## Contents



## QUESTION 1 COMMENTING

```
% DO NOT REMOVE THE LINE BELOW
clear; close all; clc;
```

## QUESTION 2 Thinking in Three Domains 1

```
% LOAD AUDIO
% MAKE SURE 'music.wav' is in the same directory!
[x, fs] = audioread('music.wav');

% DEFINE AXES
w = -pi:pi/8000:pi-pi/8000;
t = 1/fs:1/fs:length(x)/fs;

N = 100;
n = 0:(N-1);
x1 = zeros(2*N, 1);
x1(1) = 1;
```

## 2 (a) Answer question

```
% A Low-pass filter is used because the string bass have lower frequencies
% compared to guitars, which have higher frequencies. A Low-pass filter
% allows the lower frequencies of the pass to pass through while
% accentuating higher frequencies, the guitar.
```

## 2 (b) Answer question

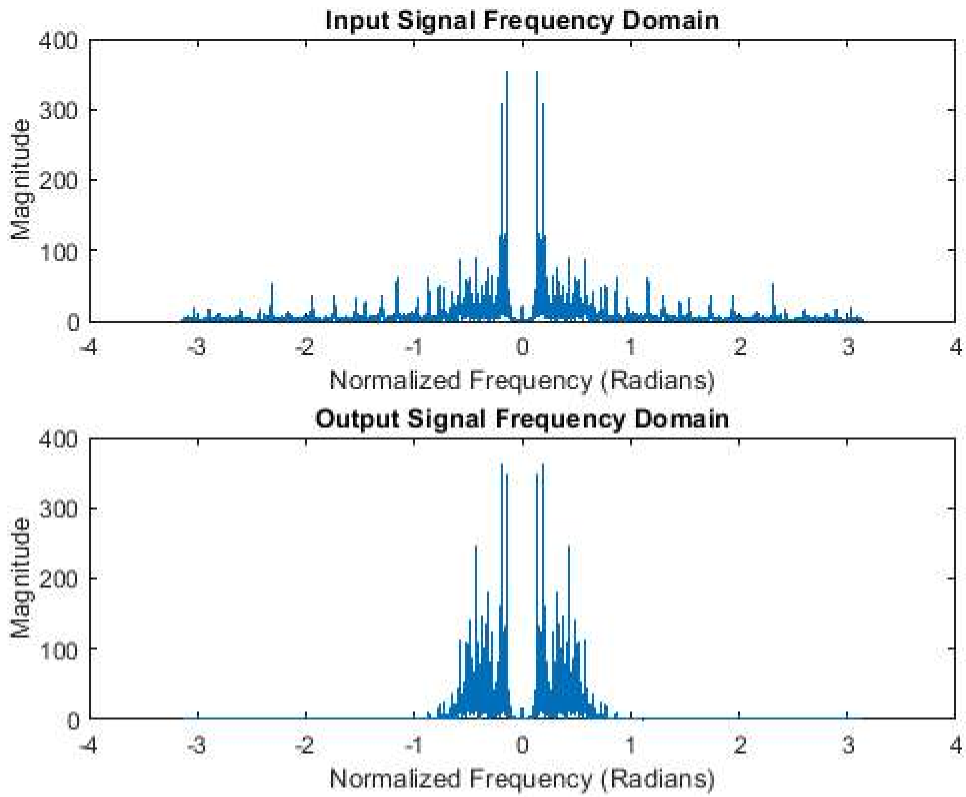
```
% IIR Filters because even though there are only a limited number of values
% for poles and zeroes, poles are used to amplify signals arount certain center
% frequencies
```

