Contents

QUESTION 1 COMMENTING

```
% DO NOT REMOVE THE LINE BELOW
% MAKE SURE 'eel3135_lab05_comment.m' IS IN SAME DIRECTORY AS THIS FILE
clear; close all;
type('eel3135_lab05_comment.m')
```

```

%% USER-DEFINED VARIABLES

w = -pi:(pi/100):pi;
% <-- Answer: Why is w from -pi to pi?
%
% The variable 'w' represents the normalized radian frequency range for the frequency response analysis.
% It is defined from -pi to pi to cover the full spectrum of frequencies for periodic signals,
% allowing for the examination of both positive and negative frequencies, which is important in
% signal processing to understand the behavior of filters across the entire frequency range.

%% HIGHPASS FILTER

% FREQUENCY RESPONSE
H2 = (1-exp(-1j*w*1));
% <-- Answer: What is the difference equation for this frequency response?

% ------------------->
% The difference equation corresponding to this frequency response can be derived from the
% transfer function H(z) = 1 - z^(-1). In the time domain, this can be expressed as
% y[n] = x[n] - x[n-1], where y[n] is the output and x[n] is the input signal.
% This indicates that the output is the difference between the current input and the previous input,
% allowing high-frequency components to pass while attenuating low-frequency components.
% <----------------------

% PLOT
figure;
subplot(2,1,1)
plot(w,abs(H2));    % ==> What does the abs() function do?
                    % The abs() function computes the magnitude of the complex frequency response H2.
                    % It returns the absolute value of each element in H2, which represents the amplitude of the
                    % frequency response at each frequency 'w'. This is important for understanding how much
                    % of each frequency component is passed through the filter.
                    % <==
grid on;
title('Magnitude Response')
xlabel('Normalized Radian Frequency');
ylabel('Amplitude');
subplot(2,1,2)
```

```
plot(w,angle(H2));  % ==> What does the angle() function do?
                    % The angle() function calculates the phase angle (in radians) of the complex frequency response H2.
                    % It returns the angle of each element in H2, which indicates the phase shift introduced by the
                    % filter at each frequency 'w'. This is crucial for understanding how the filter affects the timing
                    % of different frequency components in the input signal.
                    % <==
grid on;
title('Phase Response')
xlabel('Normalized Radian Frequency');
ylabel('Phase');

% <-- Answer: If you input a DC value into a highpass filter, what will be
%             its amplitude?
%
% The amplitude of a DC value (0 Hz frequency) input into a highpass filter will be 0.
% This is because highpass filters are designed to attenuate low-frequency signals, including
% DC components, effectively blocking them and allowing only higher frequency signals to pass through.
```

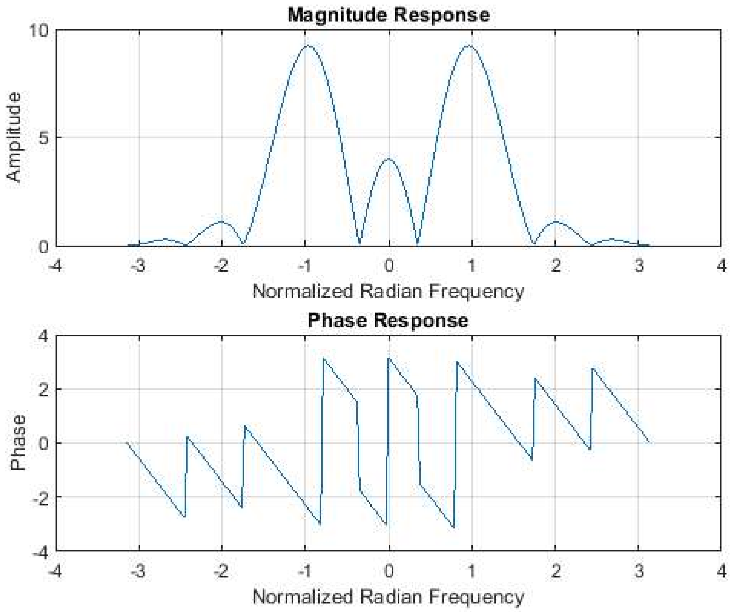
QUESTION 2 FREQUENCY FILTERING

2(a) FILL IN CODE

----------- Fill in FreqResponse function down below --------------

2(b) PLOT FREQUENCY RESPONSE

```
% Define filter coefficients
b = [-1, -2, -1, 1, 2, 1, -1, -2, -1];
w = -pi:(pi/100):pi; % Frequency range

% Generate frequency response
H = FreqResponse(b, w);

% Plot magnitude and phase responses
figure;
subplot(2,1,1);
plot(w, abs(H)); % Magnitude response
grid on;
title('Magnitude Response');
xlabel('Normalized Radian Frequency');
ylabel('Amplitude');

subplot(2,1,2);
plot(w, angle(H)); % Phase response
grid on;
title('Phase Response');
xlabel('Normalized Radian Frequency');
ylabel('Phase');
```



