School of Engineering and Applied Science (SEAS)
Ahmedabad University

BTech(ICT) Semester VI:Digital Signal Processing

Laboratory Assignment-6

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AIM :: Lab6 helps to solve the properties of signal like LTI Systems, DTFT-IDTFT, Sampling Theorem, Digital Frequency signal in MATLAB .**The filter's impulse response is a sinc function in the time domain, and its frequency response is a rectangular function**

1. Solution Problem-1

(a) Matlab Script:

```
clc;
close all ;

fC=1/6;                    %cut-off freq
fL=1/9;                    %lower edge cut-off freq for Band-Pass and Band-Stop
    Filter
fH=1/3;                    %higher edge cut-off freq for Band-Pass and Band-Stop
    Filter
n=linspace(-20, 20);     %given interval in Question

Hn_LowPass=2*fC*sinc(2*fC*n);             %impulse response of low pass filter
Hn_HighPass=sinc(n)-2*fC*sinc(2*fC*n);    %impulse response of high pass filter
Hn_BandPass=(2*fH*sinc(2*fH*n))-(2*fL*sinc(2*fL*n));    %impulse response of Band
    Pass filter
Hn_BandStop=(2*fL*sinc(2*fL*n))-(2*fH*sinc(2*fH*n));    %impulse response of Band
    Stop filter

sgtitle('Impluse response of filters ')                 %main title for figure

subplot(2,2,1)
plot(n, Hn_LowPass);
grid on;
xlabel('n');
ylabel('H(n)');
title("Low Pass Filter");

subplot(2,2,3)
plot(n, Hn_HighPass);
grid on;
xlabel('n');
ylabel('H(n)');
title("High Pass Filter");

subplot(2,2,2)
plot(n, Hn_BandPass);
grid on;
xlabel('n');
ylabel('H(n)');
title("Band Pass Filter");

subplot(2,2,4)
plot(n, Hn_BandStop);
grid on;
xlabel('n');
ylabel('H(n)');
title("Band Stop Filter");
```

(b) Approach:
In question, we've been given rectangular function of Low Pass, High Pass, Band Pass and Band Stop filter,
which is **frequency response H(w)** of that filter so it's **Rectangular function below :**
f_C= cut-off frequency , f_L=lower band edge and f_H=upper band edge

$H(w) = rect(\frac{w}{2w_C})$
$H(f) = rect(\frac{f}{2f_C})$
$\therefore$H(f)=1 ,for -$f_C \leq f \leq f_C$
$so, -w_C \leq w \leq w_C$

Next,**impulse response H(n)** of that filter which is **Inverse Fourier Transform of frequency response H(w)** :

