SIO 207A Final Project Report

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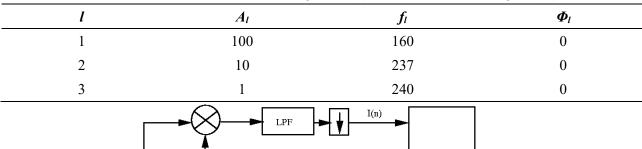
I. Data Set

This report provides an implementation of the complex basebanding and Fast Fourier Transform (FFT) architecture as shown in Figure 1. A discrete-time signal x(n) with 4096 data points (n = 0, 1, 2...4095) is selected for analysis. The discrete-time signal x(n) has the analog form x(t) as defined in the following equation:

$$x(t) = \sum_{l=1}^{3} A_{l} \cos(2\pi f_{l} t + \phi_{l})$$

where the parameters are defined in Table 1. The sampling frequency of the signal is $f_s = 1000$ Hz.

Table 1. Parameters used in the analog form x(t) for the discrete-time signal x(n).



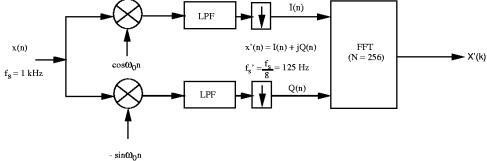


Figure 1. Complex basebanding and Fast Fourier Transform (FFT) architecture.

The first 256 data points of the discrete-time signal x(n) are plotted in Figure 2. The logarithmic magnitude of its Fourier Transform using a Kaiser-Bessel window (α =2.5 or β =7.85) with NFFT = 256 is also presented in Figure 2. The analog frequency bin width of the FFT is 3.906 Hz, which is calculated based on the procedures as presented in Appendix 1. The bin indices for each analog frequency component are summarized in Table 2. These analog frequency components reside in 4 different frequency bins, which correspond to the 4 different peaks in the FFT magnitude plot. Because of the large analog frequency bin width (3.906 Hz), frequency components f_2 and f_3 both reside in bin #61. The two frequency components cannot be identified separately, so only one peak is observed for frequency components f_2 and f_3 (similarly $-f_2$ and $-f_3$ are also observed as one peak). Given the sampling frequency of 1000 Hz and NFFT = 256, the frequency bin indices for f_2 and f_3 are always 61, which is not affected by the window applied to the signal. Therefore, rectangular windowed FFT will not indicate distinct spectral peaks for f_2 and f_3 .

Table 2. Frequency components of the original signal.

f_l [Hz]	Bin Index	$ X(f_l) $ [dB]	f_l [Hz]	Bin Index	$ X(f_i) $ [dB]
160	41	0	-160	-41	0
237	61	-20	-237	-61	-20
240	61	-20	-240	-61	-20

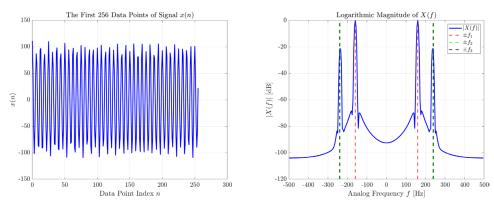


Figure 2. The first 256 data points of signal x(n) and the logarithmic magnitude of its Fourier Transform X(f).

II. Decimation Filter Design

A decimation filter is designed based on the equiripple FIR filter design algorithm. The filter is a 64-coefficient linear phase FIR low-pass filter with passband cutoff frequency of 40 Hz and stopband cutoff frequency of 85 Hz. The passband/stopband weight ratio of the filter is 50. Figure 3 shows the impulse response h(n) and frequency response H(f) of the decimation filter. The frequency response H(f) is calculated by applying the Fourier Transform on the impulse response h(n) using a rectangular window with NFFT = 1024. An expanded view of H(f) with analog frequency -40 Hz $\leq f \leq$ 40 Hz (within passband frequency range) is also presented in Figure 3. Smooth magnitude is observed for the passband magnitude. The magnitude of the passband ripple is less than 4 dB.

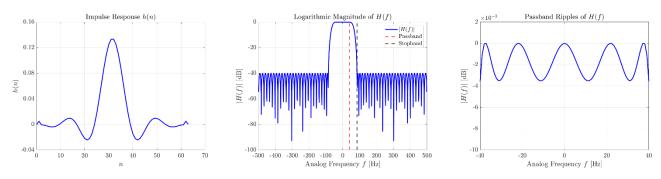


Figure 3. Impulse response and frequency response of the FIR low-pass filter.