2(c) EVALUATE FREQUENCY RESPONSE FOR CERTAIN FREQUENCIES

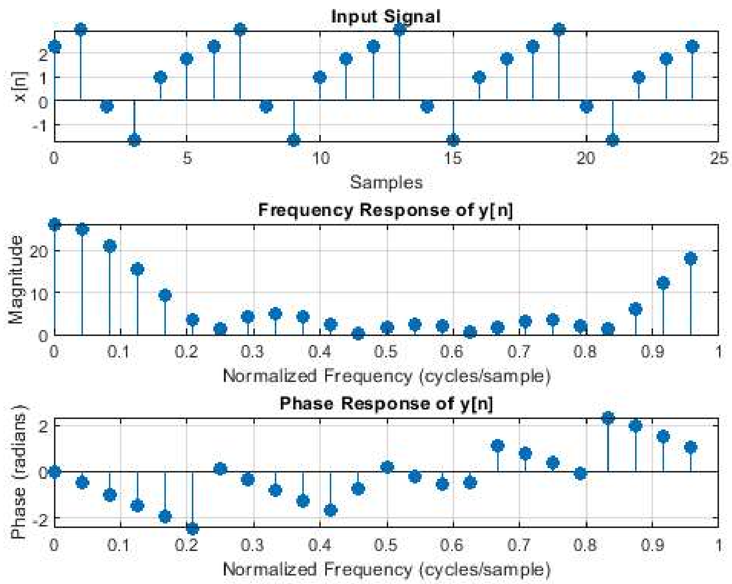
```
frequencies = [0, pi/3, 2*pi/3];
for wb = frequencies
    H_wb = FreqResponse(b, wb);
    disp(['H(' num2str(wb) ') = ' num2str(H_wb)]);
end
```

```
H(0) = -4
H(1.0472) = -4.5+7.7942i
H(2.0944) = 0.5+0.86603i
```

2(d) COMPUTE AND PLOT OUTPUT

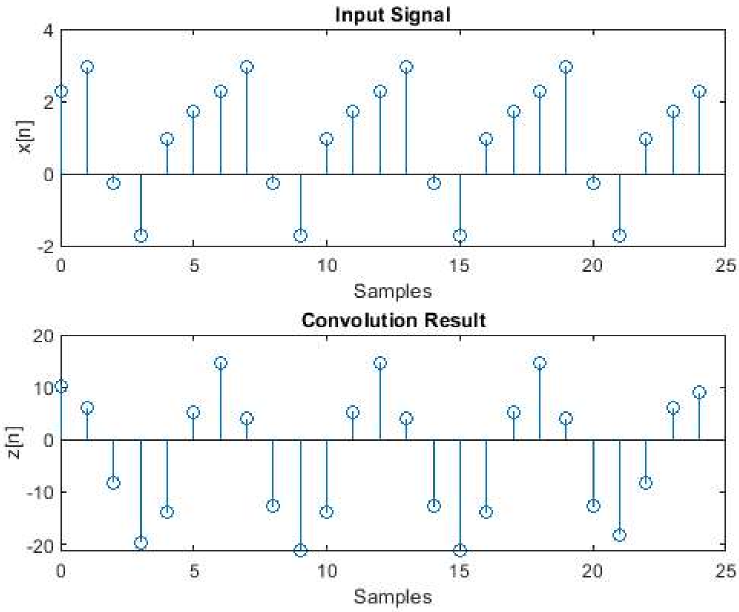
```
n = 0:24; % Define range for n
x = 1 + 2*cos((pi/3)*n) - cos((2*pi/3)*n + pi/4); % Input signal

% Compute the frequency response of y[n]
N = 24; % Number of points for FFT
f = (0:N-1)*(1/N); % Frequency vector

% Compute output signal y[n] by convolving x[n] with the FIR filter
y = FreqResponse(x, f);

% Plot x[n]
figure;
subplot(3,1,1);
stem(n, x, 'filled');
xlabel('Samples');
ylabel('x[n]');
title('Input Signal');
grid on;

% Plot frequency response of y[n]
subplot(3,1,2);
stem(f, abs(y), 'filled'); % Magnitude response
xlabel('Normalized Frequency (cycles/sample)');
ylabel('Magnitude');
title('Frequency Response of y[n]');
grid on;

% Phase Response of y[n]
subplot(3,1,3);
stem(f, angle(y), 'filled'); % Phase response
xlabel('Normalized Frequency (cycles/sample)');
ylabel('Phase (radians)');
```

```
title('Phase Response of y[n]');
grid on;
```



