Digital Signal Processing
MATLAB HW2 - q1

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Professor: Dr. Sheikhzadeh Author: [SeyedAli] - [SeyedHosseini] E-mail: [alisnake@aut.ac.ir] %which I`m about to change ASAP University: Amirkabir University of Technology

Clear recent data

```
clear; close all; clc;
```

Part A

```
wp = 0.2*pi; %Frequency of Passing
ws = 0.4*pi; %Frequency of Stoping
tr_width = ws - wp; %Transition Band
Mb = ceil(11*pi/tr_width) + 1 ; % Length Of Window With Exact Formula
 in Table 7.1 of Signal Processing Book
nb=[0:1:Mb-1] ; %Samples
display(Mb)
%
wc = (ws+wp)/2 ; % Ideal LPF cutoff frequency
hdb = ideal_lp(wc,Mb); %Impulse Response of Desierd h(n)(BlackMan)
w_black = (blackman(Mb))'; %blackman window with M = 56 but we must
 transpose it in order to stop miscalculation
hb = hdb .* w_black; %Impulse Response of Designed Filter
[db,mag,pha,grd,w] = freqz_m(hb,[1]); %Modified Version of Frequency
 domain response
delta_w = 2*pi/1000; %Minimum of Error in PassBand Magnitude
```

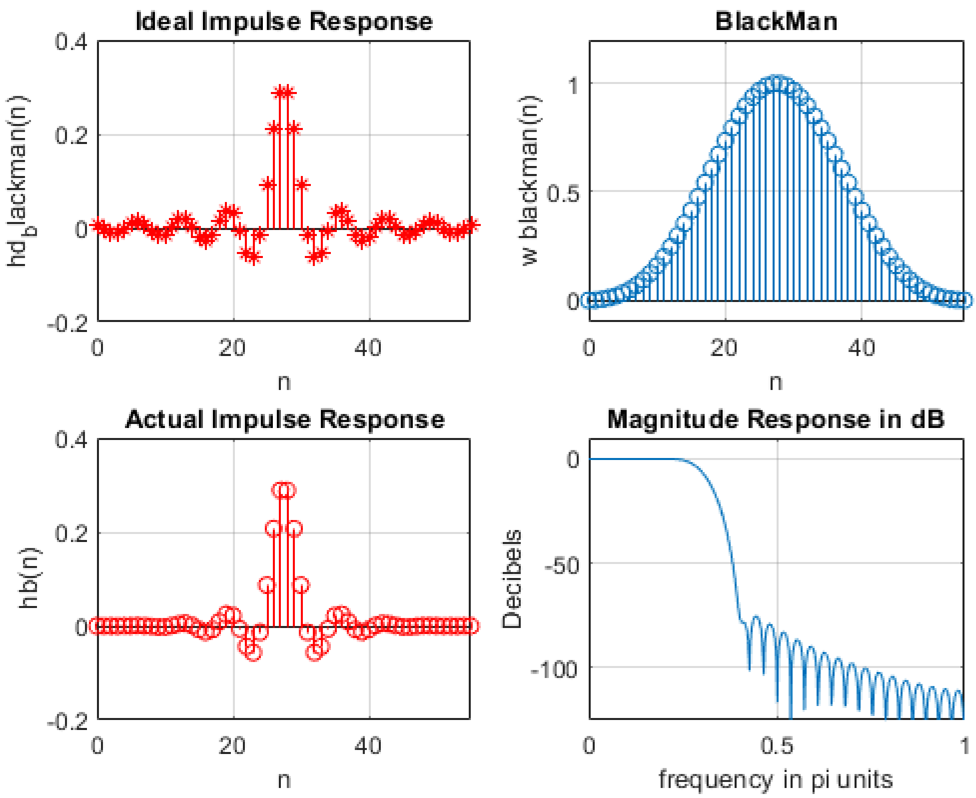
```
Mb =

    56
```

plots

```
figure(1)
```

```
subplot(2,2,1);
stem(nb,hdb,"r*"); grid on
title('Ideal Impulse Response')
axis([0 Mb-1 -0.2 0.4]);
xlabel('n');
ylabel('hd_blackman(n)');

subplot(2,2,2);
stem(nb,w_black); grid on
title('BlackMan')
axis([0 Mb-1 -0.1 1.2]);
xlabel('n');
ylabel('w blackman(n)');

subplot(2,2,3);
