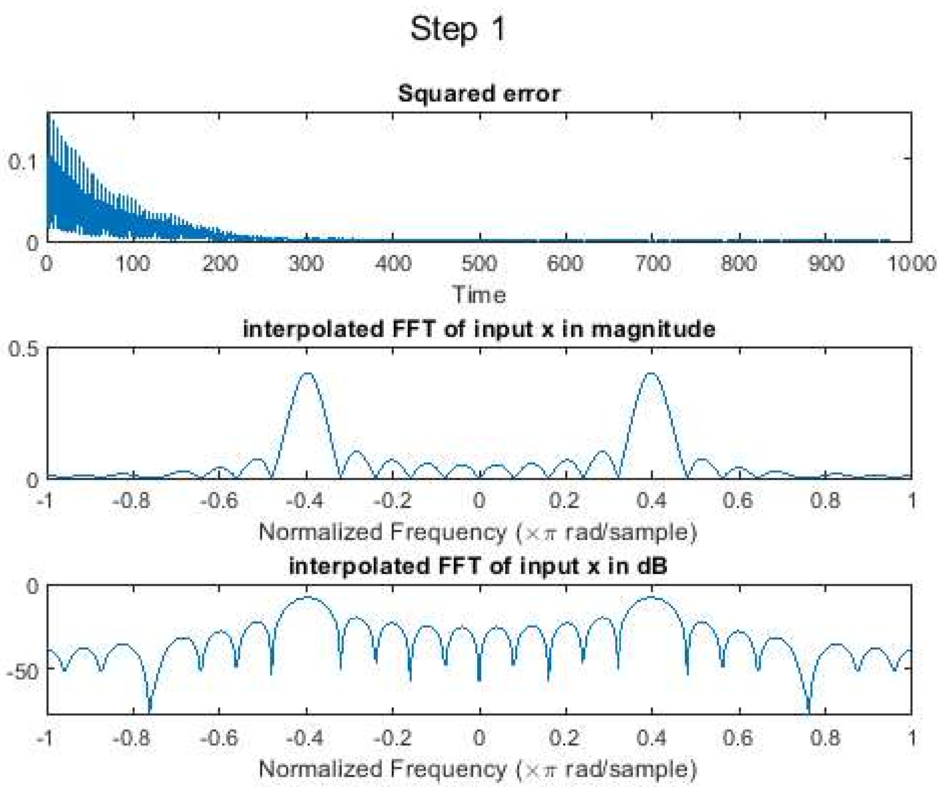
Contents


```
close all
clear
clc
```

Step 1

```
M = 25;
L = 1000;
mu = 0.001;
a = 0.1;

w      = zeros(M,1);
yout   = zeros(M,1);
errout = zeros(M,1);

x = cos(2*pi*0.2*(0:(L-1))) + cos( 2*pi*0.38*(0:(L-1)))+a*randn(1,L);
d = 0.4*cos(2*pi*0.2*(0:(L-1)) + pi/5);

for n = 1:L-M
    xn = x(n:n+M-1)';
    yn = w'*xn;
    yout(n) = yn;
    en = d(n) - yn;
    errout(n) = en;
    w = w + mu * en * xn;
end

N  = L;
fv = (0:(N-1)) - floor(N/2);
fvM0 = (0:(512-1)) - floor(512/2);
fv = 2*fv/N;
fvM0 = 2*fvM0/512;

figure()
sgtitle('Step 1')
subplot(3,1,1)
plot(errout.*errout);
xlabel('Time')
title('Squared error')
subplot(3,1,2)
plot(fvM0, abs(fftshift(fft(w,512))));
xlabel('Normalized Frequency (\times\pi rad/sample)')
title('interpolated FFT of input x in magnitude')
subplot(3,1,3)
plot(fvM0, mag2db(abs(fftshift(fft(w,512)))));
xlabel('Normalized Frequency (\times\pi rad/sample)')
title('interpolated FFT of input x in dB')

%{
```

```
Plot the interpolated fft of the filter w both in magnitude and dB.
Compare this result to the least squares filter result from the prelab.

The adaptive filter has 2 spikes while the squares one doesn't. The FFT is
sesen in dB as there is a difference in distancce of -80 vs -60dB.
%}
```



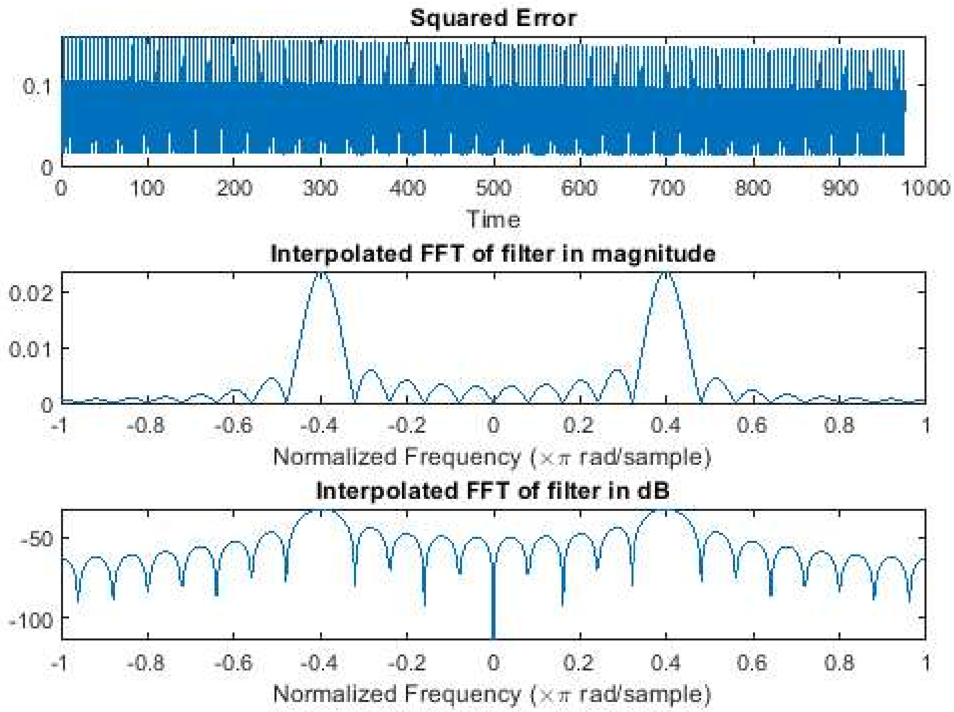
Step 2

```
i = 0;
for mu = [0.00001 0.0001 0.01 0.1 0.001 0.0001]
    M = 25;
    L = 1000;
    a = 0.1;

    if i == 4
        a = 0.5;
    end

    if i == 5
        L = 5000;
        a = 0.1;
    end

    w      = zeros(M,1);
    yout   = zeros(M,1);
    errout = zeros(M,1);

    x = cos(2*pi*0.2*(0:(L-1))) + cos( 2*pi*0.38*(0:(L-1)))+a*randn(1,L);  % this is x
    d = 0.4*cos(2*pi*0.2*(0:(L-1)) + pi/5);    % this is d

    for n = 1:L-M
```