2(e) PLOT OUTPUTS

```
% Using conv function to compute z[n]
z = conv(x, b, 'same');

% Plot x[n] and z[n]
figure;
subplot(2,1,1);
stem(n, x);
xlabel('Samples');
ylabel('x[n]');
title('Input Signal');

subplot(2,1,2);
stem(n, z);
xlabel('Samples');
ylabel('z[n]');
title('Convolution Result');
```



2(f) ANSWER QUESTION

```
% y[n] is the output from the filter applied to x[n], while z[n] is the result of convolution.
% Differences may arise due to boundary effects in convolution or differences in how the filter is applied.
```

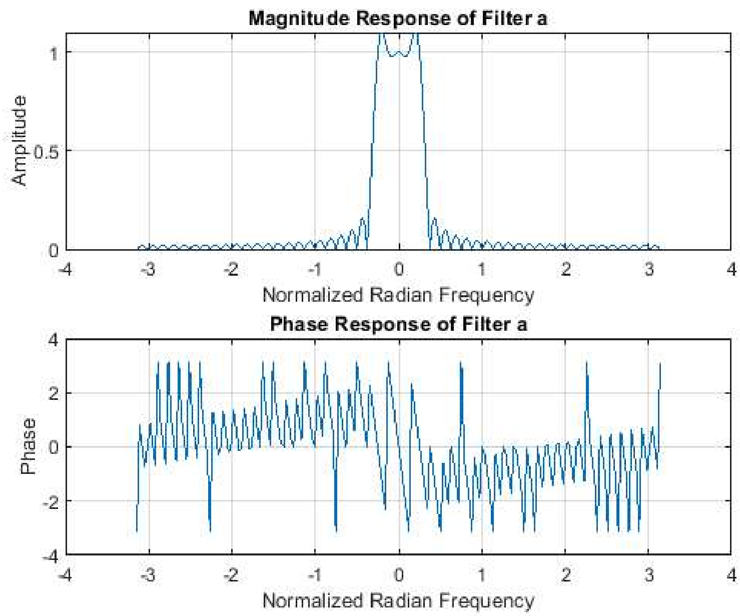
QUESTION 3

```
% DO NOT REMOVE THE LINE BELOW
% MAKE SURE 'lite_saturation.wav' IS IN SAME DIRECTORY AS THIS FILE
% Audio from: https://freemusicarchive.org/music/lite-saturation/
[x, fs] = audioread('lite_saturation.wav');
```

3(a) PLOT FREQUENCY RESPONSE

```
a = [  ...
    0.0200    0.0191    0.0163    0.0120    0.0064         0   -0.0066   -0.0130 ...
   -0.0185   -0.0225   -0.0247   -0.0247   -0.0222   -0.0172   -0.0097         0 ...
    0.0115    0.0244    0.0380    0.0517    0.0647    0.0765    0.0863    0.0938 ...
    0.0984    0.1000    0.0984    0.0938    0.0863    0.0765    0.0647    0.0517 ...
    0.0380    0.0244    0.0115         0   -0.0097   -0.0172   -0.0222   -0.0247 ...
   -0.0247   -0.0225   -0.0185   -0.0130   -0.0066         0    0.0064    0.0120 ...
    0.0163    0.0191
    ];

% Compute frequency response for filter coefficients a
Ha = FreqResponse(a, w);

% Plot magnitude and phase responses
figure;
subplot(2,1,1);
plot(w, abs(Ha));
grid on;
title('Magnitude Response of Filter a');
xlabel('Normalized Radian Frequency');
ylabel('Amplitude');

subplot(2,1,2);
plot(w, angle(Ha));
grid on;
title('Phase Response of Filter a');
xlabel('Normalized Radian Frequency');
ylabel('Phase');

% <== ANSWER TO QUESTION ==>
%
```

```
% The filter a is a lowpass filter because it allows low frequencies to
% pass while attenuating high frequencies.
```