III. Complex Basebanding and Desampling

The complex multiplication $e^{-j \, \omega n} x(n)$ is implemented where $\omega = 2\pi (f_0/f_s)$ and the center frequency $f_0 = 250$ Hz. The Fourier Transform of the first 256 data points in new complex sequence is calculated using a Kaiser Bessel window (α =2.5 or β =7.85) with NFFT = 256. The logarithmic magnitude of the Fourier Transform is shown in Figure 4 (left). The frequency components for the signal after complex basebanding are presented in Table 3. It is observed that the frequency component after complex basebanding f_i' can be calculated from the frequency component of the original signal f_i and the center frequency f_0 based on the following equation (detailed calculations are presented in Appendix 1):

$$f_l$$
'= f_l-f_0

The mathematical derivation of this frequency shift is presented as follows. Signal after complex basebanding:

$$\tilde{x}(n) = x(n) \cdot e^{-j2\pi (f_0/f_s)n}$$

Fourier Transform of the original signal with frequency component fi:

$$X(f_l) = \sum_{n=-\infty}^{\infty} x(n) \cdot e^{-j2\pi (f_l/f_s)n}$$

Fourier Transform of the signal after complex basebanding:

$$\widetilde{X}(f) = \sum_{n = -\infty}^{\infty} \widetilde{x}(n) \cdot e^{-j2\pi(f_l/f_s)n} = \sum_{n = -\infty}^{\infty} x(n) \cdot e^{-j2\pi(f_0/f_s)n} \cdot e^{-j2\pi(f_l/f_s)n} = \sum_{n = -\infty}^{\infty} x(n) \cdot e^{-j2\pi(f_0+f_l/f_s)n}$$

$$\widetilde{X}(f) = X(f_l + f_0)$$

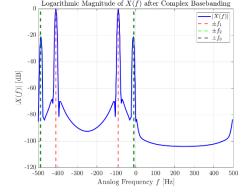
Therefore, the spectrum of the signal translates to the left with the magnitude of f_0 after complex basebanding. Table 3. Frequency components of the signal after complex basebanding.

f_l [Hz]	f_l' [Hz]	$ X(f_l') $ [dB]	f_l [Hz]	f_l' [Hz]	$ X(f_i') $ [dB]
160	-90	0	-160	-410	0
237	-13	-20	-237	-487	-20
240	-10	-20	-240	-490	-20

The sequence after complex basebanding is filtered by the FIR low-pass filter designed in Section II. The filtered signal is obtained as y(n). The Fourier Transform of the windowed filtered signal y(n) (n = 256, 257...511) is calculated using a Kaiser Bessel window (α =2.5 or β =7.85) with NFFT = 256. The logarithmic magnitude of the Fourier Transform is shown in Figure 4 (right). The frequency components of the signal y(n) are presented in Table 4. The frequency values are the same compared to the signal before filtering because filtering does not shift the frequency of the signal. However, the magnitude of the peak values outside the passband frequency range (-40 Hz $\leq f \leq$ 40 Hz) are attenuated compared to the signal before filtering. Detailed calculations for the frequency components of the signal y(n) are listed in Appendix 1.

Table 4. Frequency components of the signal after complex basebanding and low-pass filtering.

f_l [Hz]	<i>f</i> _l ' [Hz]	$ Y(f_i') $ [dB]	f_l [Hz]	f_{l}' [Hz]	Y(fi') [dB]
160	-90	-20	-160	-410	-20
237	-13	0	-237	-487	-40
240	-10	0	-240	-490	-40



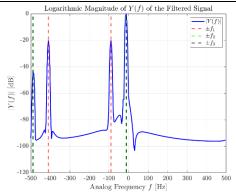


Figure 4. Logarithmic magnitude of the Fourier Transform for the complex baseband signal before (left) and after (right) passing the FIR low-pass filter.

IV. High Resolution Spectral Analysis

The complex filtered sequence is desampled by a factor of 8. The desampled sequence x'(n) has a new sampling frequency $f_s' = 125$ Hz. The Fourier Transform of the windowed filtered signal x'(n) (n = 256, 257...511) is calculated using a Kaiser Bessel window (α =2.5 or β =7.85) with NFFT = 256. The logarithmic magnitude of the Fourier Transform is shown in Figure 5. The analog frequency bin width of the FFT is 0.488 Hz, which is 1/8 of the frequency bin width for the FFT of the original signal. The frequency values corresponding to the peaks of |X'(f)| are presented in Table 5. Detailed calculations are presented in Appendix 1. Since the absolute values of some frequency components are greater than the Nyquist frequency ($f_s'/2 = 62.5$ Hz), aliasing is observed in the Fourier transform magnitude plot.