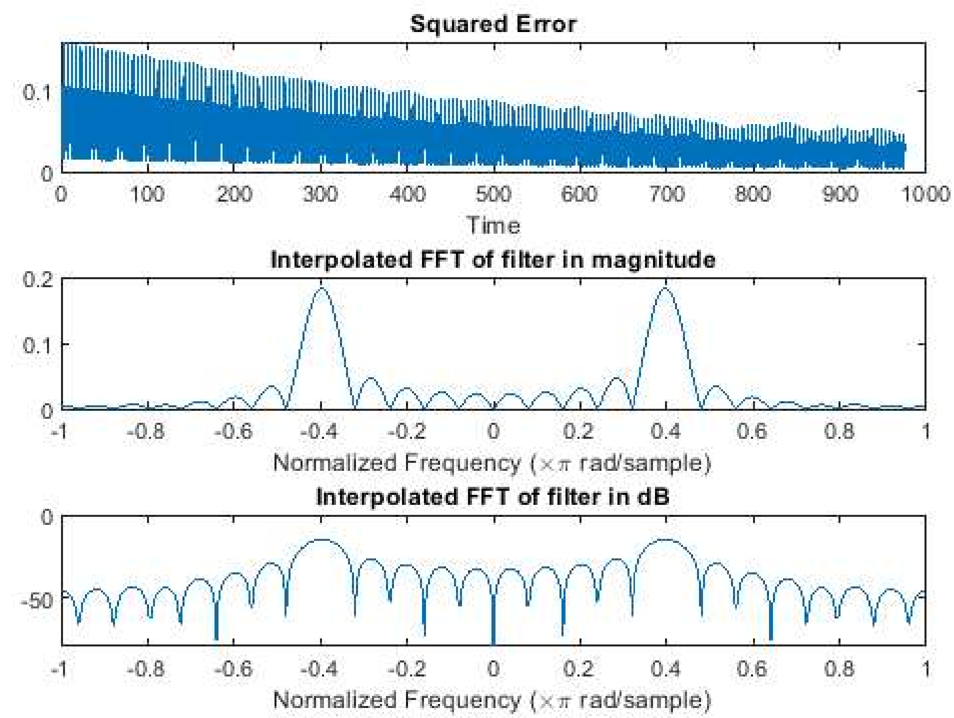
```
        xn = x(n:n+M-1)';
        yn = w'*xn;
        yout(n) = yn;
        en = d(n) - yn;
        errout(n) = en;
        w = w + mu * en * xn;
    end

    N  = L;
    fv = (0:(N-1)) - floor(N/2);
    fv = 2*fv/N;
    fvM0 = (0:(512-1)) - floor(512/2);
    fvM0 = 2*fvM0/512;

    figure()
    sgtitle("Step 2 : mu = " + mu + ", a = " + a + ", L = " + L)
    subplot(3,1,1)
    plot(errout.*errout);
    xlabel('Time')
    title('Squared Error')
    subplot(3,1,2)
    plot(fvM0, abs(fftshift(fft(w,512))));
    xlabel('Normalized Frequency (\times\pi rad/sample)')
    title('Interpolated FFT of filter in magnitude')
    subplot(3,1,3)
    plot(fvM0, mag2db(abs(fftshift(fft(w,512)))));
    xlabel('Normalized Frequency (\times\pi rad/sample)')
    title('Interpolated FFT of filter in dB')
    i = i+1;
end

%{
0.00001:-----------------------------------------
What is the lowest value of the error? If the error goes to close to zero, how many iterations does it take?
0.144

How much does the shape of the fft of the filter look like the least squares filter fft from the prelab?
Very similar.

What is the maximum value of the fft magnitude? How does that compare to the least squares fft from the
prelab?
0.19 which is less but not completely different.

0.01:------------------------------------------------------------
What is the lowest value of the error? If the error goes to close to zero, how many iterations does it take?
0 after 70ish iterations

How much does the shape of the fft of the filter look like the least squares filter fft from the prelab?
Very similar.

What is the maximum value of the fft magnitude? How does that compare to the least squares fft from the
prelab?
0.4 which is the same from that of the prelab

0.1:
What is the lowest value of the error? If the error goes to close to zero, how many iterations does it take?
0 after only 1

How much does the shape of the fft of the filter look like the least squares filter fft from the prelab?
Not similar.
```

```
What is the maximum value of the fft magnitude? How does that compare to the least squares fft from the
prelab?
HUGEEEEE. Way bigger than that of the prelab like 6.5*10^80

Change noise level from 0.1 to 0.5
How do these results compare to Step 1 results? Why do you think the squared error curve is different?
The noise probably makes it harder for the filter to keep the error at zero
and therefore the noise limits the filter's efficacy in this scenario.

Change input signal duration from 1000 to 5000 samples
Compare the results to the result earlier in this step which had the same value of mu and a, but had L =
1000.
The error gests to zero but takes A LOT of iterations until that happens.

%}
```