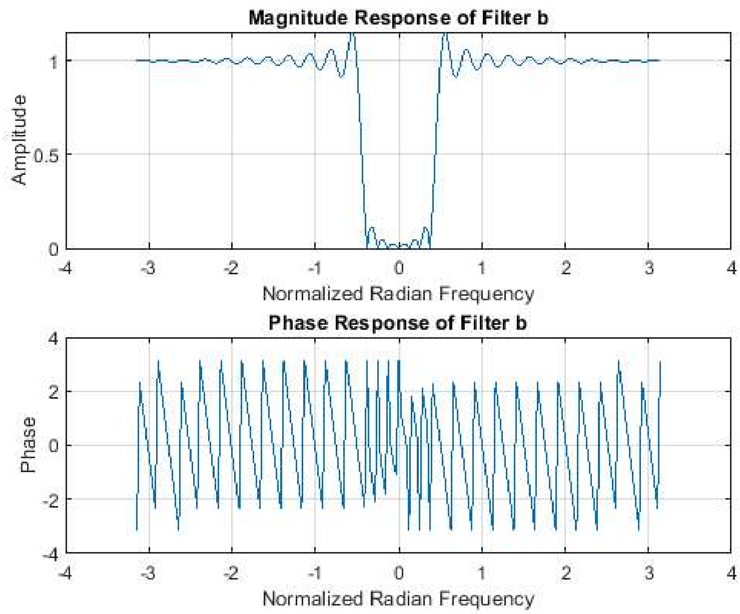
3(b) APPLY FILTER

```
% Play original sound
soundsc(x, fs);

% Apply filter a on x
xa = conv(x, a, 'same');

% Play filtered sound
soundsc(xa, fs);

% <==== ANSWER TO QUESTION ====>
%
% The original sound x and the filtered sound xa differ in that xa has
% reduced high-frequency content due to the lowpass filtering effect.
```

3(c) PLOT FREQUENCY RESPONSE

```
b = [ ...
    0.0200    0.0181    0.0128    0.0051   -0.0039   -0.0124   -0.0188   -0.0221 ...
   -0.0212   -0.0162   -0.0076    0.0033    0.0147    0.0247    0.0311    0.0324 ...
    0.0272    0.0153   -0.0029   -0.0262   -0.0524   -0.0790   -0.1034   -0.1230 ...
   -0.1356    0.8600   -0.1356   -0.1230   -0.1034   -0.0790   -0.0524   -0.0262 ...
   -0.0029    0.0153    0.0272    0.0324    0.0311    0.0247    0.0147    0.0033 ...
   -0.0076   -0.0162   -0.0212   -0.0221   -0.0188   -0.0124   -0.0039    0.0051 ...
    0.0128    0.0181
    ];

% Compute frequency response for filter coefficients b
Hb = FreqResponse(b, w);

% Plot magnitude and phase responses
figure;
subplot(2,1,1);
plot(w, abs(Hb));
grid on;
title('Magnitude Response of Filter b');
xlabel('Normalized Radian Frequency');
ylabel('Amplitude');

subplot(2,1,2);
plot(w, angle(Hb));
grid on;
title('Phase Response of Filter b');
```

```
xlabel('Normalized Radian Frequency');
ylabel('Phase');

% <==== ANSWER TO QUESTION ====>
%
% The filter b is a bandstop filter because it attenuates frequencies in a
% specific range while allowing others to pass.
```



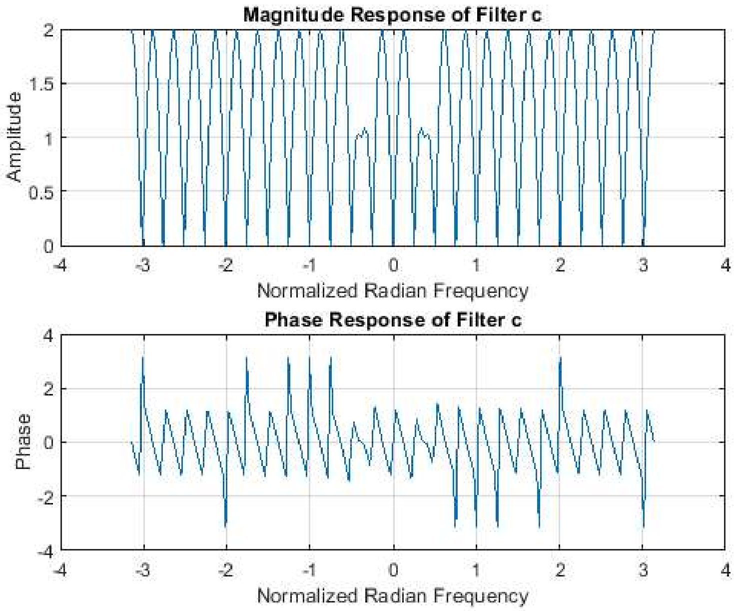
3(d) APPLY FILTER

```
% Apply filter b on x
xb = conv(x, b, 'same');

% Play filtered sound
soundsc(xb, fs);

% <==== ANSWER TO QUESTION ====>
%
% The original sound x and the filtered sound xb differ significantly.
% The bandstop filter b removes certain frequency components from the original sound, resulting in a sound that lacks those frequencies.
```