T 1	1 7 17	4	C .1	. 1	C	1	1 1 1.	1	c	1	1	1	1.
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f_l [Hz]	f_l' [Hz]	$ X'(f_l') $ [dB]	f_l [Hz]	f_i' [Hz]	$ X'(f_l') $ [dB]
160	35	-20	-160	-35	-20
237	-13	0	-237	13	-40
240	-10	-20	-240	10	-60

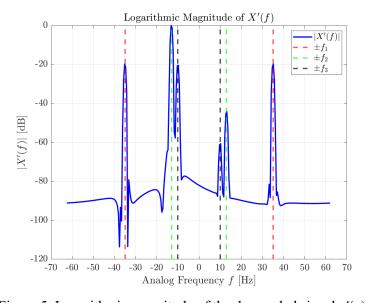


Figure 5. Logarithmic magnitude of the desampled signal x'(n).

A second iteration in the design of the FIR low-pass filter is conducted. All requirements are to remain as originally given except for the passband/stopband weight ratio. Figure 6 and Figure 7 present the analysis result for the filter with passband/stopband weight ratio of 10 and 100 (the original weight ratio is 50). It is observed that a lower weight ratio has larger passband ripples and lower Fourier Transform magnitude in the stopband (larger stopband attenuation). In contrast, a higher weight ratio has smaller passband ripples and larger Fourier Transform magnitude in the stopband (smaller stopband attenuation). In practice, the selection of the passband/stopband weight ratio depends on the required behavior of the filter. A lower weight ratio is suggested if we want to remove the signal within the stopband (e.g. noise within the stopband has a significant effect on the results) regardless of the potential error introduced to the signal within the passband because of the large passband ripples. A higher weight ratio is suggested if we want to keep an accurate and smooth signal within the passband regardless of the potential effect from the signal within the stopband because of the smaller stopband attenuation.

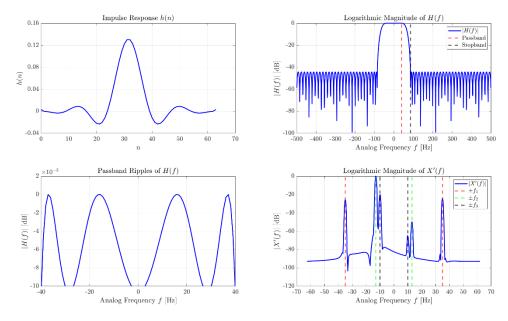


Figure 6. Analysis result for the case of passband/stopband weight ratio = 10.

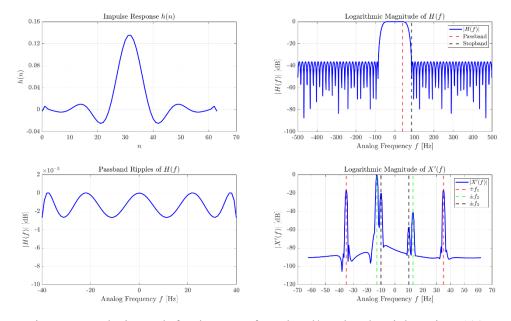


Figure 7. Analysis result for the case of passband/stopband weight ratio = 100.

Appendix 1. Hand Calculation

I. Data Set

$$f_1 \coloneqq 160 \; \textit{Hz}$$
 $f_2 \coloneqq 237 \; \textit{Hz}$ $f_3 \coloneqq 240 \; \textit{Hz}$ Analog frequency components of the signal

$$A_1 = 100$$
 $A_2 = 10$ Linear amplitude for each analog frequency component

$$f_s = 1000 \; Hz$$
 Sampling frequency

$$NFFT = 256$$

$$BinWidth := \frac{f_s}{NEFT} = 3.906 \; Hz$$
 Analog frequency bin width of the FFT

$$Bin_{f_1} = \text{round}\left(\frac{f_1}{BinWidth}\right) = 41$$
 Frequency bin index of analog frequency component f_1

$$Bin_{f_2} = \text{round}\left(\frac{f_2}{BinWidth}\right) = 61$$
 Frequency bin index of analog frequency component f_2

$$Bin_{f_3} = \text{round}\left(\frac{f_3}{BinWidth}\right) = 61$$
 Frequency bin index of analog frequency component f_3