stem(nb,hb,"r"); grid on
title('Actual Impulse Response');
axis([0 Mb-1 -0.2 0.4]);
xlabel('n');
ylabel('hb(n)');

subplot(2,2,4);
plot(w/pi,db);
title('Magnitude Response in dB');
grid on;
axis([0 1 -125 10]);
xlabel('frequency in pi units');
ylabel('Decibels');

%Actual Rp and As
Rp = -(min(db(1:1:wp/delta_w+1))); % Actual Passband Ripple
As = -round(max(db(ws/delta_w+1:1:501))); % Min Stopband attenuation
display(Rp)
display(As)
```

```
Rp =

    0.0033


As =

    74
```



Part B (Kaiser Design)

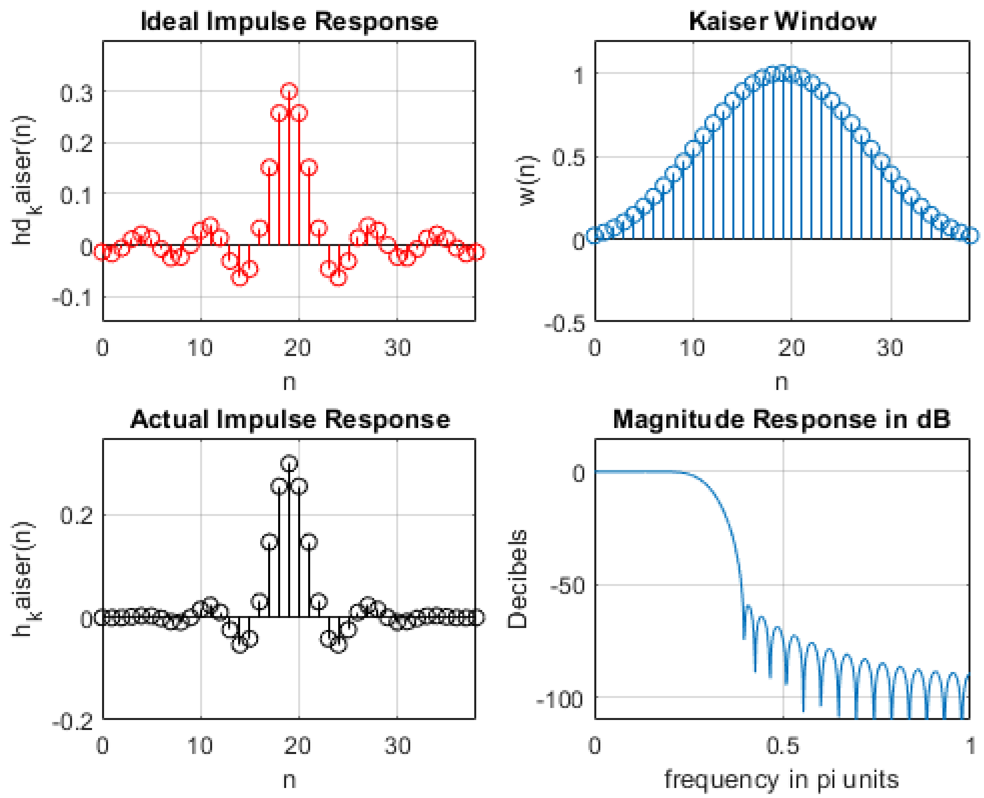
```
wp = 0.2*pi; %PassBand Frequency
ws = 0.4*pi; %StopBand Frequency
As = 60; %Desired StopBand Attenuation
tr_width = ws - wp; %Transition Band
Mk = ceil((As-7.95)/(2.285*tr_width)+1)  +  1 ; % Length Of Window
display(Mk)

nk=[0:1:Mk-1]; %Smaples
beta = 0.1102*(As-8.7) ; %For As >= 50
display(beta)
wc = (ws+wp)/2; %CutOff Frequency
hdk = ideal_lp(wc,Mk); %Desired h(n)
w_kai = (kaiser(Mk,beta))';% Kaiser Window with M = 39 (Notice that
 length of kaiser window is less than Blackman)
hk = hdk .* w_kai; %Impulse response of Desired filter h(n)
[db,mag,pha,grd,w] = freqz_m(hk,[1]);%Modified Version of Frequency
 domain response
delta_w = 2*pi/1000; %Minimum of Error in PassBand Magnitude
As_k = -round(max(db(ws/delta_w+1:1:501))) % Min Stopband Attenuation
```

```
Mk =

    39
```

```
beta =

    5.6533


As_k =

    59
```