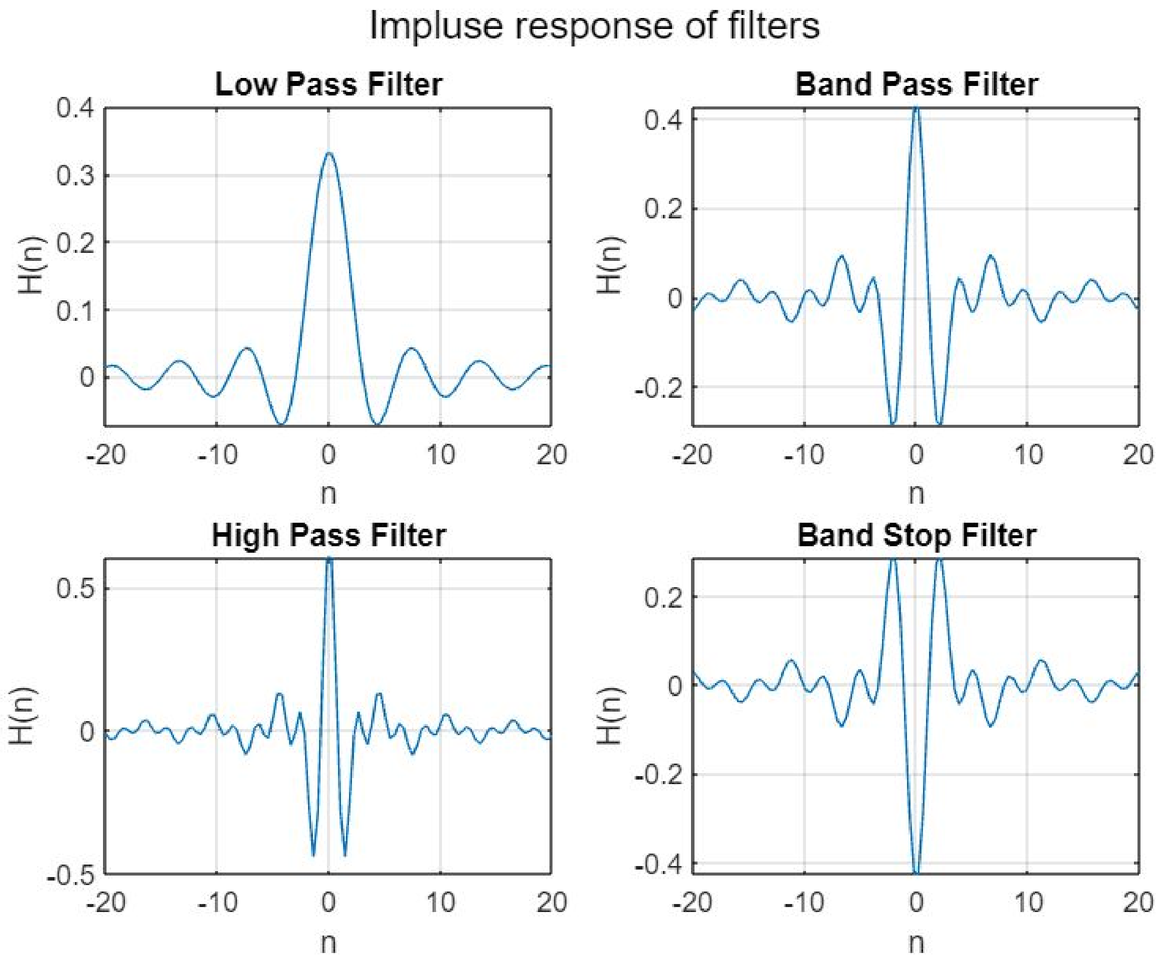
$H(n) = \mathcal{F}^{-1}(H(f))$
$\therefore$H(n)=$\int_{-f_C}^{f_C} e^{j2\pi fn} df$

Mentioned in question to adjust H(n) into sinc function ,
$\therefore$H(n)= $\frac{e^{j2\pi fn}}{j2\pi n} |_{-f_C}^{f_C} = \frac{1}{\pi n}\frac{e^{j2\pi f_C n} - e^{-j2\pi f_C n}}{2j} = \frac{1}{\pi n} sin(2\pi f_C n) = 2f_C sinc(2f_C n)$

Using this $H_{LP}(n) = 2f_C sinc(2f_C n)$ function plotted the graph for Low pass filter. Inverting impulse response or subtract from sinc function of low pass we get High Pass filter's impulse response, $H_{HP}(n) = sinc(n) - 2f_C sinc(2f_C n)$

Band Pass filter with lower band edge f_L and upper band edge f_H is just the difference of two such sinc filters if phase is zero, $H_{BP}(n) = 2f_H sinc(2f_H n) - 2f_L sinc(2f_L n)$. and Inverting impulse response of Band Pass filter it gives Band Stop filter's impulse response, $H_{BS}(n) = 2f_L sinc(2f_L n) - 2f_H sinc(2f_H n)$.