Given the sampling frequency of 1000 Hz and NFFT = 256, the frequency bin indices for f_2 and f_3 are always 61, which is not affected by the window applied to the signal. Therefore, rectangular windowed FFT will not indicate distinct spectral peaks for f_2 and f_3 .

$$20 \cdot \log (A_1) = 40$$
 Magnitude of the peak corresponding to +/-fi

$$20 \cdot \log (A_2) + 20 \cdot \log (A_3) = 20$$
 Magnitude of the peak corresponding to $+/-f_2$ and $+/-f_3$ (same frequency bin)

Therefore, after the normalization, the magnitude of the peak corresponding to $\pm f$ is 0 dB and the magnitude of the peak corresponding to $\pm f$ (same frequency bin) is -20 dB.

III. Complex Basebanding and Desampling

1. Complex basebanding:

$$f_0 = 250 \; Hz$$
 Center frequency

Frequency components after complex basebanding:

$$f_1' := f_1 - f_0 = -90 \; Hz$$
 $f_2' := f_2 - f_0 = -13 \; Hz$ $f_3' := f_3 - f_0 = -10 \; Hz$

$$f_1'' \coloneqq -f_1 - f_0 = -410 \; \textbf{\textit{Hz}} \qquad \qquad f_2'' \coloneqq -f_2 - f_0 = -487 \; \textbf{\textit{Hz}} \qquad \qquad f_3'' \coloneqq -f_3 - f_0 = -490 \; \textbf{\textit{Hz}}$$

Complex basebanding does not change the magnitude of the signal. Therefore, the magnitude of peaks corresponding to each frequency component remain the same as previous.

2. Low-pass filtering:

Low-pass filtering does not shift the frequency of the signal. Therefore, frequency values of the signal remain the same as previous.

$$20 \cdot \log (A_1) - 40 = 0$$
 Magnitude of the peak corresponding to f_1
$$20 \cdot \log (A_2) + 20 \cdot \log (A_3) = 20$$
 Magnitude of the peak corresponding to f_2 and f_3 (same frequency bin)

$$20 \cdot \log (A_1) - 40 = 0$$
 Magnitude of the peak corresponding to -fi

$$20 \cdot \log(A_2) + 20 \cdot \log(A_3) - 40 = -20$$
 Magnitude of the peak corresponding to $-f_2$ and $-f_3$ (same frequency bin)

Therefore, after the normalization, the magnitude of the peak corresponding to f_1 is -20 dB; the magnitude of the peak corresponding to f_2 and f_3 (same frequency bin) is 0 dB; the magnitude of the peak corresponding to $-f_1$ is -20 dB; the magnitude of the peak corresponding to $-f_2$ and $-f_3$ (same frequency bin) is -40 dB;

IV. High Resolution Spectral Analysis

$$f_s' \coloneqq \frac{f_s}{8} = 125 \; \textit{Hz}$$
 Sampling frequency after decimation
$$BinWidth \coloneqq \frac{f_s'}{NFFT} = 0.488 \; \textit{Hz}$$
 Analog frequency bin width of the FFT after desampling

Frequency components after desampling:

$$f_1' \coloneqq f_1 - f_0 + f_s' = 35$$
 Hz $f_2' \coloneqq f_2 - f_0 = -13$ Hz $f_3' \coloneqq f_3 - f_0 = -10$ Hz $f_1'' \coloneqq -f_1 - f_0 + 3$ $f_s' = -35$ Hz $f_2'' \coloneqq -f_2 - f_0 + 4$ $f_s' = 13$ Hz $f_3'' \coloneqq -f_3 - f_0 + 4$ $f_s' = 10$ Hz $20 \cdot \log (A_1) - 40 = 0$ Magnitude of the peak corresponding to f_1 (-20 dB after normalization) $20 \cdot \log (A_2) = 20$ Magnitude of the peak corresponding to f_2 (0 dB after normalization) $20 \cdot \log (A_3) = 0$ Magnitude of the peak corresponding to f_3 (-20 dB after normalization) $20 \cdot \log (A_1) - 40 = 0$ Magnitude of the peak corresponding to f_3 (-20 dB after normalization) $20 \cdot \log (A_2) - 40 = -20$ Magnitude of the peak corresponding to f_3 (-40 dB after normalization) $20 \cdot \log (A_3) - 40 = -40$ Magnitude of the peak corresponding to f_3 (-60 dB after normalization)