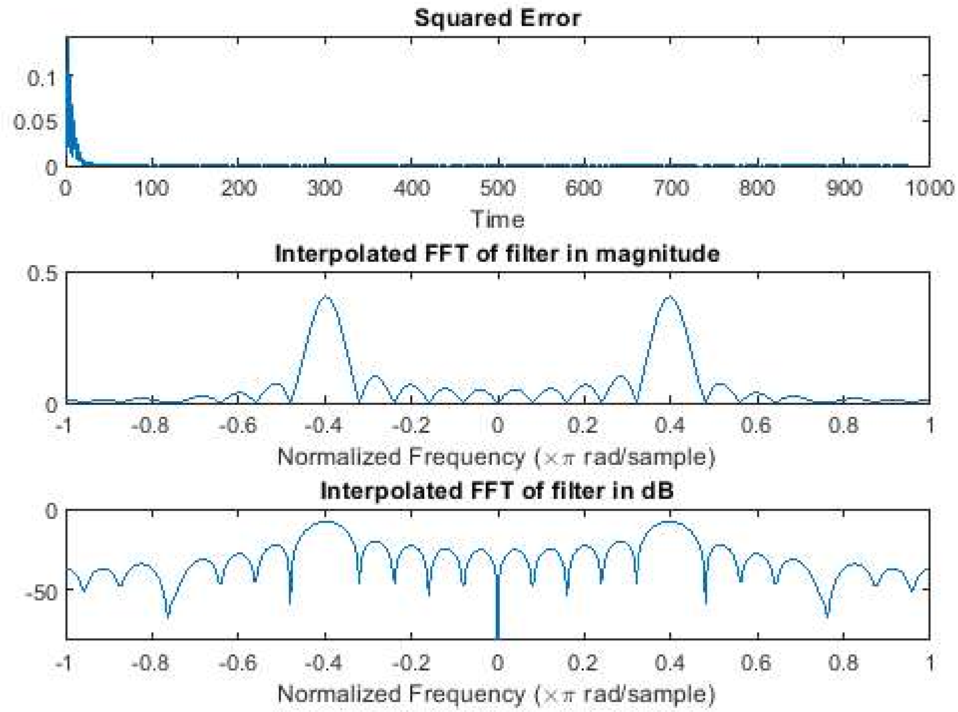
Step 2 : mu = 1e-05, a = 0.1, L = 1000



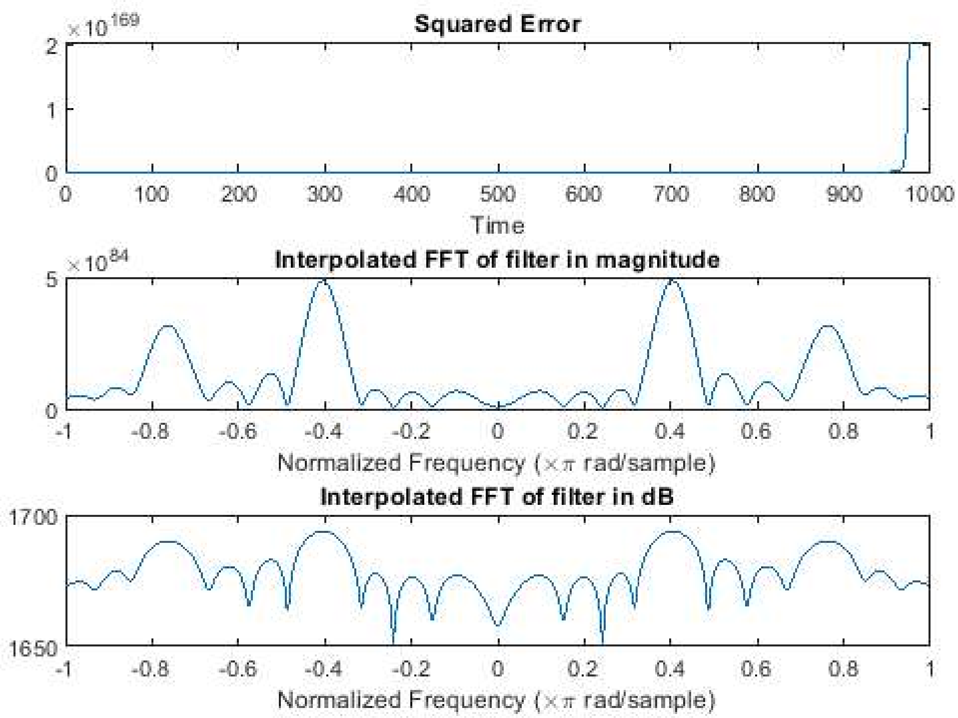
Step 2 : mu = 0.0001, a = 0.1, L = 1000

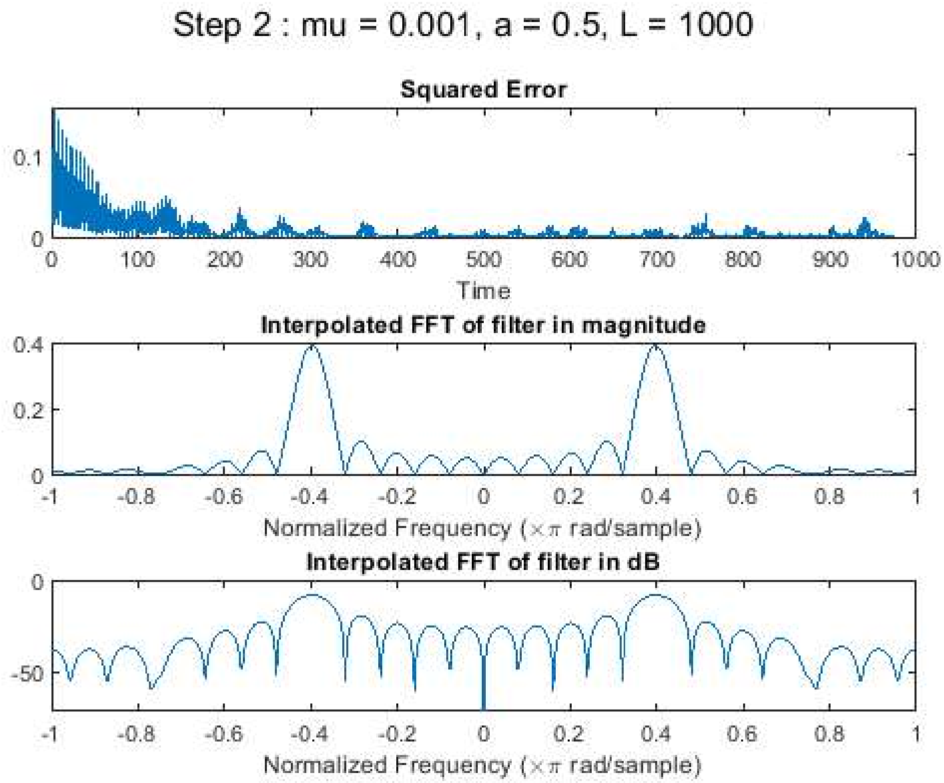


