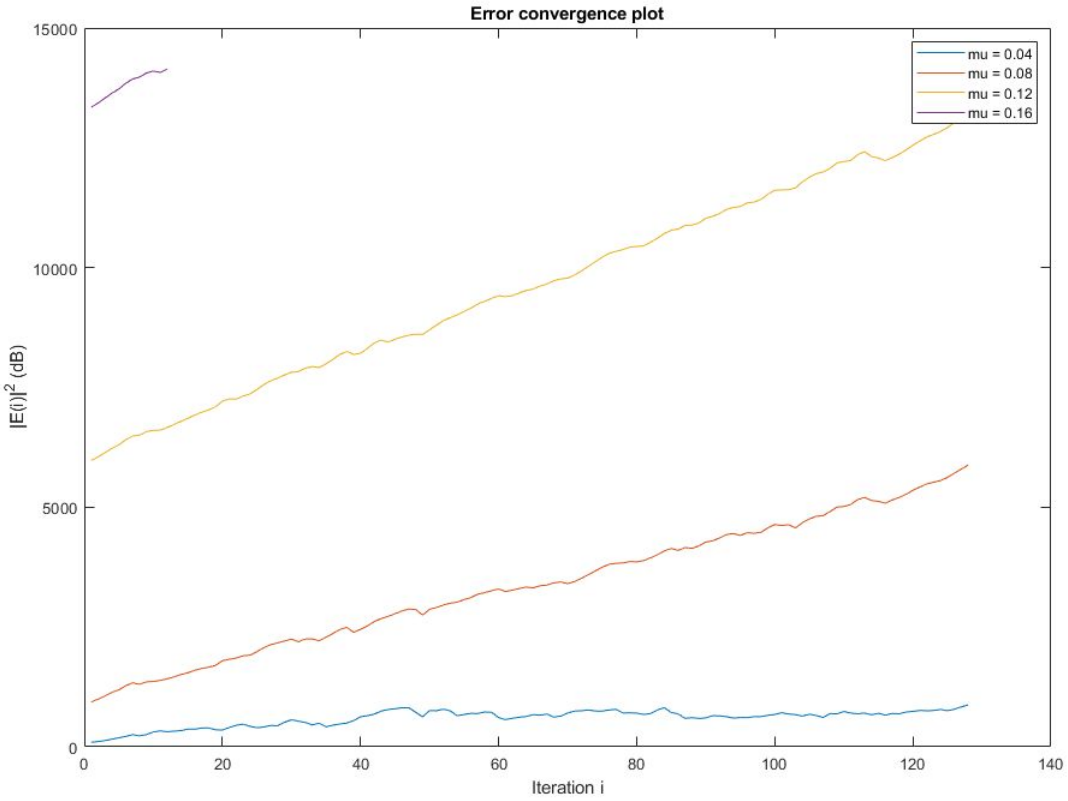
N = 64:


Error convergence plot
mu = 0.04
mu = 0.08
mu = 0.12
mu = 0.16
|E(i)|^2 (dB)
Iteration i

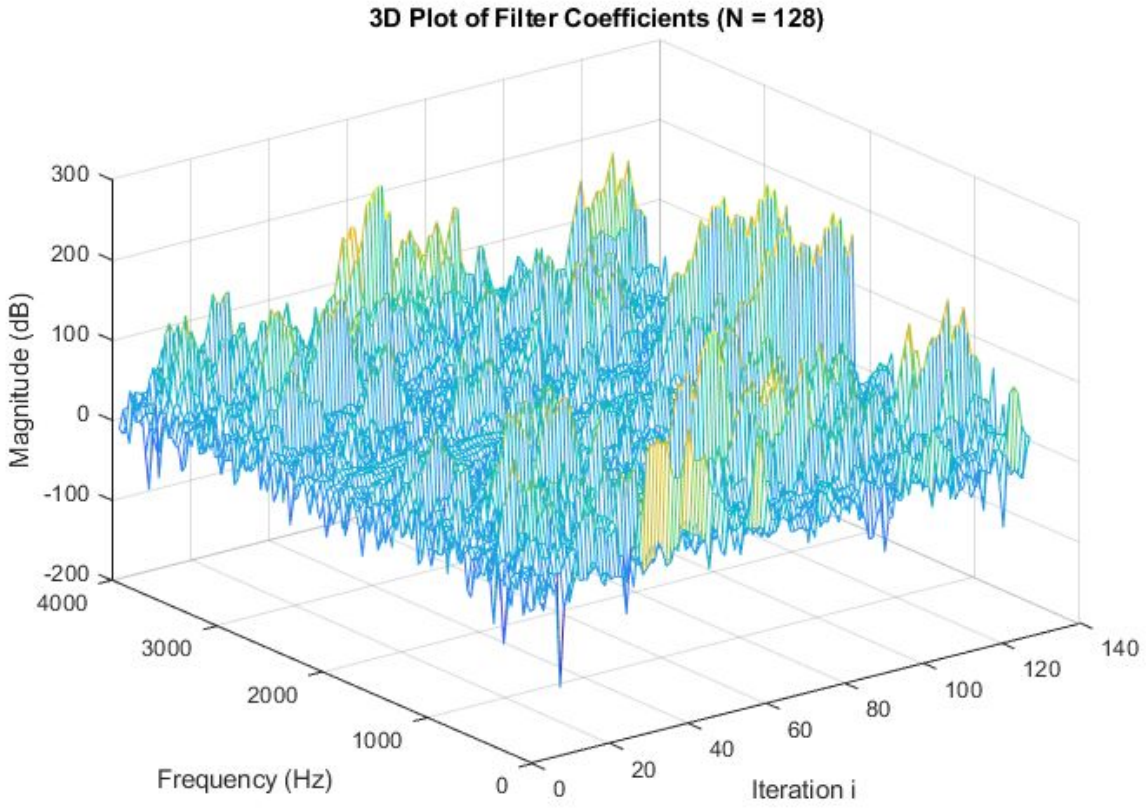
N = 128:


Error convergence plot
mu = 0.04
mu = 0.08
mu = 0.12
mu = 0.16
|E(i)|^2 (dB)
Iteration i
0
2000
4000
6000
8000
10000
12000
0
20
40
60
80
100
120
140

N = 256:



The following 3D plot shows how the filter coefficients change at different frequencies across each adaptive iteration:



Remarks

In the MATLAB program the variable B holds the filter coefficients. B(k) is a two-dimensional slice of the 3D mesh above taken at any frequency value (k).

The N-point FFT increases the computational complexity of the algorithm as N increases. When N is increased the spectral resolution improves but the order of the filter is held constant (some terms may be zero).

A larger value of mu increases the flexibility of the filter to react to each incoming data frame. However as the filter is made more flexible (more willing to adapt to each subsequent frame rather than resisting change) the stability of the filter comes into question. In general values of mu below mu = 0.15 lead to stable filters. However at a large value of N (such as N = 256) a smaller value of mu was required to avoid stability issues.

The filter coefficients after each subsequent frame were dependent on the FFT of each input signal frame. This frame-style signal segmentation allowed for parallel processing of data and improved computation time.

The computation time for the program is rather fast. I was able to run about 100 simulations in a manner of 30 seconds when determining optimal mu coefficients.

Appendix:

Main Code:

```
% Louis Rosenblum
% Project 2
% EEE 509 - ASU
% 06/10/2020

%% Initialization

%clear all
close all

cd 'C:\Users\Louis\Desktop\DSP\Project 2'

%% Set runtime constants

N = 128;
nFrames = 128;

if (N == 4)
    mu = 0.0007;
elseif (N == 16)
    mu = 0.0636;
elseif (N == 64)
    mu = 0.1115;
elseif (N == 128)
    mu = 0.058;
else (N == 256)
    mu = 0.0235;
end

%% Read audio from file

[mic1, Fs1] = audioread('mic1_2019.wav');
[mic2, Fs2] = audioread('mic2_2019.wav');

% Calculate starting SNR
snr1 = snr(mic1)

%% Create frame data structures
```

```
[mic1_frames, len1] = createFrames(mic1,nFrames);
[mic2_frames, len2] = createFrames(mic2,nFrames);

%% Perform N-point fft

mic1_fft = fft(mic1_frames,N);
mic2_fft = fft(mic2_frames,N);

%% Main adaptive filter algorithm

Es = [];
Bs = [];
es = [];
B = zeros(N,1);

k = 1:nFrames;
figure();

for i = 1:nFrames
    % Calculate for current frame
    diaganol = diag(mic1_fft(:,i));
    E = mic2_fft(:,i) - diaganol * B;
    e = ifft(E);

    Es = [Es E];
    Bs = [Bs B];
    es = [es e];

    % Prepare for next frame
    B = B + 2*mu * diaganol'*E;

    p = 10*log(1/N * E'*E);
    error = [error p]
end

title("Error convergence plot")

f = ones(1,nFrames);

for i = 1:nFrames
    f(i) = Fs1/2 / 128 *i;
end

figure()
mesh(k,f,20*log(abs(Bs)))
title("3D Plot of Filter Coefficients (N = 128)")
```

```
xlabel("Iteration i")
ylabel("Frequency (Hz)")
zlabel("Magnitude (dB)")


%% Revert frames to vector

final_signal = createVector(es);

% Calculate ending SNR
snr2 = snr(final_signal)

diff = snr2 - snr1

%% Create plots

% Plot original signal

t = 0:1/Fs1:(length(mic1)-1)/Fs1;
t2 = 0:1/Fs1:(length(final_signal)-1)/Fs1;

figure();
plot(t,mic1);
title("Input Signal")
xlabel("Time (s)")
ylabel("Amplitude")

% Plot filtered signal
figure();
plot(t2,final_signal);
title("Output Signal")
xlabel("Time (s)")
ylabel("Amplitude")
%% Function definitions

function [data, len] = createFrames(audio,nFrames)
    len = length(audio);
       frameSize = ceil(len/nFrames);
    total = frameSize*nFrames;
    z = total - len;
    pad = [audio;zeros(z,1)];
    data = reshape(pad,frameSize,nFrames);
end

function data = createVector(audio)
    [m,n] = size(audio);
    data = [];
```

```
    for i = 1:n
        data = [data rot90(audio(:,i))];
    end
end
```

Optimal-mu Calculator:

```
results = []
mus = []

for i = 1:100
    mu = rand(1,1) * 0.15;
    project2;
    results = [results diff];
    mus = [mus mu];
end

[M,I] = max(results);

Max_diff = M

Max_mu = mus(I)
```