(c) Simulation Output:



2. **Solution Problem-2**

   (a) Matlab Script:

```
clc;
close all ;

fC=1/12;                                                    %cut-off frequency for filter
%if using fc1=1/20 for LP filter and fc2=1/8 for HP filter impulse response it gives same output because here i'm using common fc=1/12 which pass low freq in low pass filter(1/20) and passes high freq in high pass filter(1/8)

x_lower_index=input('Enter the lower index signal :');
x_upper_index=input('Enter the upper index signal :');
dn=0.01;
xn=x_lower_index:dn:x_upper_index;                          %time range of input signal

%initializing array as zeros
Hn_LowPass=zeros(1,length(xn));
Hn_HighPass=zeros(1,length(xn));
Xn=zeros(1,length(xn));

%assigning value of signal to array in interval
for i=1:length(xn)
    Hn_LowPass(i)=2*fC*sinc(2*fC*xn(i));                    %ideal low pass filter impulse response
    Hn_HighPass(i)=sinc(xn(i))-2*fC*sinc(2*fC*xn(i));            %ideal high pass filter impulse response
    Xn(i)=(10*cos(2*pi*xn(i)/20))+(5*cos(2*pi*xn(i)/8));%sinusoidal input signal
end
```

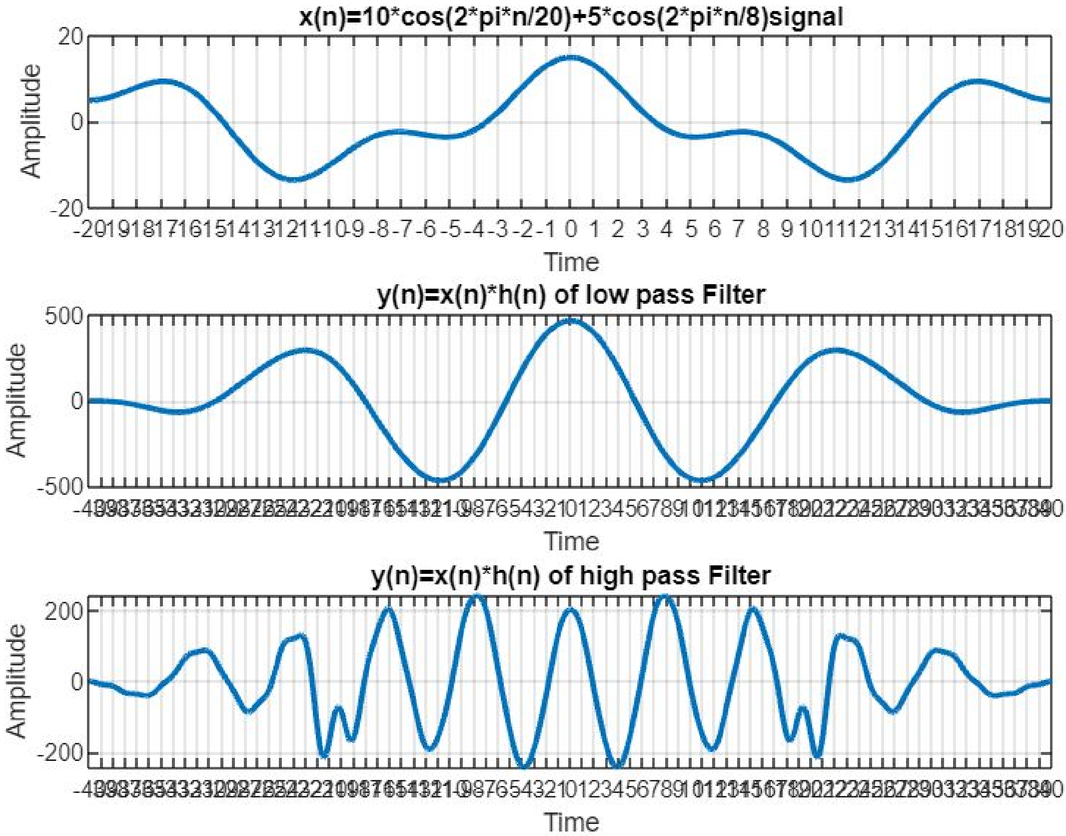
```
y_lower_index=2*x_lower_index;                %lower index of convolution as y
y_upper_index=2*x_upper_index;                %upper index of convolution as y

subplot(3,1,1)
plot(xn,Xn,'linewidth' , 2);                  %x(n) input signal
hold on;
grid on;
title('x(n)=10*cos(2*pi*n/20)+5*cos(2*pi*n/8)signal');
xlabel('Time');
ylabel('Amplitude');
xticks(x_lower_index:1:x_upper_index);

%low pass Filter----------------------------------------------------------
m=[];
y=[];
yn=y_lower_index:dn:y_upper_index;            %time range of convo output signal

%Matrix creation
%h_lowpass*x(i)
for i=1:length(Xn)
    g=Hn_LowPass.*Xn(i);
    m=[m;g];%this sequence store array at new row so one matrix will be formed
end
%summation of diagonal(right) elements an store it into one array
%11 12 13 14
%21 22 23 24
%31 32 33 34
%r=3,c=4,k=7,diagonal_index_sum=2(1+1 first element) because max we have sum=k
[r c]=size(m);
k=r+c;
diagonal_index_sum=2;
element=0;
%column and row wise searching for sum=diagonal_index_sum
while(diagonal_index_sum<=k)
    for i=1:r
        for j=1:c
            if((i+j)==diagonal_index_sum)
               element=element+m(i,j);
            end
        end
    end
    diagonal_index_sum=diagonal_index_sum+1;            %for next diagonal
    y=[y element];
    element=0;
end
disp(y);
subplot(3,1,2)
plot(yn,y, 'linewidth' , 2);
hold on;
grid on;
title('y(n)=x(n)*h(n) of low pass Filter');
xlabel('Time');
ylabel('Amplitude');
xticks(y_lower_index:1:y_upper_index);


%high pass Filter----------------------------------------------------------
m=[];
y=[];
yn=y_lower_index:dn:y_upper_index;            %time range of convo output signal

%Matrix creation
%h_highpass*x(i)
for i=1:length(Xn)
    g=Hn_HighPass.*Xn(i);
    m=[m;g];%this sequence store array at new row so one matrix will be formed
end
%summation of diagonal(right) elements an store it into one array
%11 12 13 14
```

```
%21 22 23 24
%31 32 33 34
%r=3,c=4,k=7,diagonal_index_sum=2(1+1 first element) because max we have sum=k
[r c]=size(m);
k=r+c;
diagonal_index_sum=2;
element=0;
%column and row wise searching for sum=diagonal_index_sum
while(diagonal_index_sum<=k)
    for i=1:r
        for j=1:c
            if((i+j)==diagonal_index_sum)
               element=element+m(i,j);
            end
        end
    end
    diagonal_index_sum=diagonal_index_sum+1;            %for next diagonal
    y=[y element];
    element=0;
end
disp(y);
subplot(3,1,3)
plot(yn,y, 'linewidth' , 2);
hold on;
grid on;
title('y(n)=x(n)*h(n) of high pass Filter');
xlabel('Time');
ylabel('Amplitude');
xticks(y_lower_index:1:y_upper_index);
```