Appendix 2. MATLAB Code

```
1 % SIO 207A Final Project
 2 % Ruipu Ji
 4 % Initialization and default plot settings.
 5 clear; clc; close all;
 7 set(0, 'DefaultAxesFontSize', 15);
 8 set(0, 'DefaultTextFontSize', 15);
10 set(0, 'DefaultTextInterpreter', 'latex');
11 set(0, 'DefaultLegendInterpreter', 'latex');
12 set(0, 'DefaultAxesTickLabelInterpreter', 'latex');
13
14 %% Data Set.
15 % Generate a 4096-point discrete-time signal x(n). ------
16 A = [100 10 1]; % Magnitude.
17 f = [160 237 240]; % Analog frequency [Hz].
18 Phi = [0 \ 0 \ 0]; % Phase angle [rad].
20 fs = 1000; % Sampling frequency [Hz].
21 n = 4096; % Number of data points.
22 t = (0:1:n-1)'*1/fs; % Time vector.
23
24 x = zeros(n,1); % Initialize the signal sequence as an empty array.
26 % Generate the discrete-time signal sequence x(n).
27 for idx = 1:size(A, 2)
      x = x + A(idx) *cos(2*pi*f(idx)*t+Phi(idx));
29 end
30
31 % Plot the first 256 data points of the signal x(n). -----
32 NFFT = 256;
33
34 figure('Position', [0, 0, 1800, 600]);
35
36 subplot(1,2,1);
37 hold on;
38 plot(0:1:NFFT-1, x(1:NFFT), 'b', 'LineWidth', 2);
39 grid on;
40 box on;
41 xlim([0 300]);
42 xticks(0:50:300);
43 ylim([-150 150]);
44 yticks(-150:50:150);
45 xlabel('Data Point Index $n$');
46 ylabel('$x(n)$');
47 title('The First 256 Data Points of Signal $x(n)$');
49 % 256-point FFT of the signal using the Kaiser-Bessel window. -----
50 % Kaiser-Bessel window (alpha = 2.5 or beta = alpha*pi = 7.85).
51 KaiserBesselWindow = kaiser(NFFT, 7.85);
53 % Window the first 256 data points of the signal x(n).
54 x 256 KB = x(1:NFFT) .* KaiserBesselWindow;
56 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
```

```
magnitude.
 57 X 256 KB magnitude = 20*log10(abs(fftshift(fft(x 256 KB, NFFT))));
 58 X 256 KB magnitude = X 256 KB magnitude - max(X 256 KB magnitude); % Normalize the ✓
results.
 59
 60 % Calculate bin width of the FFT.
 61 BinWidth = fs/NFFT;
 63 % Calculate the frequency bin index for analog frequencies.
 64 f idx = zeros(1, size(f,2));
 66 for idx = 1:size(f idx, 2)
             f idx(1,idx) = round(f(idx)/BinWidth);
 69
 70 % Plot the result.
 71 f fft = (-fs/2:BinWidth:fs/2-BinWidth)';
 72
 73 subplot(1,2,2);
 74 hold on;
 75 plot(f fft, X 256 KB magnitude, 'b', 'LineWidth', 2);
 76 xline(f(1), 'r--', 'LineWidth', 2);
 77 xline(f(2), 'g--', 'LineWidth', 2);
 78 xline(f(3), 'k--', 'LineWidth', 2);
 79 xline(-f(1), 'r--', 'LineWidth', 2);
 80 xline(-f(2), 'g--', 'LineWidth', 2);
 81 xline(-f(3), 'k--', 'LineWidth', 2);
 82 grid on;
 83 box on;
 84 xlim([-500 500]);
 85 xticks(-500:100:500);
 86 ylim([-120 0]);
 87 yticks(-120:20:0);
 88 xlabel('Analog Frequency $f$ [Hz]');
 89 ylabel('$|X(f)|$ [dB]');
 90 legend('|X(f)|', '|X(f)|', '|X(f)
 91 title('Logrithmic Magnitude of $X(f)$');
 92
 93 exportgraphics(gcf, 'Figure1-PartI.png', 'ContentType', 'image');
 95 %% Decimation Filter Design.
 96 N coefficients = 64; % Number of coefficients.
 97 fc passband = 40; % Passband cutoff frequency [Hz].
 98 fc stopband = 85; % Stopband cutoff frequency [Hz].
 99 Weight = [50 1]; % Define weight ratio vector [passband stopband].
100
101 % Filter design using an equiripple FIR filter design algorithm.