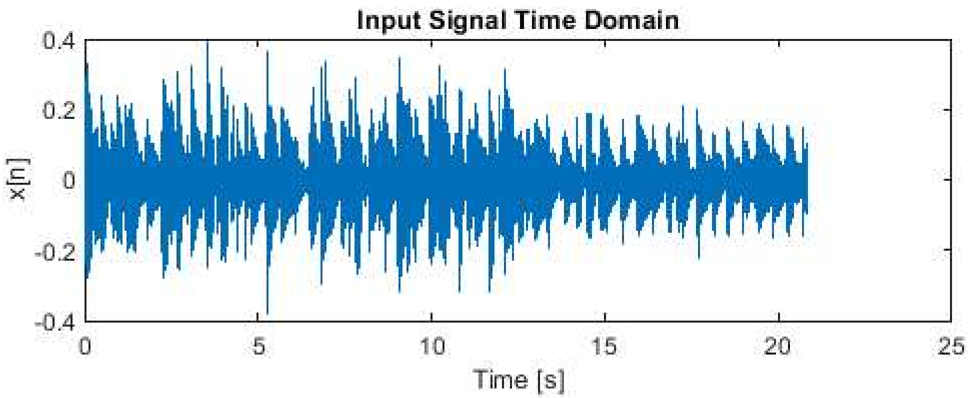
## 2 (c) Plot inputs and outputs

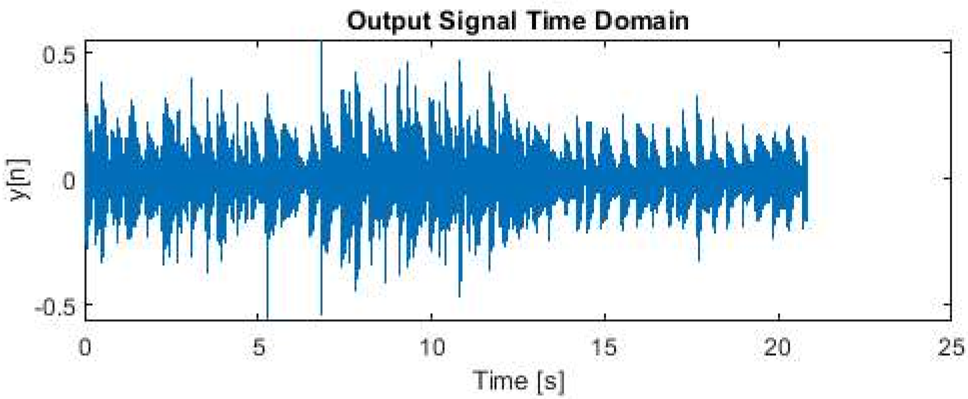
```
% DEFINE POLES (for a low-pass filter)
center = (0.5e4/5e4)*2*pi*2*pi;

mypoles = [ ...
    0.9*exp(1j*center*0.1) ...
```

```
    0.9*exp(-1j*center*0.1) ...
    0.85*exp(1j*center*0.25) ...
    0.85*exp(-1j*center*0.25) ...
    0.8*exp(1j*center*(1/10)) ...
    0.8*exp(-1j*center*(1/10)) ...
    0.8*exp(1j*center*(1/6)) ...
    0.8*exp(-1j*center*(1/6)) ...
    0.8*exp(1j*center*(1/5)) ...
    0.8*exp(-1j*center*(1/5)) ...
];

% DEFINE ZEROS (to shape the filter response)
myzeros = [ ...
    -1 ...
    0.99*exp(1j*(((2*pi*0.98)/fs)+0.5)*2*pi) ...
    0.99*exp(-1j*(((2*pi*0.98)/fs)+0.5)*2*pi) ...
    0.99*exp(1j*(((2*pi*0.95)/fs)+0.5)*2*pi) ...
    0.99*exp(-1j*(((2*pi*0.95)/fs)+0.5)*2*pi) ...
    0.99*exp(1j*(((2*pi*0.97)/fs)+0.5)*2*pi) ...
    0.99*exp(-1j*(((2*pi*0.97)/fs)+0.5)*2*pi) ...
    0.99*exp(1j*(((2*pi*0.93)/fs)+0.5)*2*pi) ...
    0.99*exp(-1j*(((2*pi*0.93)/fs)+0.5)*2*pi) ...
];

% CONVERT POLES AND ZEROS INTO COEFFICIENTS
[b1, a1] = pz2ba(mypoles, myzeros);

% COMPUTE GAIN TO MAINTAIN SIGNAL AMPLITUDE AROUND SOME FREQUENCY
G = abs(sum(a1.*exp(-1j.*0.5e4/5e4*2*pi.*(0:(length(a1)-1))))./sum(b1.*exp(-1j.*0.5e4/5e4*2*pi.*(0:(length(b1)-1)))));

% FILTER INPUT
y = filter(G*b1, a1, x);

% PLOT FREQUENCY DOMAIN AND TIME DOMAIN INPUT / OUTPUT

% COMPUTE FREQUENCY DOMAIN REPRESENTATION
X = DTFT(x, w);
Y = DTFT(y, w);

% PLOT FREQUENCY DOMAIN
figure;
subplot(211);
plot(w, abs(X));
xlabel('Normalized Frequency (Radians)');
```

```
ylabel('Magnitude');
title('Input Signal Frequency Domain');

subplot(212)
plot(w, abs(Y));
xlabel('Normalized Frequency (Radians)');
ylabel('Magnitude');
title('Output Signal Frequency Domain');

% PLOT TIME DOMAIN
figure;
subplot(211);
plot(t, x);
xlabel('Time [s]');
ylabel('x[n]');
title('Input Signal Time Domain');

subplot(212);
plot(t, y);
xlabel('Time [s]');
ylabel('y[n]');
title('Output Signal Time Domain');
```


Input Signal Frequency Domain
Magnitude
Normalized Frequency (Radians)
Output Signal Frequency Domain
Magnitude
Normalized Frequency (Radians)