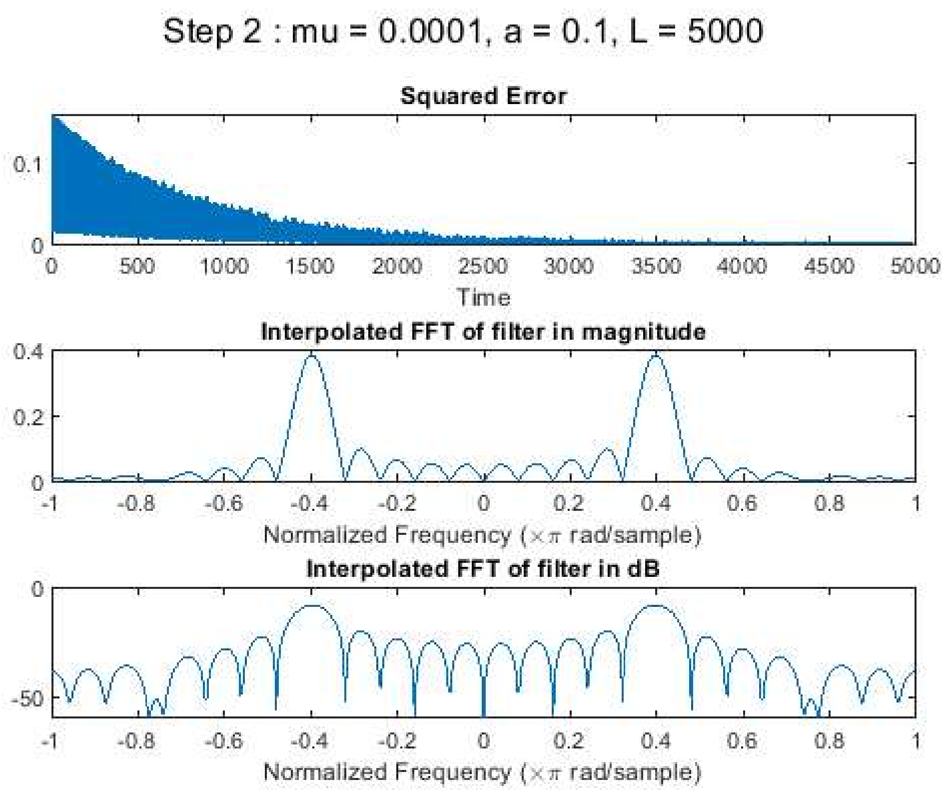
Step 2 : mu = 0.01, a = 0.1, L = 1000



Step 2 : mu = 0.1, a = 0.1, L = 1000







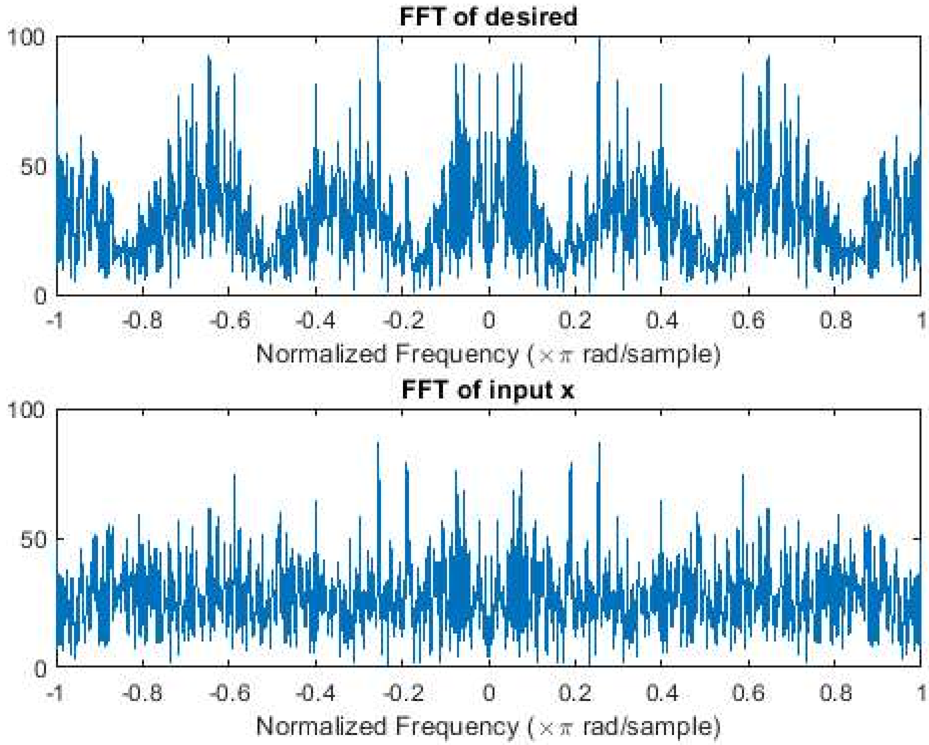
Step 3

```
M  = 25;
L  = 1000;
a  = 0.1;
```