3(e) PLOT FREQUENCY RESPONSE

```
% Compute new filter coefficients c
c = [1, zeros(1, length(b)-1)] - b - a; % c[n] = δ[n] - b[n] - a[n]
Hc = FreqResponse(c, w); % Compute frequency response for c

% Plot magnitude and phase responses
figure;
subplot(2,1,1);
plot(w, abs(Hc));
grid on;
title('Magnitude Response of Filter c');
xlabel('Normalized Radian Frequency');
ylabel('Amplitude');

subplot(2,1,2);
plot(w, angle(Hc));
grid on;
title('Phase Response of Filter c');
xlabel('Normalized Radian Frequency');
ylabel('Phase');

% <==== ANSWER TO QUESTION ====>
%
% The filter c is a highpass filter because it allows high frequencies to
% pass while attenuating low frequencies.
```

```
%
% This result occurs because the filter coefficients are designed to
% subtract the effects of the lowpass filters a and b from the input signal.
```

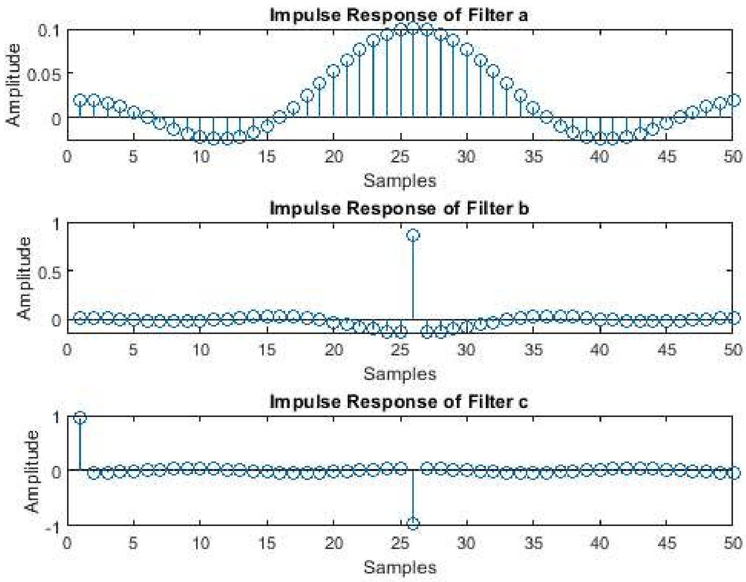


3(f) APPLY FILTER

```
% Apply filter c on x
xc = conv(x, c, 'same');

% Play filtered sound
soundsc(xc, fs);

% <==== ANSWER TO QUESTION ====>
%
% The original sound x and the filtered sound xc differ in that xc emphasizes higher frequency components while attenuating lower frequencies.
% This is consistent with the highpass nature of filter c, which removes low-frequency content from the audio signal.
```

3(g) ANSWER QUESTION

```
% Plot impulse responses of the three filters
figure;
subplot(3,1,1);
stem(a);
title('Impulse Response of Filter a');
xlabel('Samples');
ylabel('Amplitude');

subplot(3,1,2);
stem(b);
title('Impulse Response of Filter b');
xlabel('Samples');
ylabel('Amplitude');

subplot(3,1,3);
stem(c);
title('Impulse Response of Filter c');
xlabel('Samples');
ylabel('Amplitude');

% <==== ANSWER TO QUESTION ====>
%
% The distinguishing difference between these three impulse responses is their shape and the frequency characteristics they represent.
% Filter a has a lowpass characteristic, filter b has a bandstop characteristic, and filter c has a highpass characteristic.
```



ALL FUNCTIONS SUPPORTING THIS CODE %%

```
function H = FreqResponse(b,w)
%  ===> FreqResponse computes the frequency response of an FIR filter.
    % Inputs:
    %   b - vector of filter coefficients
    %   w - vector of angular frequencies
    % Output:
    %   H - complex-valued frequency response <===

    M = length(b) - 1; % Order of the filter
    H = zeros(size(w)); % Initialize frequency response

    for k = 0:M
        H = H + b(k+1) * exp(-1j * w * k); % Compute frequency response
    end

end
```

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