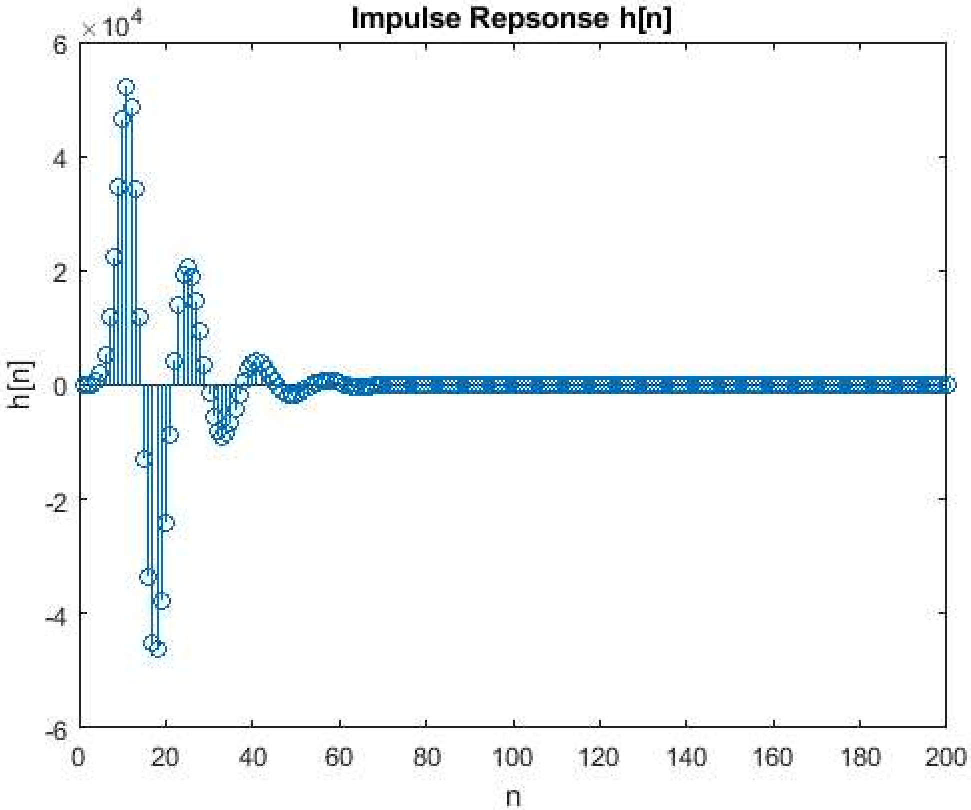




### 2 (d) Answer question

```
% The plots show that the filter attentuates high frequencies whil passing
% the lower frequncies. So the flter does show the areas where the guitar
% dominates as they are lower in the output time domain.
```

### 2 (e) Plot impulse response

```
y1 = filter(b1, a1, x1);

figure;
stem(y1);
title('Impulse Repsonse h[n]');
xlabel('n');
ylabel('h[n]');
```

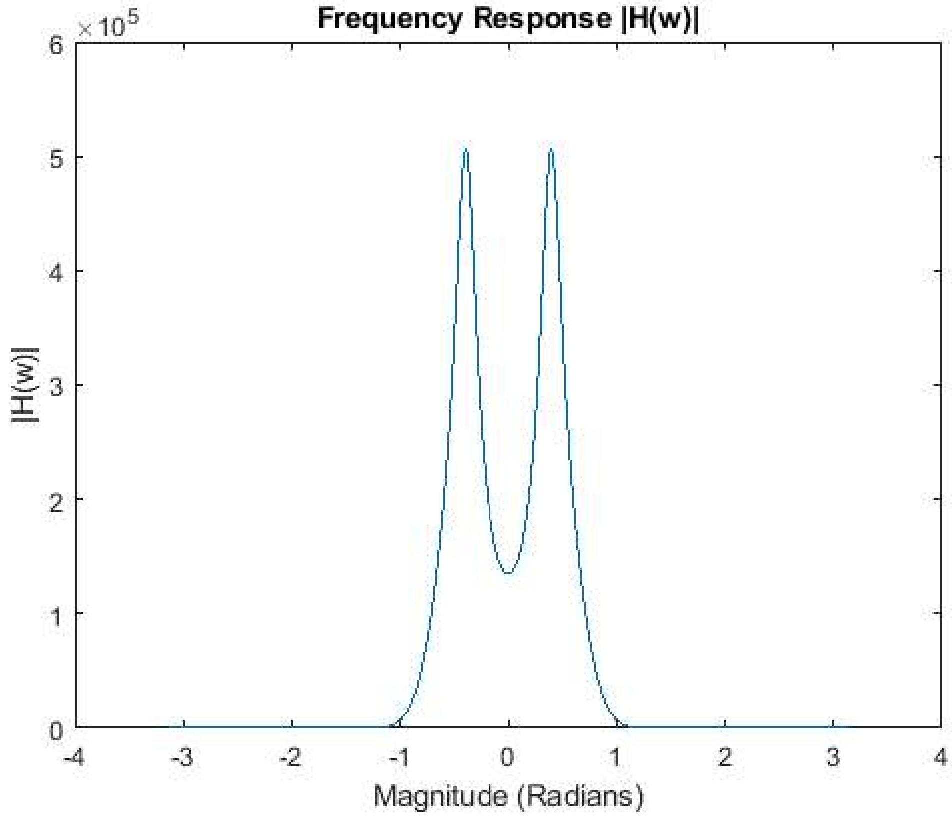



## 2 (f) Answer question

```
% The system is an IIR filter with a damped response where there are zeros
% with magnitudes less than 1. Also, since there are a lot of poles near
% the unit cercle smaller than one, the systme is stable.
```

## 2 (g) Plot freqeuency response

```
H_1 = DTFT(y1, w);

figure;
plot(w,abs(H_1));
title('Frequency Response |H(w)|');
xlabel('Magnitude (Radians)');
ylabel('|H(w)|');
```