(b) Approach:
-Using Ideal Low pass and band pass impulse response $H_{LP}(n) = 2f_C sinc(2f_C n)$ function plotted the graph for Low pass filter. Inverting impulse response of low pass we get High Pass filter's impulse response, $H_{HP}(n) = sinc(n) - 2f_C sinc(2f_C n)$
-With cut-off frequency =1/12 Taking input range from user as -n to n, computing signal using range of interval.Using this discrete signal first $HLowPass(n)\ and\ x(n)$, calculated length of convolution array and multiplying one by one element of x(n) with whole array h(n) and storing into one matrix m.Then,I did addition of right diagonal from first element $(m_{1X1}, where\ diagonalIndexSum = 1 + 1 = 2)$to last element (m_{rXc}) .Sum of those element which satisfies diagonal index sum (2,3,4,..,r+c).
-Same as above for $HHighPass(n)\ and\ x(n)$,After that plotted the value of y in appropriate range.

if using fc1=1/20 for low pass filter and fc2=1/8 for high pass filter impulse response it gives same output because here i'm using common fc=1/12 which pass low freq in low pass filter(1/20) and passes high freq in high pass filter(1/8).

(c) Simulation Output:


x(n)=10*cos(2*pi*n/20)+5*cos(2*pi*n/8)signal
Amplitude
20
0
-20
Time
y(n)=x(n)*h(n) of low pass Filter
Amplitude
500
0
-500
Time
y(n)=x(n)*h(n) of high pass Filter
Amplitude
200
0
-200
Time