102 f parameter = [0 \text{ fc passband/(fs/2) fc stopband/(fs/2) 1}];
103 a parameter = [1 1 0 0];
104 h = firpm(N coefficients-1, f parameter, a parameter, Weight)'; % Filter design.
105
106 % Calculate the frequency response of the filter.
107 NFFT filter = 1024; % NFFT for the filter calculation.
108 f filter fft = (-0.5:1/NFFT filter:0.5-1/NFFT_filter)' * fs; % Frequency vector for <math>\checkmark
the plot.
109 RectangularWindow = rectwin(NFFT filter); % Rectangular window.
```

```
110 h padded = padarray(h, [NFFT filter-size(h,1) 0], 'post'); % Pad the filter to the
same length as NFFT.
111
112 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
113 H magnitude = 20*log10(abs(fftshift(fft(h padded.*RectangularWindow, NFFT filter))));
114 H magnitude = H magnitude - max(H magnitude); % Normalize the result.
116 % Plot the result.
117 figure('Position', [0, 0, 2500, 500]);
118
119 subplot(1,3,1);
120 hold on;
121 plot(0:1:N coefficients-1, h, 'b', 'LineWidth', 2);
122 grid on;
123 box on;
124 xlim([0 70]);
125 xticks(0:10:70);
126 ylim([-0.04 0.16]);
127 yticks (-0.04:0.04:0.16);
128 xlabel('$n$');
129 ylabel('$h(n)$');
130 title('Impulse Response $h(n)$');
131
132 subplot (1, 3, 2);
133 hold on;
134 plot(f filter fft, H magnitude, 'b', 'LineWidth', 2);
135 xline(fc passband, 'r--', 'LineWidth', 2);
136 xline(fc stopband, 'k--', 'LineWidth', 2);
137 grid on;
138 box on;
139 xlim([-500 500]);
140 xticks(-500:100:500);
141 ylim([-100 0]);
142 yticks(-100:20:0);
143 xlabel('Analog Frequency $f$ [Hz]');
144 ylabel('$|H(f)|$ [dB]');
145 legend('$|H(f)|$', 'Passband', 'Stopband', 'Location', 'northeast');
146 title('Logrithmic Magnitude of $H(f)$');
147
148 subplot(1,3,3);
149 hold on;
150 plot(f filter fft, H magnitude, 'b', 'LineWidth', 2);
151 grid on;
152 box on;
153 xlim([0 40]);
154 xticks(0:10:40);
155 ylim([-0.01 0.002]);
156 yticks(-0.01:0.002:0.002);
157 xlabel('Analog Frequency $f$ [Hz]');
158 ylabel('$|H(f)|$ [dB]');
159 title('Passband Ripples of $H(f)$');
161 exportgraphics(gcf, 'Figure2-PartII.png', 'ContentType', 'image');
163 %% Complex Basebanding and Desampling.
```

```
164 % Complex multiplication on the discrete-time signal x(n). ------
165 f0 = 250; % Center frequency [Hz].
166 w0 = 2*pi*f0/fs; % Center frequency in rad.
167 x complex = zeros(size(x,1), 1); % Initialization for the complex sequence.
168
169 for idx = 1:size(x,1)
170 x \text{ complex(idx)} = \exp(-1i*w0*idx)*x(idx);
171 end
172
173 % 256-point FFT of the complex sequence. -----
174 % Window the first 256 data points of the complex sequence.
175 x complex 256 KB = x complex(1:NFFT) .* KaiserBesselWindow;
176
177 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
178 X complex 256 KB magnitude = 20*log10(abs(fftshift(fft(x complex 256 KB, NFFT))));
179 X complex 256 KB magnitude = X complex 256 KB magnitude - max ✔
(X complex 256 KB magnitude); % Normalize the results.
181 % Plot the results.
182 figure('Position', [0, 0, 1800, 600]);
183
184 subplot(1,2,1);
185 hold on;
186 plot(f_fft, X_complex_256_KB_magnitude, 'b', 'LineWidth', 2);
187 x line(f(1)-f0, 'r--', 'LineWidth', 2);
188 xline(f(2)-f0, 'g--', 'LineWidth', 2);
189 x = (f(3)-f0, 'k--', 'LineWidth', 2);