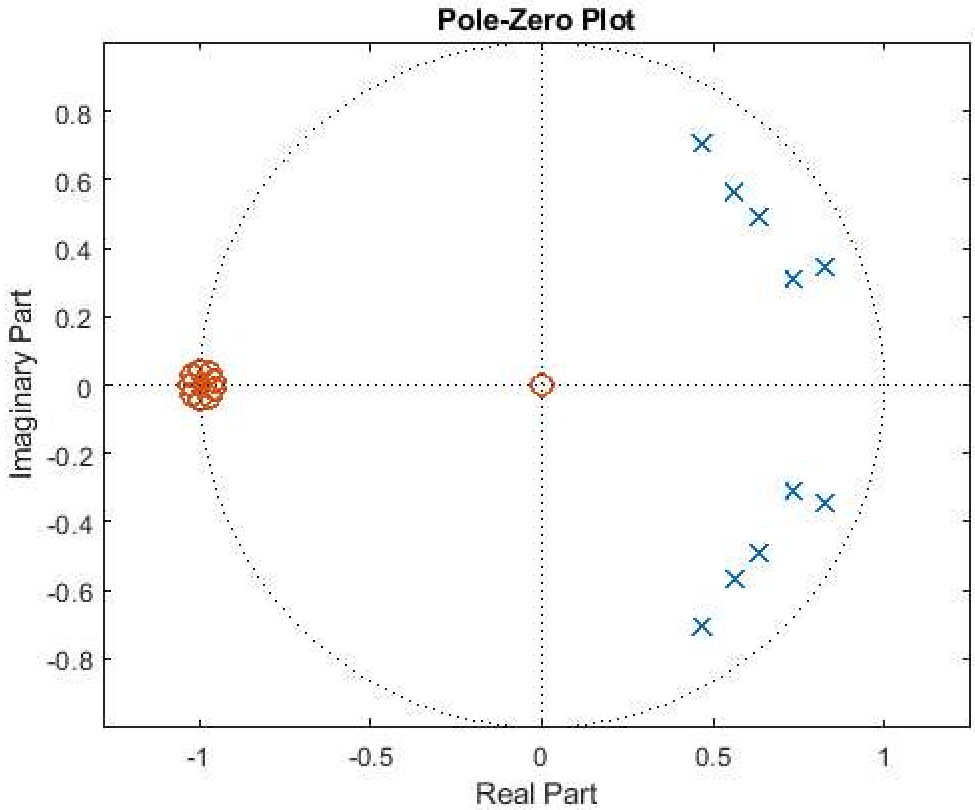



## 2 (h) Answer question

```
% The filter shows a low pass response, which is what we are looking for.
% It does have a transition band slope that could be caused by anything.
% The filter also has a mirror effect, but looking only at the positive
% numbers, it is low-pass
```

## 2 (i) Plot pole-zero plot

```
% PLOT POLE-ZERO PLOT
figure;
pzplot(b1, a1);
axis equal;
```




### 2 (j) Answer question

```
% The filter is a lowpass filter since all the poles are centered around
% lower frequencies to aplify them. And the zeros are centered around the
% higher frequencies. There is a zero at the origin, but there isn't much
% signigicance since there are much more poles than zeros. The system is
% table since all the poles have a magnitude less than 0.
```

### 2 (k) Submit file on Canvas

```
% PLAY MUSIC

disp('Playing Original Music ... ')
soundsc(x, fs)
pause(length(x)/fs*1.1)
```

```
disp('Playing Filtered Music ... ')
soundsc(y, fs)

% SAVE RESULT

% NORMALIZE THE SIGNAL TO AVOID CLIPPING
audiowrite('q2.wav', y./max(abs(y)), fs)
save('q2.mat', 'b1', 'a1');
```

```
Playing Original Music ...
Playing Filtered Music ...
```

## QUESTION 3 Thinking in Three Domains 2

```
% LOAD AUDIO
% MAKE SURE 'music.wav' is in the same directory!
[x, fs] = audioread('music.wav');

% DEFINE AXES
w = -pi:pi/8000:pi-pi/8000;
t = 1/fs:1/fs:length(x)/fs;

N = 200;
n = 0:(N-1);
x1 = zeros(2*N, 1);
x1(1) = 1;
```

## 3 (a) Answer question

```
% The type of filter is high pass because the guitar has higher frequencies
% compared to the string bass, so high-pass filter will allow for the
% higher frequencies while attentuating the lower frequencies. The bass is
% somewhat removed, so the sound is really smalle, but due to the harmonics
% of the bass, some of the sound isn't moved, espacially at the end.
```

## 3 (b) Answer question

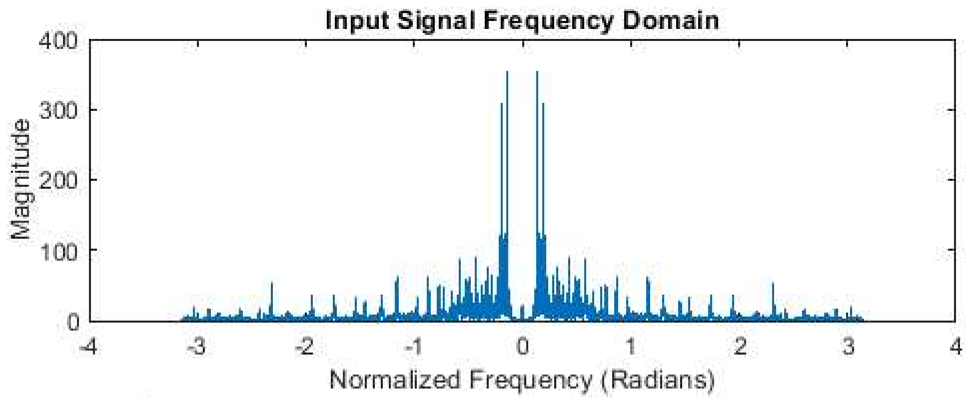
```
% FIR Filter because FIR filters are generally stable and can be designed
% to have linear phase responses, which is good for audio applications.
```

