Digital Signal Processing Lab

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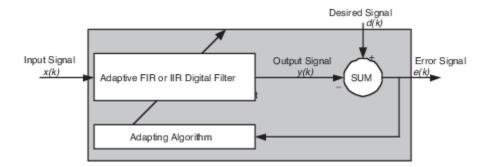
Roll No: 1401010

Lab Report - 1

Theory:-

Adaptive Filtering

Adaptive filtering involves the changing of filter parameters (coefficients) over time, to adapt to changing signal characteristics. An adaptive filter self-adjusts the filter coefficients according to an adaptive algorithm. One such algorithm is LMS Algorithm. We have implemented LMS Algorithm in MATLAB in this lab.



(Diagram for generic Adaptive Filter provided by The MathWorks, Inc. in MATLAB Documentation)

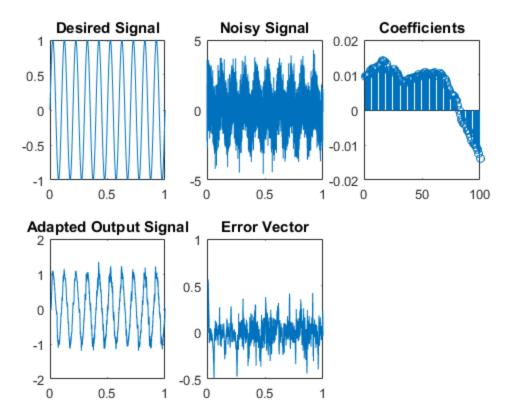
Lab Work:-

Lab - 0

1).LMS Test:-

```
%Lab_1_1 LMS Test clear;
```

```
clc;
% Time specifications:
Fs = 10300;
                             % samples per second or Sampling frequency
dt = 1/Fs;
                            % seconds per sample
                            % seconds
stop = 1;
t = 0:dt:stop;
                             % seconds
% Sine wave:
Fc = 10; % hertz
d = sin(2*pi*Fc*t);
x = d + randn(1, Fs+1);
N=100;
                           %Number of Coefficients
delta=0.001;
[h, Y, E] = LMS(x,d,delta,N);
figure;
subplot(2,3,1);
plot(t,d);
title('Desired Signal');
%xtitle('Time');
%ytitle('Signal Value(d)');
subplot(2,3,2);
plot(t,x);
title('Noisy Signal');
%xtitle('Time');
%ytitle('Signal Value(x)');
subplot(2,3,3);
stem(h);
title('Coefficients');
%ytitle('h');
subplot(2,3,4);
plot(t,Y);
title('Adapted Output Signal');
%xtitle('Time');
%ytitle('Signal Value(Y)');
subplot(2,3,5);
plot(t,E);
title('Error Vector');
%ytitle('E');
```

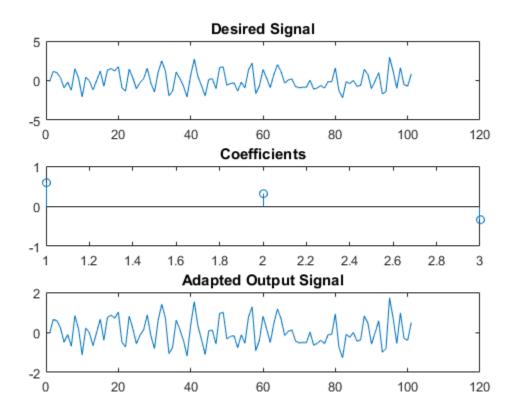


2).FIR Filter:-

```
subplot(3,1,1);
plot(d(900:1000));
title('Desired Signal');

subplot(3,1,2);
stem(h);
title('Coefficients');

subplot(3,1,3);
plot(Y(900:1000));
title('Adapted Output Signal');
```



3).LMS Algorithm:-

```
%Linear Mean Square Algorithm (LMS Algorithm)

function [h,Y,E] = LMS(x,d,delta,N)

M=length(x);
Y=zeros(1,M);
h=zeros(1,N);
E=zeros(1,M);

for n=N:M

    x1=x(n:-1:n-N+1);
    Y(n)=h*x1';
    e=d(n)-Y(n);
    h=h+(delta*e*x1);
    E(n)=e;

end
end
```

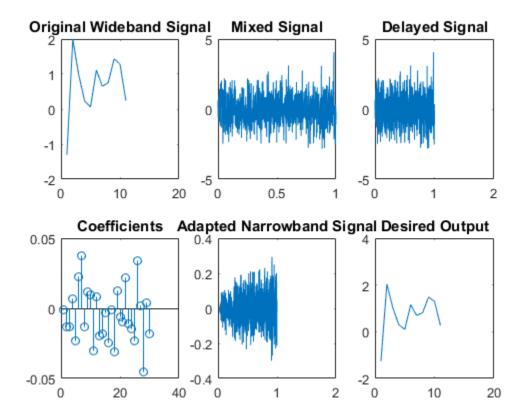

Test - 1

1).<u>Test-1</u>:-

```
t_new = 0:dt:stop+dt;
                      % Seconds
% Sine wave:
Fc = 10;
                           % hertz
d = 5*sin(2*pi*Fc*t);
abs d=sqrt(d*d');
                           %Normalizing d vector
d_m=d./abs_d;
w=randn(1,Fs+1);
x = d m + w;
N=30;
                            %Number of Coefficients
D=1;
                           %Delay Parameter
delay x=Delay(x,D);
                           %Delayed X
x new=[x zeros(1,D)];
delta=0.001;
                           %Step Parameter
[h, Y, E] = LMS(delay_x,x_new,delta,N);
figure;
subplot(2,3,1);
plot(w(30:40));
title('Original Wideband Signal');
%xtitle('Time');
%ytitle('Signal Value(d)');
subplot(2,3,2);
plot(t,x);
title('Mixed Signal');
%xtitle('Time');
%ytitle('Signal Value(x)');
subplot(2,3,3);
plot(t_new,delay_x);
title('Delayed Signal');
%xtitle('Time');
%ytitle('Signal Value(x)');
subplot(2,3,4);
stem(h);
title('Coefficients');
%ytitle('h');
subplot(2,3,5);
plot(t_new,Y);
title('Adapted Narrowband Signal');
```

```
%xtitle('Time');
%ytitle('Signal Value(Y)');

subplot(2,3,6);
plot(E(30:40));
title('Desired Output');
%ytitle('E');
```



2).LMS Algorithm:-

```
%Linear Mean Square Algorithm (LMS Algorithm)

function [h,Y,E] = LMS(x,d,delta,N)

M=length(x);
Y=zeros(1,M);
h=zeros(1,N);
E=zeros(1,M);

for n=N:M
```

```
x1=x(n:-1:n-N+1);
Y(n)=h*x1';
e=d(n)-Y(n);
h=h+(delta*e*x1);
E(n)=e;
end
```

3). Delay Function:-