190 xline(-f(1)-f0, 'r--', 'LineWidth', 2);
191 xline(-f(2)-f0, 'g--', 'LineWidth', 2);
192 xline(-f(3)-f0, 'k--', 'LineWidth', 2);
193 grid on;
194 box on;
195 xlim([-500 500]);
196 xticks(-500:100:500);
197 ylim([-120 0]);
198 yticks(-120:20:0);
199 xlabel('Analog Frequency $f$ [Hz]');
200 ylabel('$|X(f)|$[dB]');
201 legend('\$|X(f)|\$', '\$pm f 1\$', '\$pm f 2\$', '\$pm f 3\$', 'Location', 'northeast');
202 title('Logrithmic Magnitude of $X(f)$ after Complex Basebanding');
204 % Low pass filter the complex sequence in the time domain. ------
205 y complex = filter(h, 1, x complex);
207 % Window the data points (n = 256-511) of the signal y(n).
208 y complex 256 KB = y complex(NFFT+1:2*NFFT) .* KaiserBesselWindow;
209
210 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
211 Y complex 256 KB magnitude = 20*log10(abs(fftshift(fft(y complex 256 KB, NFFT))));
212 Y_complex_256_KB_magnitude = Y_complex_256_KB_magnitude - max 🗸
(Y complex 256 KB magnitude); % Normalize the results.
213
214 % Plot the results.
215 subplot(1,2,2);
```

```
216 hold on;
217 plot(f fft, Y complex 256 KB magnitude, 'b', 'LineWidth', 2);
218 xline(f(1)-f0, 'r--', 'LineWidth', 2);
219 xline(f(2)-f0, 'g--', 'LineWidth', 2);
220 xline(f(3)-f0, 'k--', 'LineWidth', 2);
221 xline(-f(1)-f0, 'r--', 'LineWidth', 2);
222 xline(-f(2)-f0, 'g--', 'LineWidth', 2);
223 xline(-f(3)-f0, 'k--', 'LineWidth', 2);
224 grid on;
225 box on;
226 xlim([-500 500]);
227 xticks(-500:100:500);
228 ylim([-120 0]);
229 yticks(-120:20:0);
230 xlabel('Analog Frequency $f$ [Hz]');
231 ylabel('$|Y(f)|$ [dB]');
232 legend(\$|Y(f)|\$", \$\pm f 1\$", \$\pm f 2\$", \$\pm f 3\$", 'Location', 'northeast');
233 title('Logrithmic Magnitude of $Y(f)$ of the Filtered Signal');
235 exportgraphics(gcf, 'Figure3-PartIII.png', 'ContentType', 'image');
236
237 % Desample the complex filtered sequence y(n) by a factor of 8. -----
238 factor downsample = 8;
239 x prime = y complex(1:factor downsample:end);
240
241 %% High Resolution Spectral Analysis.
242 % Window the data points (n = 256-511) of the signal x'(n).
243 x prime 256 KB = x prime(NFFT+1:2*NFFT) .* KaiserBesselWindow;
244
245 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
246 X prime 256 KB magnitude = 20*log10(abs(fftshift(fft(x prime 256 KB, NFFT))));
247 X prime 256 KB magnitude = X prime 256 KB magnitude - max(X prime 256 KB magnitude); 🗸
% Normalize the results.
249 % Calculate the upadated bin width of the FFT.
250 fs prime = fs/factor downsample;
251 BinWidth prime = fs prime/NFFT;
252 f prime fft = (-fs prime/2:BinWidth prime:fs prime/2-BinWidth prime)';
253
254 % Plot the results.
255 figure('Position', [0, 0, 800, 600]);
256 hold on;
257 plot(f prime fft, X prime 256 KB magnitude, 'b', 'LineWidth', 2);
258 xline(f(1)-f0+fs prime, 'r--', 'LineWidth', 2);
259 xline(f(2)-f0, 'g--', 'LineWidth', 2);
260 xline(f(3)-f0, 'k--', 'LineWidth', 2);
261 xline(-f(1)-f0+3*fs prime, 'r--', 'LineWidth', 2);
262 xline(-f(2)-f0+4*fs prime, 'g--', 'LineWidth', 2);
263 xline(-f(3)-f0+4*fs prime, 'k--', 'LineWidth', 2);
264 grid on;
265 box on;
266 xlim([-70 70]);
267 xticks(-70:10:70);
268 ylim([-120 0]);
269 yticks(-120:20:0);
```

```
270 xlabel('Analog Frequency $f$ [Hz]');