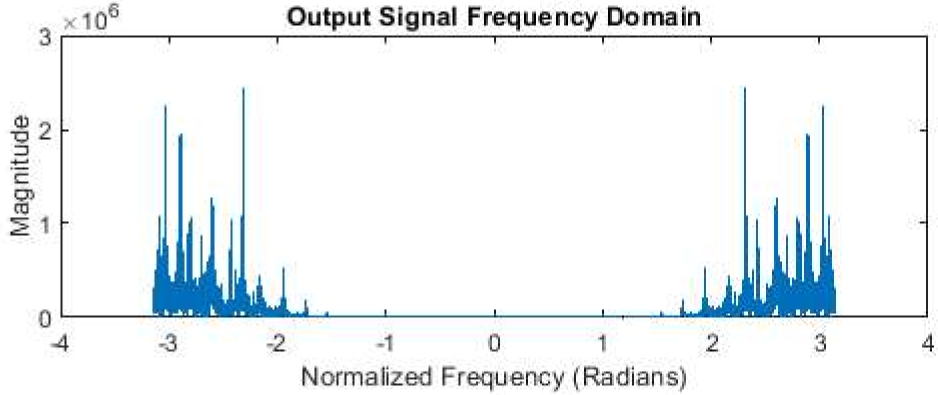
```
% Additionally the requirement of having no more than non-zero coefficients
% fits well with FIR.
```

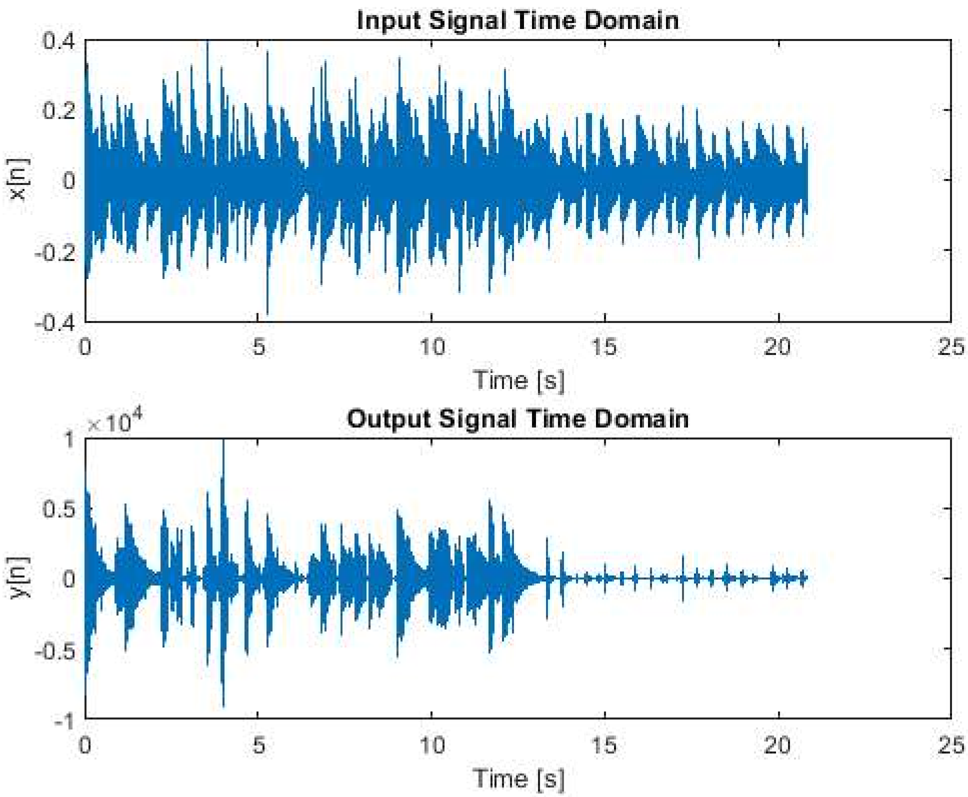
## 3 (c) Plot inputs and outputs

```
% DEFINE POLES (for a high-pass filter)
mypoles = [];

% DEFINE ZEROS (to shape the filter response)
center = (0.5e4/5e4)*2*pi*2*pi;
myzeros = [ ...
    0.9*exp(1j*center*0.1) ...
    0.9*exp(-1j*center*0.1) ...
    0.85*exp(1j*center*0.25) ...
    0.85*exp(-1j*center*0.25) ...
    0.8*exp(1j*center*(1/10)) ...
    0.8*exp(-1j*center*(1/10)) ...
    0.8*exp(1j*center*(1/6)) ...
    0.8*exp(-1j*center*(1/6)) ...
    0.8*exp(1j*center*(1/5)) ...
    0.8*exp(-1j*center*(1/5)) ...
];

% CONVERT POLES AND ZEROS INTO COEFFICIENTS
[b2, a2] = pz2ba(mypoles, myzeros);

% COMPUTE GAIN TO MAINTAIN SIGNAL AMPLITUDE AROUND SOME FREQUENCY
G = abs(sum(a2.*exp(-1j.*0.5e4/5e4*2*pi.*(0:(length(a2)-1))))./sum(b2.*exp(-1j.*0.5e4/5e4*2*pi.*(0:(length(b2)-1)))));

% FILTER INPUT
y = filter(G*b2, a2, x);

% PLOT FREQUENCY DOMAIN AND TIME DOMAIN INPUT / OUTPUT

% COMPUTE FREQUENCY DOMAIN REPRESENTATION
X = DTFT(x, w);
Y = DTFT(y, w);

% PLOT FREQUENCY DOMAIN
figure;
subplot(211);
plot(w, abs(X));
```

```
xlabel('Normalized Frequency (Radians)');
ylabel('Magnitude');
title('Input Signal Frequency Domain');

subplot(212)
plot(w, abs(Y));
xlabel('Normalized Frequency (Radians)');
ylabel('Magnitude');
title('Output Signal Frequency Domain');

% PLOT TIME DOMAIN
figure;
subplot(211);
plot(t, x);
xlabel('Time [s]');
ylabel('x[n]');
title('Input Signal Time Domain');

subplot(212);
plot(t, y);
xlabel('Time [s]');
ylabel('y[n]');
title('Output Signal Time Domain');
```