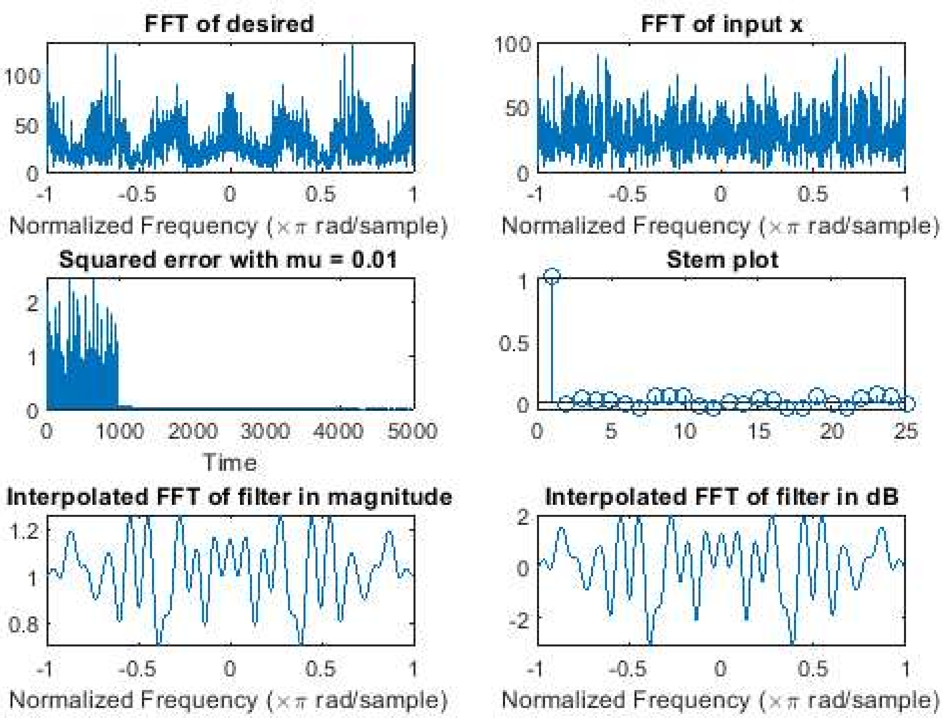
```
i  = 0;
N  = L;

fv = (0:(N-1)) - floor(N/2);
fv = 2*fv/N;

fvM0 = (0:(512-1)) - floor(512/2);
fvM0 = 2*fvM0/512;

for mu = [0.001 0.01 0.05 0.08 0.0001 0.01]
    x = randn(1,L);  % this is x
    h = [1 0 0 0 0 0 0.5];
    xUnk = conv(h,x);
    d = xUnk(1:end - length(h)+1);

    if i == 5
        d = xUnk(length(h):end);
    end

    for n = 1 : L-M
        xn = x(n:n+M-1)';
        yn = w'*xn;
        yout(n) = yn;
        en = d(n) - yn;
        errout(n) = en;
        w = w + mu * en * xn;
    end

    if i == 0
        figure()
        sgtitle("Step 3 : mu = " + mu + ", a = " + a + ", L = " + L)
        subplot(2,1,1)
        plot(fv, abs(fftshift(fft(d,L))));
        xlabel('Normalized Frequency (\times\pi rad/sample)')
        title('FFT of desired')

        subplot(2,1,2)
        plot(fv, abs(fftshift(fft(x,L))));
        xlabel('Normalized Frequency (\times\pi rad/sample)')
         title('FFT of input x')

    else
        figure()
        sgtitle("Step 3 : mu = " + mu + ", a = " + a + ", L = " + L)
        subplot(3,2,1)
        plot(fv, abs(fftshift(fft(d))));
        xlabel('Normalized Frequency (\times\pi rad/sample)')
        title('FFT of desired')

        subplot(3,2,2)
        plot(fv, abs(fftshift(fft(x))));
        xlabel('Normalized Frequency (\times\pi rad/sample)')
        title('FFT of input x')

        subplot(3,2,3)
        plot(errout.*errout);
        xlabel('Time')
        title('Squared error with mu = 0.01' )

        subplot(3,2,4)
        stem(w);
```

```
        title('Stem plot')

        subplot(3,2,5)
        plot(fvM0, abs(fftshift(fft(w,512))));
        xlabel('Normalized Frequency (\times\pi rad/sample)')
        title('Interpolated FFT of filter in magnitude')

        subplot(3,2,6)
        plot(fvM0, mag2db(abs(fftshift(fft(w,512)))));
        xlabel('Normalized Frequency (\times\pi rad/sample)')
        title('Interpolated FFT of filter in dB')
    end
    i = i + 1;
end
%{
Compare the filter w with the unknown filter h. How similar is it?
What is the error level at the end of the adaptation loop?
Not simlar. Almost 0 at the end of  the adaptation loop.


Rerun the filter with mu = 0.01 and compare.
Describe the results you obtain with mu = 0.05, 0.08, and 0.0001.
Explore the importance of timing alignment: Describe the output and explain.
Not simlar. Almost 1 at the end of  the adaptation loop.

0.05 -> mu=0.01

0.08 -> mu=0.01

0.0001 -> mu=0.001

The output has an too much error with a distorted FFT
%}
```

