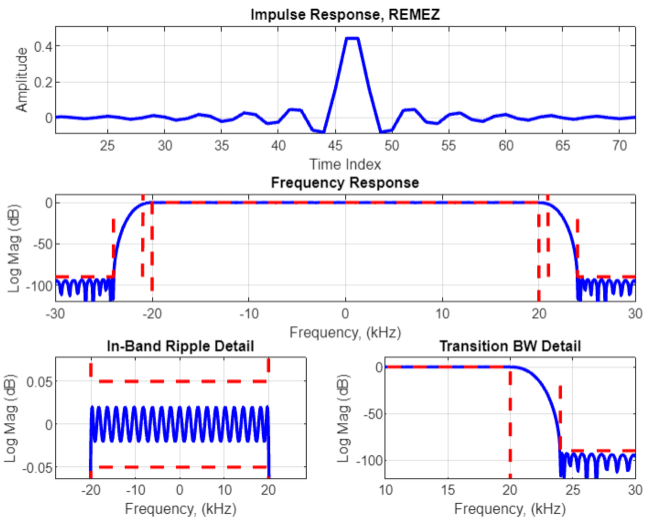
ECE_161B_Project_1

Problem 1. Use the FIRPM (Remez) algorithm to design a FIR filter with the following specifications:

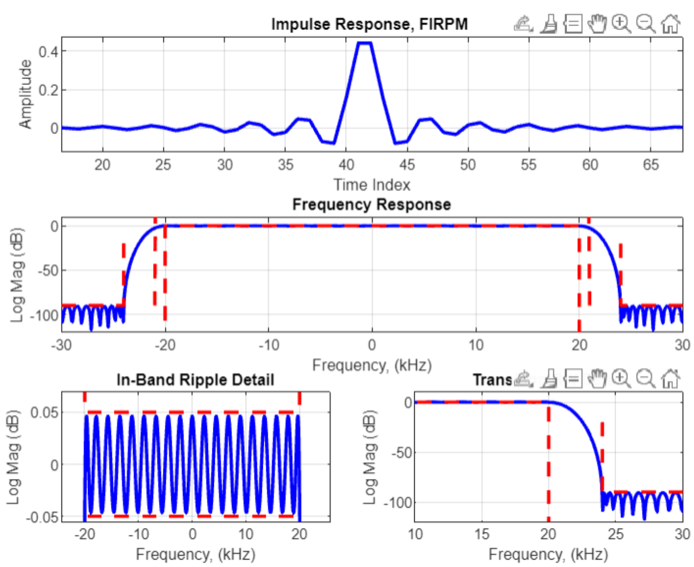
Sample rate 88.2 kHz.
passband 0 to 20 kHz. Passband Ripple 0.1 dB
Stopband 24 to 44.1 kHz. Stopband attenuation -90dB
Estimate the required filter length.
Design filter and present impulse response, Log magnitude frequency response, and Zoom to passband ripple frequency response.

91-tap REMEZ:




```
fs = 88.2;
f1 = 20;
f2 = 24;
N=(fs/(f2-f1))*(90/22);
h1=remez(91,[0 20 24 44.1]/44.1,{'myfrf',[1 1 0 0]},[1 100]);

figure(203)
subplot(3,1,1)
plot(h1,'b','linewidth',2)
grid on
axis([-1 50 -0.1 0.5])
title('Impulse Response, REMEZ')
xlabel('Time Index')
ylabel('Amplitude')

fh=fftshift(20*log10(abs(fft(h1,1024))));
```

81-tap FIRPM:




```
fs = 88.2;
f1 = 20;
f2 = 24;
r=0.1;
A_dB = -90;
dev = [(10^(r/20)-1)/(10^(r/20)+1) 10^(A_dB/20)];
[n_h1, fo_h1, ao_h1, w_h1] = firpmord([f1 f2], [1 0], dev, fs);
h1 = firpm(81, fo_h1, ao_h1, w_h1);

figure(205)
subplot(3,1,1)
plot(h1,'b','linewidth',2)
grid on
axis([-1 50 -0.1 0.5])
title('Impulse Response, FIRPM ')
xlabel('Time Index')
ylabel('Amplitude')

fh=fftshift(20*log10(abs(fft(h1,1024))));
```