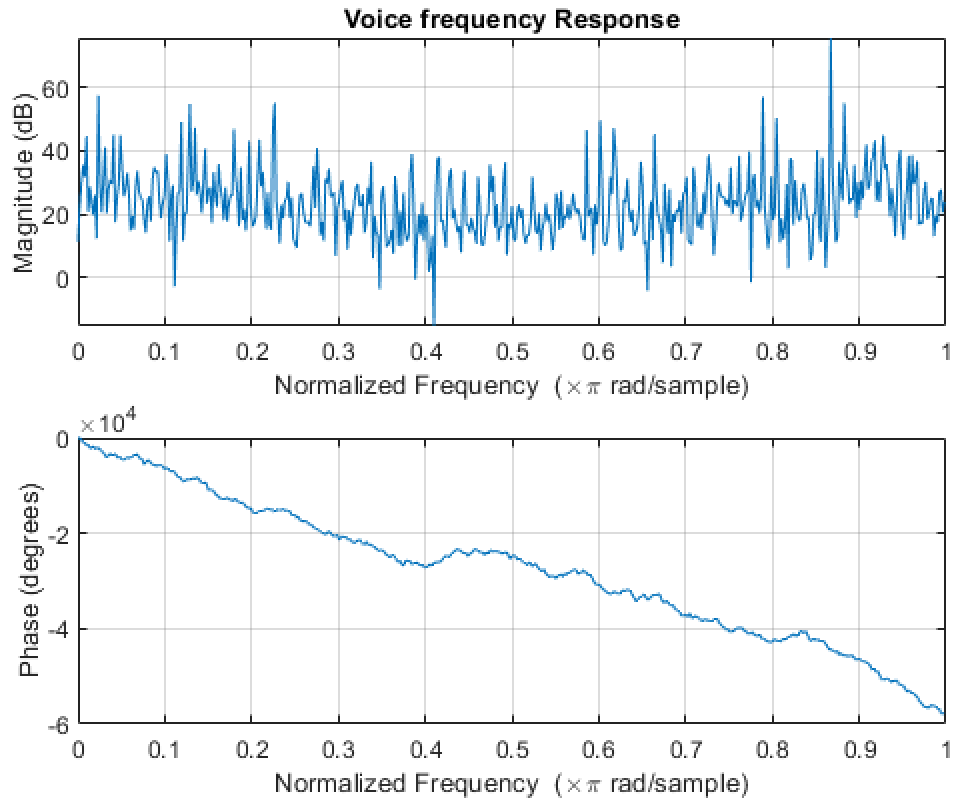
Plots

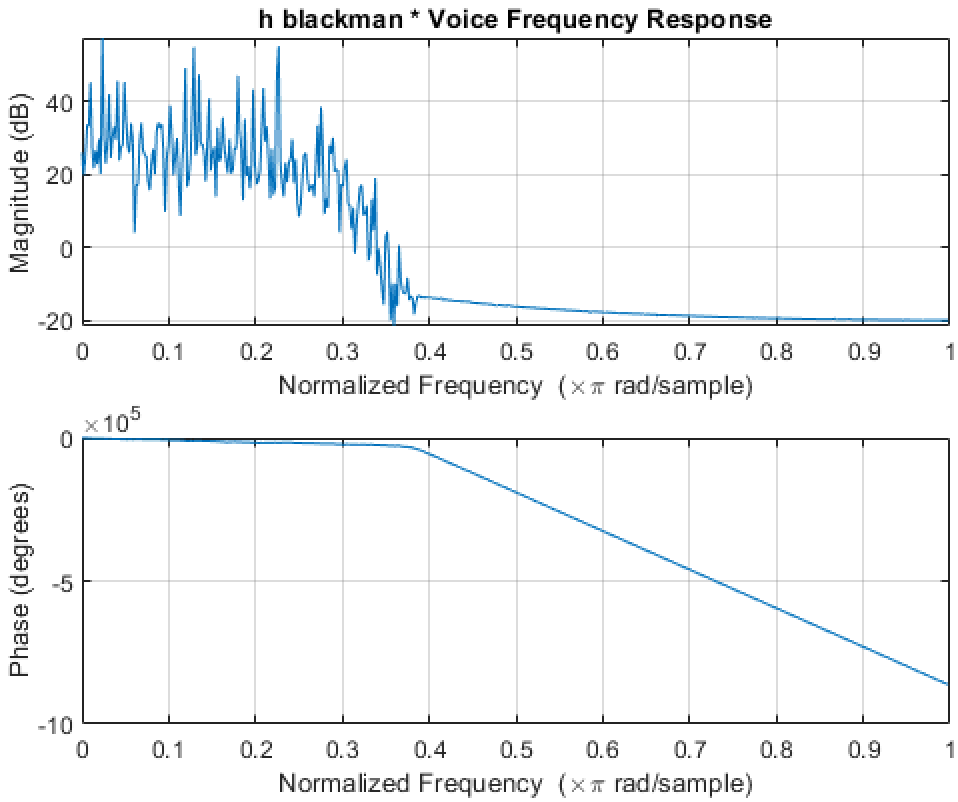
```
figure(2)
subplot(2,2,1);
stem(nk,hdk,"r-"); grid on
title('Ideal Impulse Response')
axis([0 Mk-1 -0.15 0.4]);
xlabel('n');
ylabel('hd_kaiser(n)')

subplot(2,2,2);
stem(nk,w_kai);grid on
title('Kaiser Window')
axis([0 Mk-1 -0.5 1.2]);
xlabel('n');
ylabel('w(n)')

subplot(2,2,3);
stem(nk,hk,"ko");
title('Actual Impulse Response');grid on
axis([0 Mk-1 -0.2 0.35]);
xlabel('n');
ylabel('h_kaiser(n)')

subplot(2,2,4);
plot(w/pi,db);
title('Magnitude Response in dB');grid on
axis([0 1 -110 15]);
xlabel('frequency in pi units');
ylabel('Decibels')
```

