EEE 424 - Digital Signal Processing - Coding Assignment 4 - FIR Filter

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1) We will use a rectangular window to design a LPF and a HPF. We will then combine these two systems by convolving their impulse responses.

```
% Configuration
fS = 20000;  % Sampling rate
fL = 300;    % Low cutoff frequency
fH = 3000;   % High cutoff frequency
NL = 7;      % Filter length for high-pass (must be odd)
NH = 5;      % Filter length for low-pass (must be odd)

% Low-pass filter with cutoff frequency fH
nH = 0:NH-1;
hlpf = sinc(2 * fH / fS * (nH - (NH - 1) / 2));
hlpf = hlpf / sum(hlpf);  % Normalize

% High-pass filter with cutoff frequency fL
nL = 0:NL-1;
hhpf = sinc(2 * fL / fS * (nL - (NL - 1) / 2));
hhpf = hhpf / sum(hhpf);  % Normalize
hhpf = -hhpf;
hhpf((NL + 1) / 2) = hhpf((NL + 1) / 2) + 1;  % Add impulse at center

% Convolve both filters to get band-pass filter
nom = conv(hlpf, hhpf);
nom = [nom(1:5) nom(7:11)]; %cut nom down to L=10 while keeping linear phase
property
n = 0:length(nom)-1;
```

The transfer function coefficients c_i's are given as:

```
disp([0:9;nom])
```

```
        0    1.0000    2.0000    3.0000    4.0000    5.0000    6.0000    7.0000    8.0000    9.0000
  -0.0192   -0.0520   -0.0904    0.0119    0.0872    0.0872    0.0119   -0.0904   -0.0520   -0.0192
```

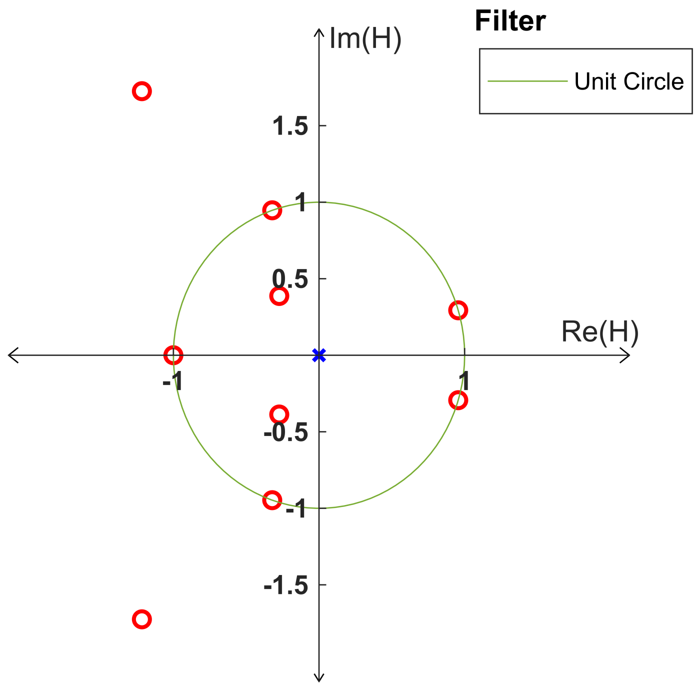
Then the impulse response and transfer function can be written as:

$$h[n] = \sum_{i=0}^{9} \delta[n-i]c_i \qquad H(z) = \sum_{i=0}^{9} z^{-i}c_i$$

We can then plot the zeros of our filter's transfer function. We can find the zeros by finding the roots of the polynomial whose coefficients is given by our impulse response.

```
clf;
polezeroplot(zeros([1 9]),roots(nom),'Filter','H','on',[400 400]);
h = gobjects(3, 1);
h(1) = plot(cos(linspace(0,2*pi,100)),sin(linspace(0,2*pi,100)),'color',[0.4660
0.6740 0.1880],'DisplayName', 'Unit Circle'); %unit circle
legend(h([1])); %add the legend
```

```
hold off
legend("Position", [0.7171 0.8192 0.2675, 0.0812])
```