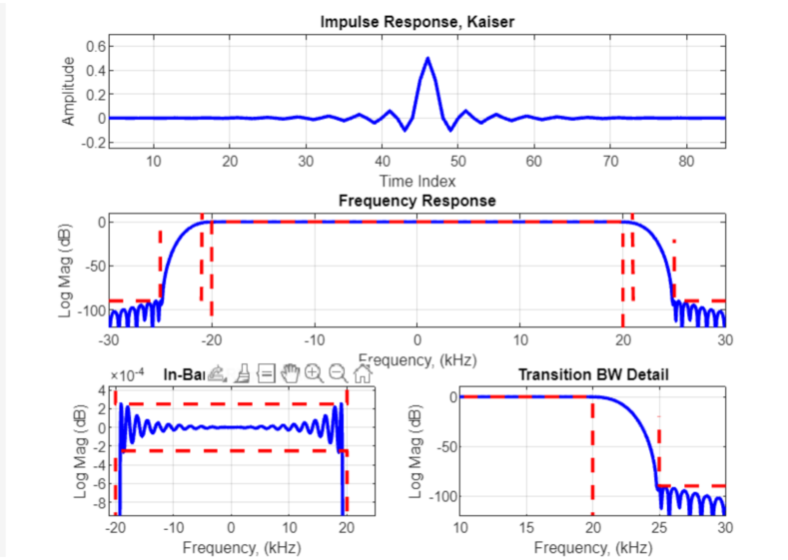
I was advised by a TA to use the filter that has the shortest length while meeting the specs but **since 91 comes up again in my solutions, I will have both for comparison's sake and will refer to the 91-tap for the rest of the project. However, both solutions are correct.**

Problem 2. Use the kaiser windowed algorithm to design a FIR filter with the same specifications as in problem 1. The passband ripple level will be significantly smaller than that in the FIRPM design. Why does that happen?

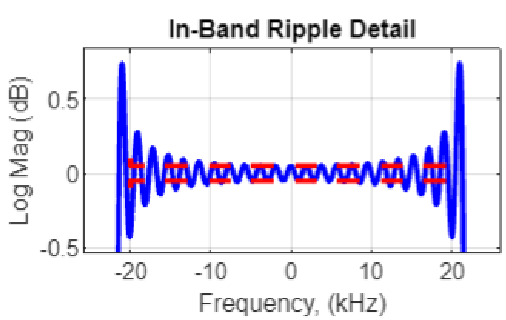
Sample rate 88.2 kHz.	
passband 0 to 20 kHz.	Passband Ripple 0.1 dB
Stopband 24 to 44.1 kHz.	Stopband attenuation -90dB

Estimate the required filter length.
Design filter and present impulse response, Log magnitude frequency response, and Zoom to passband ripple frequency response.

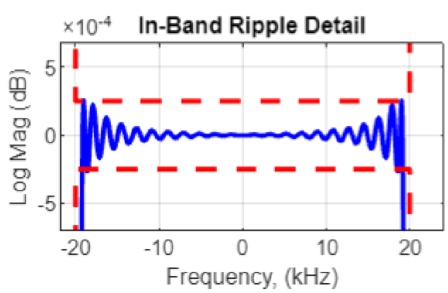
```
fs = 88.2;
f1 = 20;
f2 = 24;
A_dB = 90;
Beta = A_dB/10; %if A_dB < 40 dB have to reduce Beta
N=(fs/(f2-f1))*(A_dB/22);
M = floor((fs/(f2-f1))*A_dB/22); %For windowed design, A_dB/22
if rem(M,2)==0 %we want N to be an odd integer
    M=M+1;
end

MM=(M-1)/2; %Compute end points of array interval, NN
phi=2*pi*(-MM:MM)*(f1+f2)/(2*fs); %compute phase argument of sinc filter

h=sin(phi)./phi; %unscaled sinc filter h
h(MM+1)=1; %correct failed 0/0 computation
h0=h.*kaiser(2*MM+1,Beta)'; %Apply window to obtain Stopband Ripple
h1=h0*(f2+f1)/fs; %Scale filter gain f0/fs

figure(201)
subplot(3,1,1)
plot(h1,'b','linewidth',2)
grid on
axis([-1 50 -0.1 0.5])
title('Impulse Response, Kaiser')
xlabel('Time Index')
ylabel('Amplitude')

fh=fftshift(20*log10(abs(fft(h1,1024))));
```