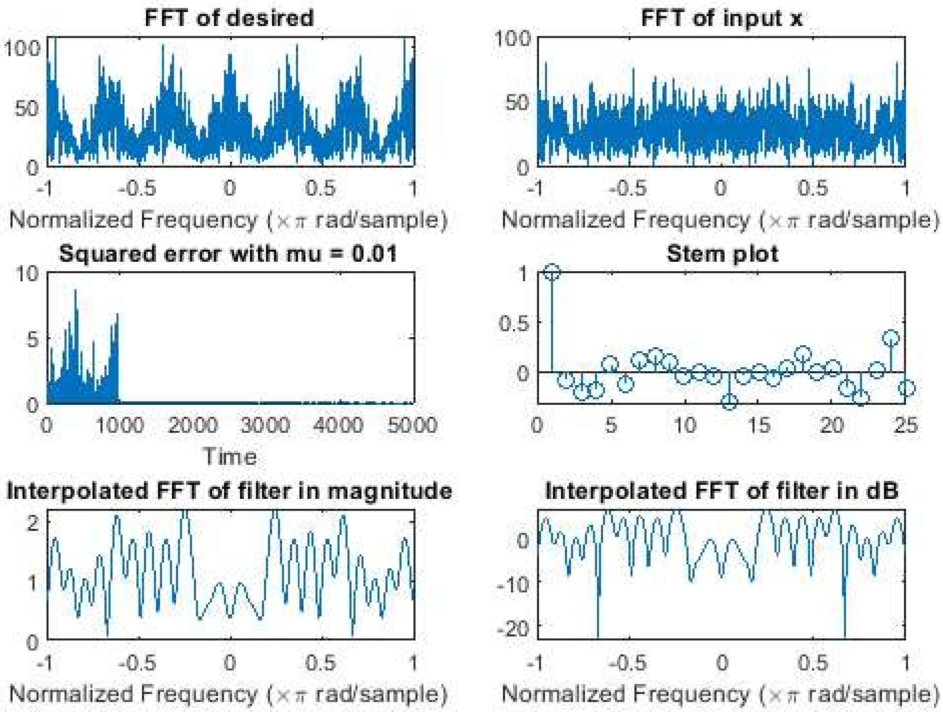
Step 3 : mu = 0.001, a = 0.1, L = 1000



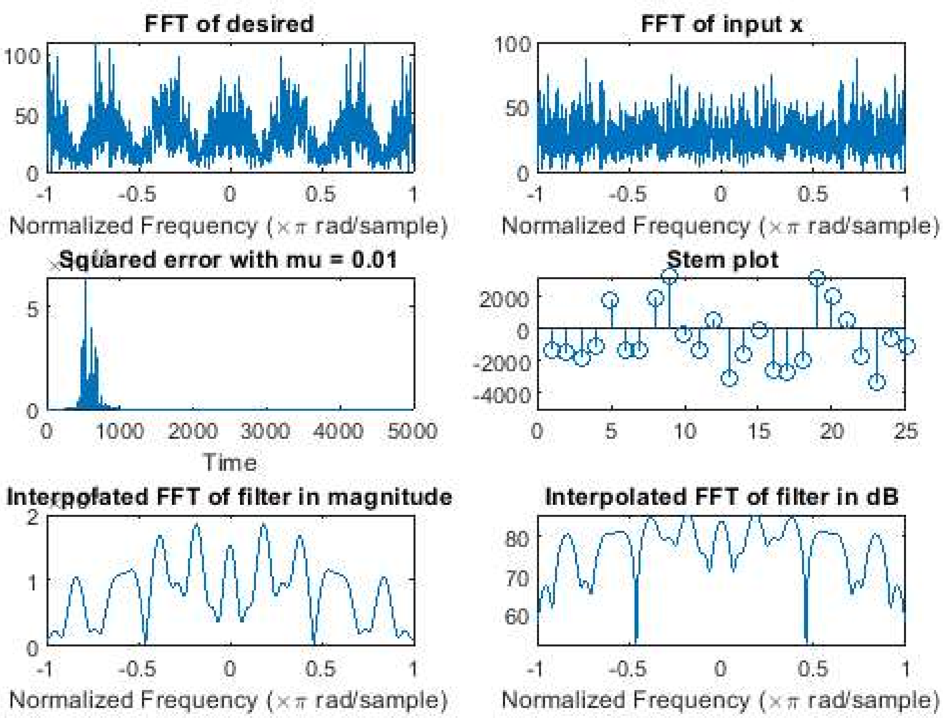
Step 3 : mu = 0.01, a = 0.1, L = 1000

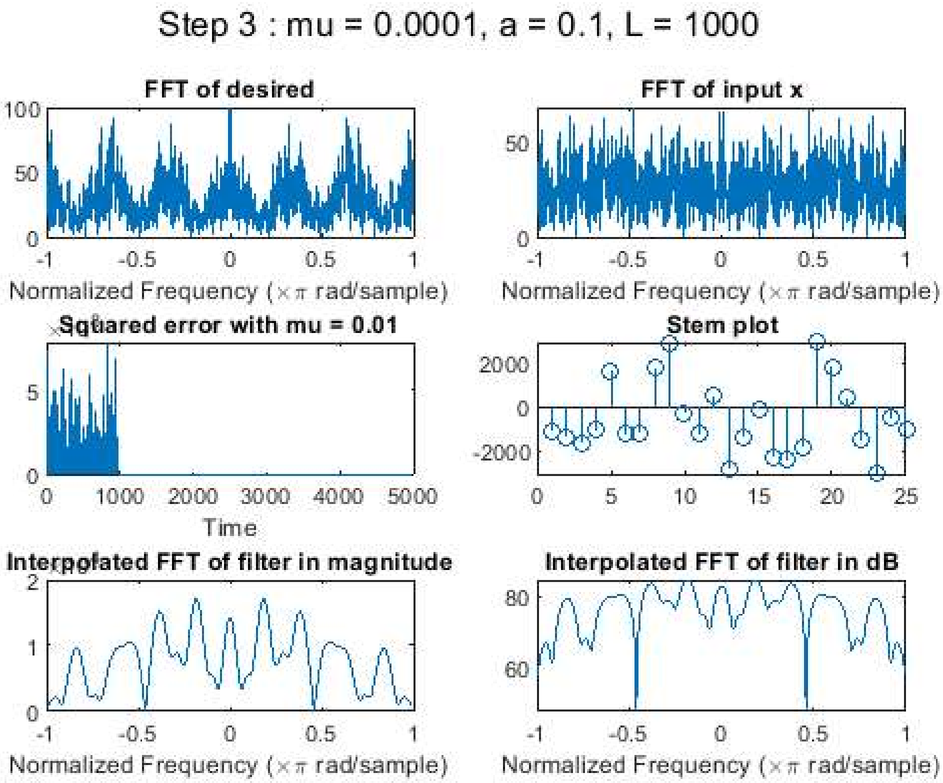


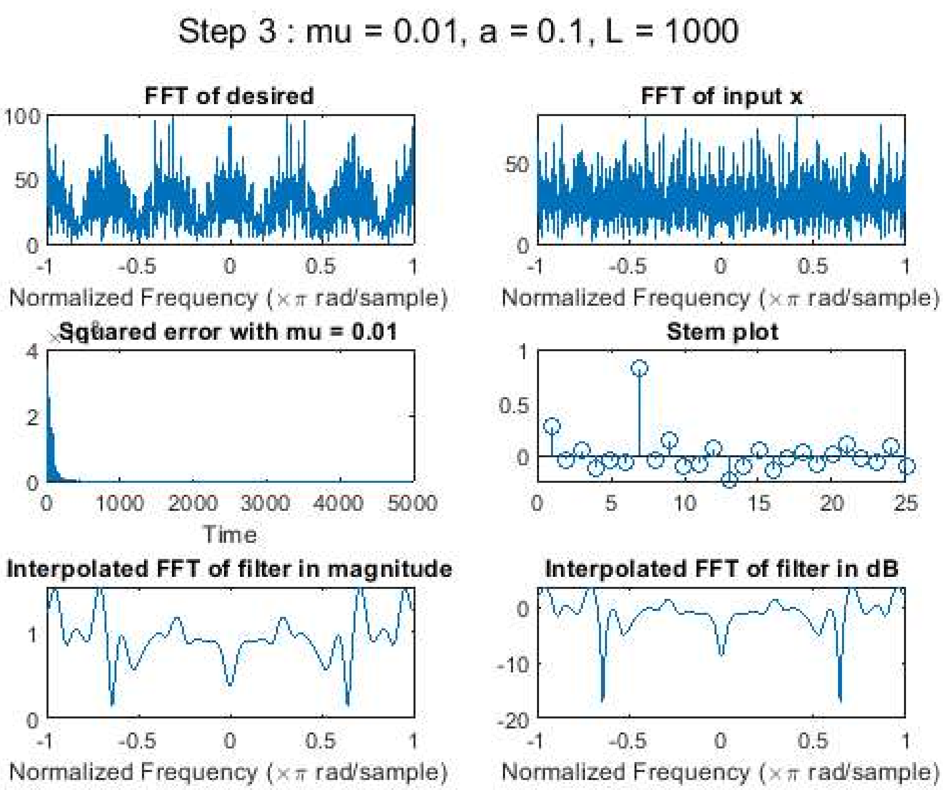
Step 3 : mu = 0.05, a = 0.1, L = 1000



Step 3 : mu = 0.08, a = 0.1, L = 1000





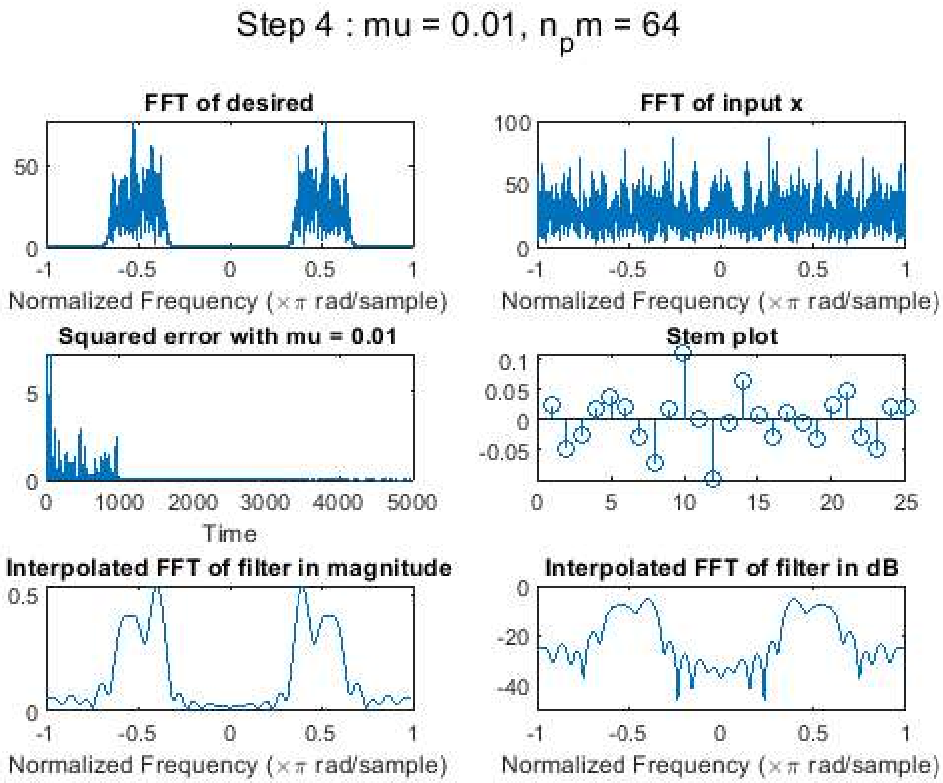


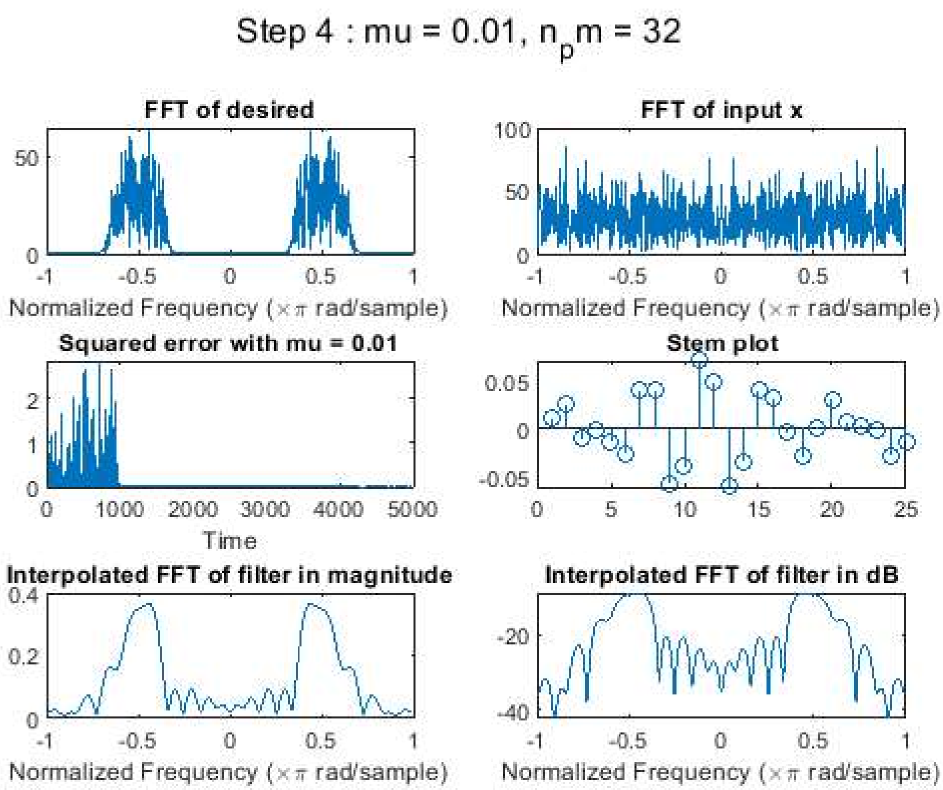
Step 4

```
M  = 25;
L  = 1000;
a  = 0.1;
```

```matlab
i  = 0;
mu = 0.01;
N  = L;
fv = (0:(N-1)) - floor(N/2);
fv = 2*fv/N;
fvM0 = (0:(512-1)) - floor(512/2);
fvM0 = 2*fvM0/512;

for n = 1:2
    f_pm = [0, 1200, 1600, 2400, 2800, 4000]/4000;
    a_pm = [0,0,1,1,0,0]; % band pass
    n_pm = 64;
    b_pm = firpm(n_pm, f_pm, a_pm);
    x = randn(1,L); % create x input for both filters
    x_unk = conv(b_pm, x);
    d = x_unk(1: end- length(b_pm)+1); % make d the same length as x

    if n == 2