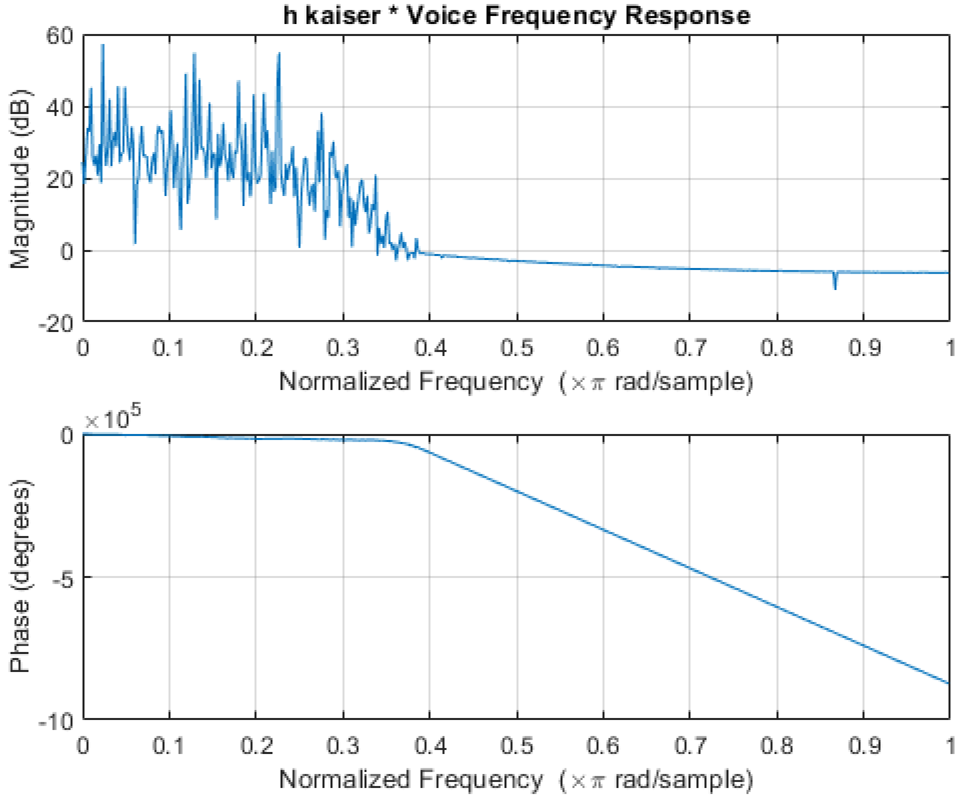


Part C

```
[voice , Fs] = audioread('multi_tone.wav') ;%STORING AUDIO in Voice
figure(3)
freqz(voice); %Plotting the Frequeny Response of Voice itself
title('Voice frequency Response')

voice_b = filter(hb,1,voice); %using filter command to convolution
 h_blackman and voice
voice_b = voice_b' ;
figure(4)
freqz(voice_b); %Plotting the Frequeny Response of h_blackman * voice
title('h blackman * Voice Frequency Response')

voice_k = filter(hk,1,voice); %using filter command to convolution
 h_kaiser and voice
voice_k = voice_k' ;
figure(5)
freqz(voice_k); %Plotting the Frequeny Response of h_kaiser * voice
title('h kaiser * Voice Frequency Response')
```


Voice frequency Response
Magnitude (dB)
60
40
20
0
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)
×10^4
Phase (degrees)
0
-2
-4
-6
Normalized Frequency (×π rad/sample)


h blackman * Voice Frequency Response
Magnitude (dB)
40
20
0
-20
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)
×10^5
Phase (degrees)
0
-5
-10
Normalized Frequency (×π rad/sample)



Function of Ideal Low Pass Filter

```
function hd = ideal_lp(wc,M);
% Ideal LowPass filter computation
% --------------------------------
% [hd] = ideal_lp(wc,M)
% hd = ideal impulse response between 0 to M-1
% wc = cutoff frequency in radians
% M = length of the ideal filter
%
alpha = (M-1)/2;
n = [0:1:(M-1)]; %time samples
m = n - alpha;
fc = wc/pi; %Cut off frequency in Hz
hd = fc*sinc(fc*m); %Creating desired H Function
end
```

Function for Frequency Response Calculations

```
function [db,mag,pha,grd,w] = freqz_m(b,a);
% Modified version of freqz subroutine
% ------------------------------------
% [db,mag,pha,grd,w] = freqz_m(b,a);
% db = Relative magnitude in dB computed over 0 to pi radians
% mag = absolute magnitude computed over 0 to pi radians
```

```
% pha = Phase response in radians over 0 to pi radians
% grd = Group delay over 0 to pi radians
% w = 501 frequency samples between 0 to pi radians
% b = numerator polynomial of H(z) (for FIR: b=h)
% a = denominator polynomial of H(z) (for FIR: a=[1])
%
[H,w] = freqz(b,a,1000,'whole');
H = (H(1:1:501));
w = (w(1:1:501));
mag = abs(H);
db = 20*log10((mag+eps)/max(mag));
pha = angle(H);
grd = grpdelay(b,a,w);
end
```

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