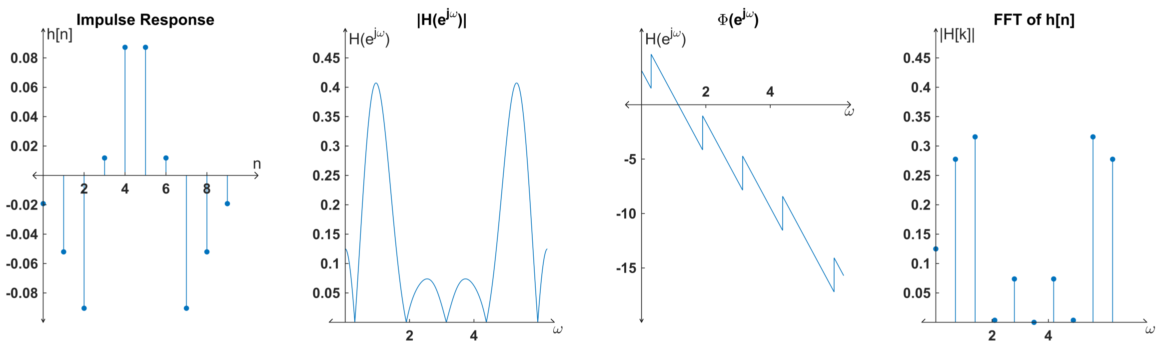


We can calculate the DTFT as:

```
omega = linspace(0,2*pi,100000);
finalsum = zeros(1,length(omega));
for i = 1:length(nom)
    finalsum = finalsum + (nom(i)*exp(1j*omega).^(-1*i));
end
```

Let's plot the impulse response, frequency response and the fft of our signal alltogeter as:

```
clf;
subplot(1,4,1) % impulse response
finestem(n,real(nom),'Impulse Response','n','h[n]',[-0.5 10.5],[-0.1 0.1],'off',
[1600 400],'resp','o');
subplot(1,4,2) % magnitutes of FT
fineplot(omega,abs(finalsum),'|H(e^{j\omega})|','\omega','H(e^{j\omega})',[-0.5
6.5],[0 0.5],'off',[1600 400],'mag','-');
subplot(1,4,3) % phases of FT
fineplot(omega,unwrap(angle(finalsum)),'\Phi(e^{j\omega})','\omega','H(e^{j\omega
})',[-0.5 6.5], [-20 7],'off',[1600 400],'phase','-');
subplot(1,4,4) % FFT of impulse response
finestem(linspace(0,2*pi,length(nom)),[abs(fft(nom))],'FFT of h[n]','\omega','|
H[k]|', [-0.5 7.5],[0 0.5],'off',[1600 400],'fft','o');
```



We obtained a filter that has a passband between $0.03\pi - 0.3\pi$. If we try to improve this filter, we will have to make a trade-off between different specifications such as stopband rejection, transition band witdh etc. Therefore, while it may be possible to optimise these trade-offs, it would not be possible to design an objectively better filter.

2) In the second question, we first open our recording and reduce it to a single channel recording.

```
clf;
recording = audioread('recording.mp3');
monoAudio = mean(recording, 2);
monoAudio = monoAudio(1:480000); %clip 10 seconds of samples
```

We then need to downsample our 48kHz sampling rate recording down to 20kHz. We first upsample with 5, then apply a lowpass filter to avoid aliasing, then downsample with 12.

```
upsampled = zeros([1,4800000]);%upsampling
for i = 0:length(monoAudio)-1
    upsampled(i*5+1:i*5+5) = [monoAudio(i+1) zeros([1,4])];
end

inter_filtered = lowpass(upsampled,10000,240000); %prevent aliasing when
downsampling

downsampled = inter_filtered(1:12:end);
```

Now we just have to pass our input through our system:

```
filtered = conv(nom,downsampled);
filtered = filtered / max(abs(filtered));
```

We can then listen to our filtered recording and export it.

```
player = audioplayer(filtered,20000);
playblocking(player);
audiowrite('filtered_recording.wav',filtered,20000)
```