        n_pm = 32;
    end

    h = [1 0 0 0 0 0 0.5];
    xUnk = conv(h,x);

    for n = 1 : L-M
        xn = x(n:n+M-1)';
        yn = w'*xn;
        yout(n) = yn;
        en = d(n) - yn;
        errout(n) = en;
        w = w + mu * en * xn;
    end

    figure()
    sgtitle("Step 4 : mu = " + mu + ", n_pm = " + n_pm)
    subplot(3,2,1)
    plot(fv, abs(fftshift(fft(d))));
    xlabel('Normalized Frequency (\times\pi rad/sample)')
    title('FFT of desired')

    subplot(3,2,2)
    plot(fv, abs(fftshift(fft(x))));
    xlabel('Normalized Frequency (\times\pi rad/sample)')
    title('FFT of input x')

    subplot(3,2,3)
    plot(errout.*errout);
    xlabel('Time')
    title('Squared error with mu = 0.01' )

    subplot(3,2,4)
    stem(w);
    title('Stem plot')

    subplot(3,2,5)
    plot(fvM0, abs(fftshift(fft(w,512))));
    xlabel('Normalized Frequency (\times\pi rad/sample)')
    title('Interpolated FFT of filter in magnitude')

    subplot(3,2,6)
    plot(fvM0, mag2db(abs(fftshift(fft(w,512)))));
```

```
    xlabel('Normalized Frequency (\times\pi rad/sample)')
    title('Interpolated FFT of filter in dB')

    i = i + 1;
end

%{
Does the error get reduced as the filter adapts? Try a few other values for mu and see if there is any
improvement. Why is this filter not working well?
No, the error doesn't get reduced as the filter adapts. The filter has too
high of an order to maintain stability regardless of mu change.

Explain why this change of n_pm made an improvement compared to the previous result. What other
change might have improved the result?
As I kmentioned earlier, the change in mu doesn't help much because of how
high our previour order was so lowering the order allows the change in mu
to have more of an impaact on the efficacy of our filter.

%}
```





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