To meet the specification, I needed to build a Windowed Sinc filter with length 91. The sinc function without the window has passband ripple similar to the remez filter, it peaks on both ends and gradually decreases to the 0.1 passband ripple we see in the remez.



However, when I applied a kaiser window to the sinc function, the passband ripple became smaller. This is because the window reduces the abruptness of the truncated ends, thereby improving the frequency response.



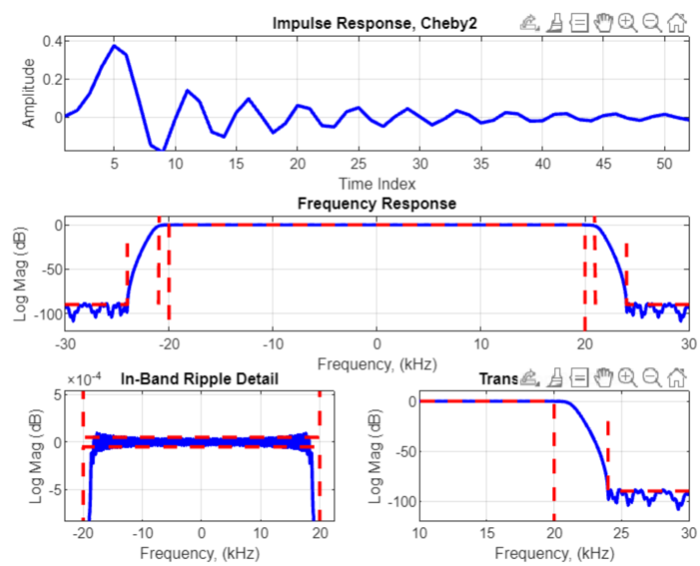
As we can see, the peaks are smaller, and the transition is smoother.

Problem 3. Use the MATLAB Cheby2 filter design algorithm to design an IIR filter with the same specifications as in problem 1. The passband ripple level will be significantly smaller than that in the FIRPM design. Why does that happen?

Sample rate 88.2 kHz.
passband 0 to 20 kHz. Passband Ripple 0.1 dB
Stopband 24 to 44.1 kHz. Stopband attenuation -90dB
What order IIR filter was required to meet the filter specifications.

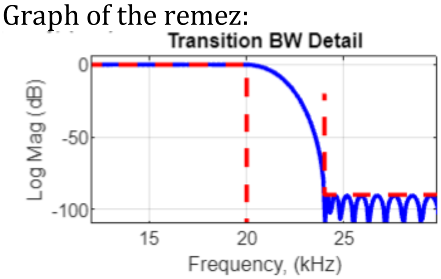
Design filter and use the Matlab TF2SOS to convert the numerator and denominator of the IIR design to second order sub-filters. Use the second order filters to determine the required scaling factor for each sub-filter. Use the cascade filters to determine the impulse response. Present the impulse response, Log magnitude frequency response, and Zoom to passband ripple frequency response.

```
fs = 88.2;
fc = 24/fs;
fe = 20/fs;
ws = fc*2;
wp = fe*2;
A = 90;
[n, wn] = cheb2ord(wp,ws,0.1, A);
[b1,a1] = cheby2(n,A,wn,"low");
[sos1, g1] = tf2sos(b1,a1);
u = [1 zeros(1,199)];
h2 = sosfilt(sos1, u);
h3 = h2*g1;

figure(200)
subplot(3,1,1)
plot(h3,'b','linewidth',2)
grid on
axis([-1 50 -0.1 0.5])
title('Impulse Response, Cheby2 ')
xlabel('Time Index')
ylabel('Amplitude')

fh=fftshift(20*log10(abs(fft(h3,1024))));
```