Input Signal Frequency Domain
Magnitude
Normalized Frequency (Radians)
400
300
200
100
0
-4
-3
-2
-1
0
1
2
3
4


Output Signal Frequency Domain
$\times 10^6$
Magnitude
Normalized Frequency (Radians)
3
2
1
0
-4
-3
-2
-1
0
1
2
3
4

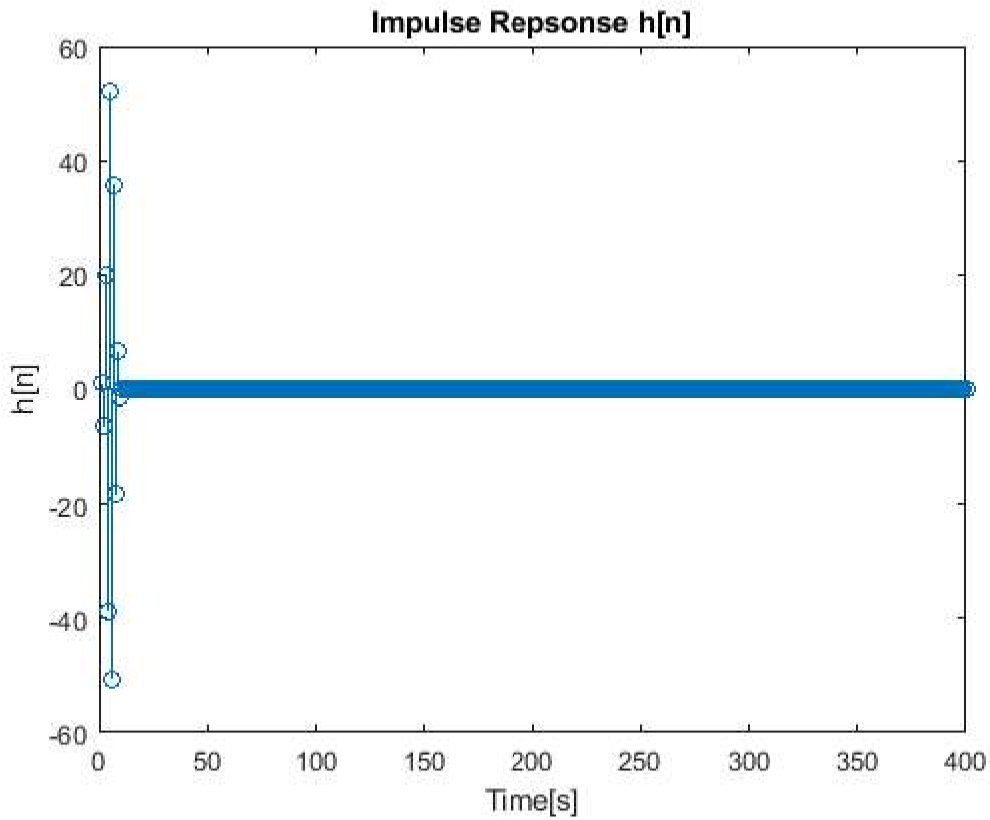



### 3 (d) Answer question

```
% The plot should show a high pass filter where the output at the end where
% the string bass is most prominent is smaller than the original input.
```

### 3 (e) Plot impulse response

```
y1 = filter(b2, a2, x1);

figure;
stem(y1);
title('Impulse Repsonse h[n]');
xlabel('Time[s]');
ylabel('h[n]');
```

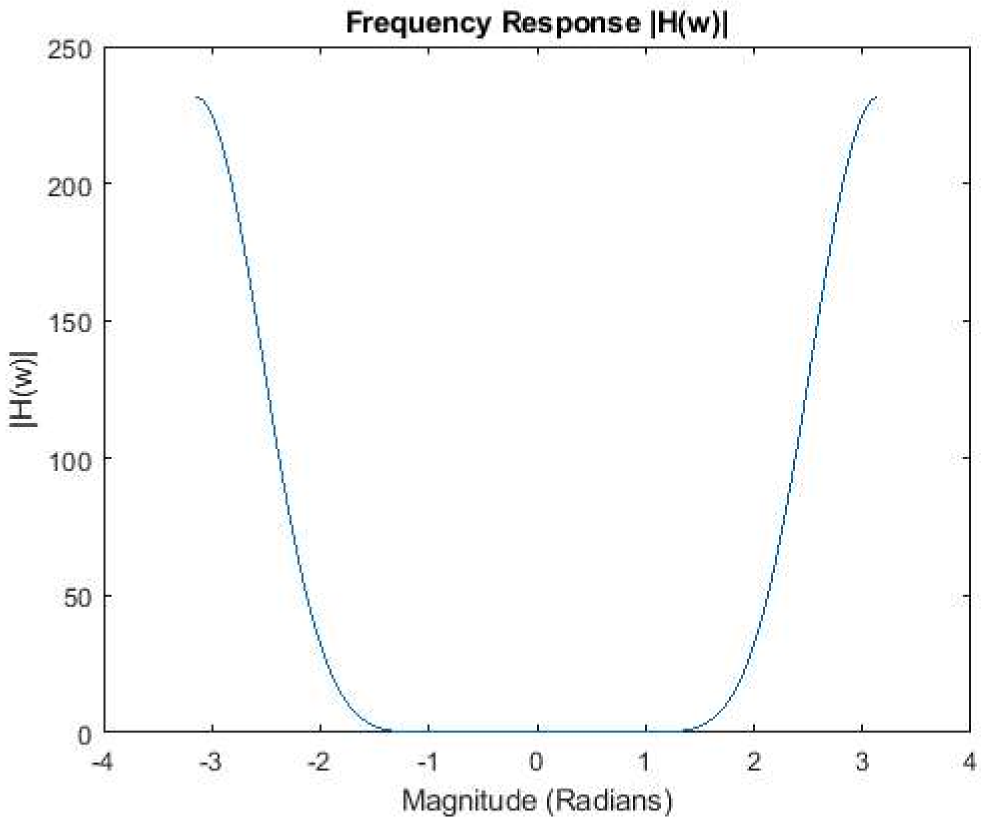



### 3 (f) Answer question

```
% The impulse response shows that the system is stable since there are no
% non-zero poles and the system is finite since there are only 10 values in
% the filtered response.
```

### 3 (g) Plot freqeuency response

```
H_1 = DTFT(y1, w);

figure;
plot(w,abs(H_1));
title('Frequency Response |H(w)|');
xlabel('Magnitude (Radians)');
ylabel('|H(w)|');
```

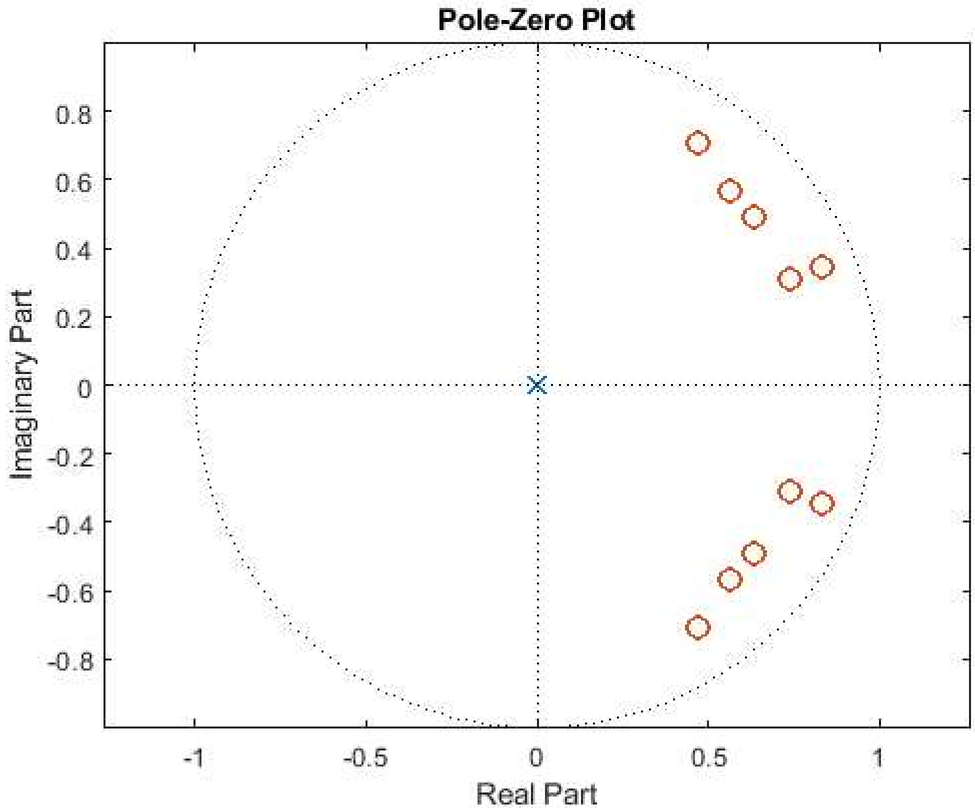



### 3 (h) Answer question

```
% The magnitude response should confirm that the filter allows high
% frequencues to pass wile attentuating lower frequencies. Due to the
% steepness we know the order of the high pass is greater than 1.
```

### 3 (i) Plot pole-zero plot

```
% PLOT POLE-ZERO PLOT
figure;
pzplot(b2, a2);
axis equal;
```




## 3 (j) Answer question

```
% The pole=zero plot shows a high pass filtering system since all the
% zeros are around the 0 radian line, and the system is FIR since there is
% only one pole at the origin. Additionally the system is stable because
% the poles are only inside the unit circle.
```