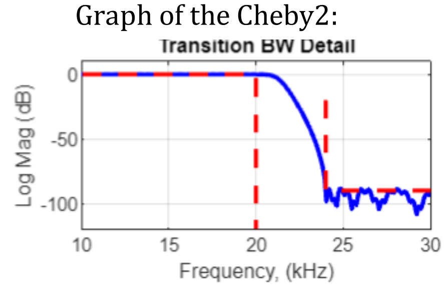


To meet the specification, I needed to build a 17th order Chebyshev II filter. The Chebyshev type filter has a steep roll-off transition band which allows for a smaller passband and equiripple in the stopband. Since the Cheby2 has a lot steeper roll-off compared to the remez, it has a smaller passband ripple.

Graph of the remez:



Graph of the Cheby2:

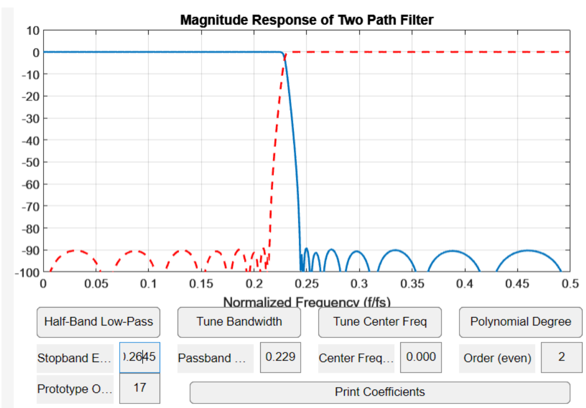


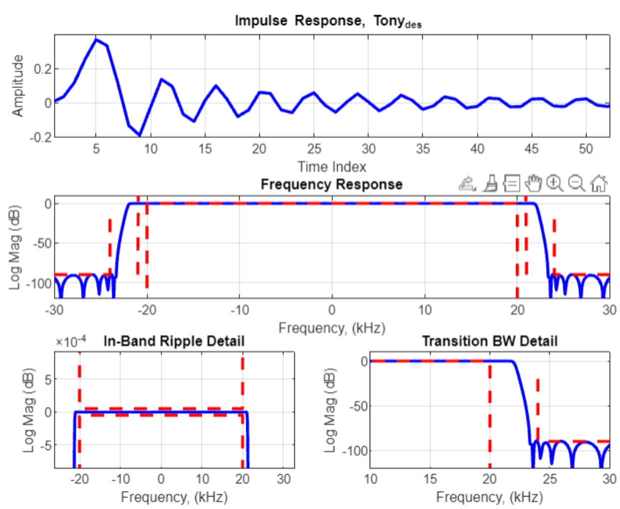
Notice how the Cheby2 transition band is less rounded than the remez.

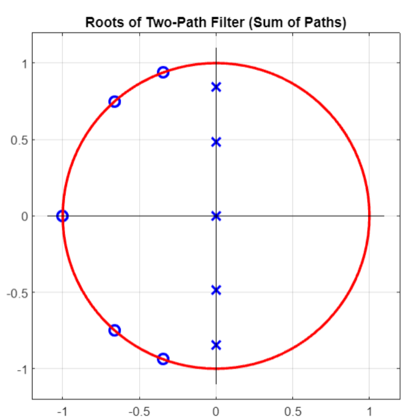
Problem 4. Use the ALL-PASS tony_des2 algorithm to design the two-path filter with the same specifications as in problem 1. The passband ripple level will be significantly smaller than that in the FIRPM design. Why does that happen?