## 3 (k) Submit file on Canvas

```
% PLAY MUSIC

disp('Playing Original Music ... ')
soundsc(x, fs)
pause(length(x)/fs*1.1)

disp('Playing Filtered Music ... ')
```

```
soundsc(y, fs)

% SAVE RESULT

% NORMALIZE THE SIGNAL TO AVOID CLIPPING
audiowrite('q3.wav', y./max(abs(y)), fs)
save('q3.mat', 'b2', 'a2');
```

```
Playing Original Music ...
Playing Filtered Music ...
```

## ALL FUNCTIONS SUPPORTING THIS CODE %% function pzplot(b,a)

PZPLOT(B,A) plots the pole-zero plot for the filter described by vectors A and B. The filter is a "Direct Form II Transposed" implementation of the standard difference equation:

```
a(1)*y(n) = b(1)*x(n) + b(2)*x(n-1) + ... + b(nb+1)*x(n-nb)
                      - a(2)*y(n-1) - ... - a(na+1)*y(n-na)
```

```
    % MODIFY THE POLYNOMIALS TO FIND THE ROOTS
    b1 = zeros(max(length(a),length(b)),1); % Need to add zeros to get the right roots
    a1 = zeros(max(length(a),length(b)),1); % Need to add zeros to get the right roots
    b1(1:length(b)) = b;    % New a with all values
    a1(1:length(a)) = a;    % New a with all values

    % FIND THE ROOTS OF EACH POLYNOMIAL AND PLOT THE LOCATIONS OF THE ROOTS
    h1 = plot(real(roots(a1)), imag(roots(a1)));
    hold on;
    h2 = plot(real(roots(b1)), imag(roots(b1)));
    hold off;

    % DRAW THE UNIT CIRCLE
    circle(0,0,1)

    % MAKE THE POLES AND ZEROS X's AND O's
    set(h1, 'LineStyle', 'none', 'Marker', 'x', 'MarkerFaceColor','none', 'linewidth', 1.5, 'markersize', 8);
    set(h2, 'LineStyle', 'none', 'Marker', 'o', 'MarkerFaceColor','none', 'linewidth', 1.5, 'markersize', 8);
    axis equal;

    % DRAW VERTICAL AND HORIZONTAL LINES
    xminmax = xlim();
    yminmax = ylim();
```

```
    line([xminmax(1) xminmax(2)],[0 0], 'linestyle', ':', 'linewidth', 0.5, 'color', [1 1 1]*.1)
    line([0 0],[yminmax(1) yminmax(2)], 'linestyle', ':', 'linewidth', 0.5, 'color', [1 1 1]*.1)

    % ADD LABELS AND TITLE
    xlabel('Real Part')
    ylabel('Imaginary Part')
    title('Pole-Zero Plot')

end


function circle(x,y,r)
% CIRCLE(X,Y,R)  draws a circle with horizontal center X, vertical center
% Y, and radius R.
%

    % ANGLES TO DRAW
    ang=0:0.01:2*pi;

    % DEFINE LOCATIONS OF CIRCLE
    xp=r*cos(ang);
    yp=r*sin(ang);

    % PLOT CIRCLE
    hold on;
    plot(x+xp,y+yp, ':', 'linewidth', 0.5, 'color', [1 1 1]*.1);
    hold off;

end



function H = DTFT(x,w)
% DTFT(X,W)  compute the Discrete-time Fourier Transform of signal X
% acroess frequencies defined by W.

    H = zeros(length(w),1);
    for nn = 1:length(x)
        H = H + x(nn).*exp(-1j*w.'*(nn-1));
    end

end
```

```
function xs = shift(x, s)
% SHIFT(x, s) shifts signal x by s such that the output can be defined by
% xs[n] = x[n - s]

    % INITIALIZE THE OUTPUT
    xs = zeros(length(x), 1);

    for n = 1:length(x)
        % CHECK IF THE SHIFT IS OUT OF BOUNDS FOR THIS SAMPLE
        if n-s > 0 && n-s < length(x)
            % SHIFT DATA
            xs(n) = x(n-s);
        end
    end

end




function [b,a] = pz2ba(p,z)
% PZ2BA(P,Z)  Converts poles P and zeros Z to filter coefficients
%             B and A
%
% Filter coefficients are defined by:
%    a(1)*y(n) = b(1)*x(n) + b(2)*x(n-1) + ... + b(nb+1)*x(n-nb)
%                          - a(2)*y(n-1) - ... - a(na+1)*y(n-na)
%

    % CONVERT ROOTS (POLES AND ZEROS) INTO POLYNOMIALS
    b = poly(z);
    a = poly(p);

end


function [p,z] = ba2pz(b,a)
% BA2PZ(B,A)  Converts filter coefficients B and A into poles P and zeros Z
%
% Filter coefficients are defined by:
%    a(1)*y(n) = b(1)*x(n) + b(2)*x(n-1) + ... + b(nb+1)*x(n-nb)
%                          - a(2)*y(n-1) - ... - a(na+1)*y(n-na)
%

    % MODIFY THE POLYNOMIALS TO FIND THE ROOTS
```

```
    b1 = zeros(max(length(a),length(b)),1); % Need to add zeros to get the right roots
    a1 = zeros(max(length(a),length(b)),1); % Need to add zeros to get the right roots
    b1(1:length(b)) = b;    % New a with all values
    a1(1:length(a)) = a;    % New a with all values

    % FIND THE ROOTS OF EACH POLYNOMIAL AND PLOT THE LOCATIONS OF THE ROOTS
    p = real(roots(a1))+1j*imag(roots(a1));
    z = real(roots(b1))+1j*imag(roots(b1));

end
```

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