Sample rate 88.2 kHz.	
passband 0 to 20 kHz.	Passband Ripple 0.1 dB
Stopband 24 to 44.1 kHz.	Stopband attenuation -90dB

What order ALL-PASS filter was required to meet the filter specifications?
Design filter and use the sum of the two-path impulse response to form the impulse response. Present the impulse response, Log magnitude frequency response, and Zoom to passband ripple frequency response. This task is similar to HW-3 All-Pass Filter design.








```
d1_top=[1.000000000000000                    0   0.048107731416794];
d2_top=[1.000000000000000                    0   0.342999823886247];
d3_top=[1.000000000000000                    0   0.660449413422730];
d4_top=[1.000000000000000                    0   0.877258630930767];

d0_bot=[1.000000000000000                    0];
d1_bot=[1.000000000000000                    0   0.175479332448430];
d2_bot=[1.000000000000000                    0   0.512332816787186];
d3_bot=[1.000000000000000                    0   0.780693586946028];
d4_bot=[1.000000000000000                    0   0.959929140336011];


x0=[1 zeros(1,1000)];
ytp1=filter(fliplr(d1_top),d1_top,x0);
ytp2=filter(fliplr(d2_top),d2_top,ytp1);
ytp3=filter(fliplr(d3_top),d3_top,ytp2);
ytp4=filter(fliplr(d4_top),d4_top,ytp3);

ybt0=filter(fliplr(d0_bot),d0_bot,x0);
ybt1=filter(fliplr(d1_bot),d1_bot,ybt0);
ybt2=filter(fliplr(d2_bot),d2_bot,ybt1);
ybt3=filter(fliplr(d3_bot),d3_bot,ybt2);
ybt4=filter(fliplr(d4_bot),d4_bot,ybt3);
h3=(ytp4+ybt4)/2;

figure(201)
subplot(3,1,1)
plot(h3,'b','linewidth',2)
grid on
axis([-1 50 -0.1 0.5])
title('Impulse Response, Tony_des ')
xlabel('Time Index')
ylabel('Amplitude')

fh=fftshift(20*log10(abs(fft(h3,1024))));
```

To meet the specification, I needed to build a 17th order Tony_des2 filter. Similar to Cheby2, since the Tony_des2 has a lot steeper roll-off in the transition band compared to the remez, it has a smaller passband ripple. However, it has even better passband ripple than the Cheby2 because the transition band is even steeper.

Final Comments: I noticed that the number of taps for the Remez and the Kaiser window are the same (91), while the Cheby2 and the Tony_des2 have the same order (17). I think this is because the Remez and Kaiser are both FIR filters and the Cheby2 and the Tony_des2 are both IIR filters.

How I graphed everything:

```
subplot(3,1,2)
plot([-0.5:1/1024:0.5-1/1024]*fs,fh,'b','linewidth',2)
hold on
plot([-20 -20 +20 +20],[-200 0 0 -200],'--r','linewidth',2)
plot([-30 -24 -24],[-90 -90 -10],'--r','linewidth',2)
plot([+30 +24 +24],[-90 -90 -20],'--r','linewidth',2)
plot([21 21 -3],[-90 -90 30000],'--r','linewidth',2)
plot([-21 -21 -3],[-90 -90 30000],'--r','linewidth',2)
hold off
grid on
axis([-30 +30 -120 10])
title('Frequency Response')
xlabel('Frequency, (kHz)')
ylabel('Log Mag (dB)')

subplot(3,2,5)
plot([-0.5:1/1024:0.5-1/1024]*fs,fh,'b','linewidth',2)
hold on
plot([-20 -20 20 20],[-0.1 -0.05 -0.05 -0.1],'--r','linewidth',2)
plot([-20 -20 20 20],[+0.1 +0.05 +0.05 +0.1],'--r','linewidth',2)
hold off
grid on
axis([-20 +20 -0.05 0.05])
title('In-Band Ripple Detail')
xlabel('Frequency, (kHz)')
subplot(3,2,6)
plot([-0.5:1/1024:0.5-1/1024]*fs,fh,'b','linewidth',2)
hold on
plot([-20 -20 +20 +20],[-200 0 0 -200],'--r','linewidth',2)
plot([+30 +24 +24],[-90 -90 -20],'--r','linewidth',2)
hold off
grid on
axis([10 +30 -120 10])
title('Transition BW Detail')
xlabel('Frequency, (kHz)')
ylabel('Log Mag (dB)')
```