271 ylabel('$|X^\prime(f)|$ [dB]');
272 legend('\$|X^prime(f)|\$', '\$pm f 1\$', '\$pm f 2\$', '\$pm f 3\$', 'Location', \checkmark
'northeast');
273 title('Logrithmic Magnitude of $X^\prime(f)$');
274
275 exportgraphics(gcf, 'Figure4-PartIV.png', 'ContentType', 'image');
276
277 %% Second Iteration on LPF Design.
278 % Trial #1: Use passband/stopband weight ratio = 10. -----------
279 Weight2 = [10 1]; % Define weight ratio vector [passband stopband].
280 h2 = firpm(N coefficients-1, f parameter, a parameter, Weight2)'; % Filter design.
281
282 % Calculate the frequency response of the filter.
283 h2 padded = padarray(h2, [NFFT filter-size(h2,1) 0], 'post'); % Pad the filter to the \checkmark
same length as NFFT.
284
285 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
286 H2 magnitude = 20*log10(abs(fftshift(fft(h2 padded.*RectangularWindow, 🗸
NFFT filter))));
287 H2_magnitude = H2_magnitude - max(H2 magnitude); % Normalize the result.
288
289 % Plot the result.
290 figure('Position', [0, 0, 1800, 1000]);
291
292 subplot (2, 2, 1);
293 hold on;
294 plot(0:1:N coefficients-1, h2, 'b', 'LineWidth', 2);
295 grid on;
296 box on;
297 xlim([0 70]);
298 xticks(0:10:70);
299 ylim([-0.04 0.16]);
300 yticks(-0.04:0.04:0.16);
301 xlabel('$n$');
302 ylabel('$h(n)$');
303 title('Impulse Response $h(n)$');
304
305 subplot(2,2,2);
306 hold on;
307 plot(f filter fft, H2 magnitude, 'b', 'LineWidth', 2);
308 xline(fc passband, 'r--', 'LineWidth', 2);
309 xline(fc stopband, 'k--', 'LineWidth', 2);
310 grid on;
311 box on;
312 xlim([-500 500]);
313 xticks(-500:100:500);
314 ylim([-100 0]);
315 yticks(-100:20:0);
316 xlabel('Analog Frequency $f$ [Hz]');
317 ylabel('$|H(f)|$ [dB]');
318 legend('$|H(f)|$', 'Passband', 'Stopband', 'Location', 'northeast');
319 title('Logrithmic Magnitude of $H(f)$');
320
321 subplot(2,2,3);
```

```
322 hold on;
323 plot(f filter fft, H2 magnitude, 'b', 'LineWidth', 2);
324 grid on;
325 box on;
326 xlim([0 40]);
327 xticks(0:10:40);
328 ylim([-0.01 0.002]);
329 yticks(-0.01:0.002:0.002);
330 xlabel('Analog Frequency $f$ [Hz]');
331 ylabel('$|H(f)|$ [dB]');
332 title('Passband Ripples of $H(f)$');
334 % Low pass filter the complex sequence in the time domain.
335 y2 complex = filter(h2, 1, x complex);
336
337 % Desample the complex filtered sequence y(n) by a factor of 8.
338 x2 prime = y2 complex(1:factor downsample:end);
339
340 % Window the data points (n = 256-511) of the signal x'(n).
341 x2 prime 256 KB = x2 prime(NFFT+1:2*NFFT) .* KaiserBesselWindow;
342
343 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
344 X2 prime 256 KB magnitude = 20*log10(abs(fftshift(fft(x2 prime 256 KB, NFFT))));
345 X2 prime 256 KB magnitude = X2 prime 256 KB magnitude - max ✔
(X2 prime 256 KB magnitude); % Normalize the results.
346
347 % Plot the results.
348 subplot (2, 2, 4);
349 hold on;
350 plot(f prime fft, X2 prime 256 KB magnitude, 'b', 'LineWidth', 2);
351 xline(f(1)-f0+fs prime, 'r--', 'LineWidth', 2);
352 xline(f(2)-f0, 'g--', 'LineWidth', 2);
353 x \text{line}(f(3)-f0, 'k--', 'LineWidth', 2);
354 xline(-f(1)-f0+3*fs prime, 'r--', 'LineWidth', 2);
355 xline(-f(2)-f0+4*fs prime, 'g--', 'LineWidth', 2);
356 xline(-f(3)-f0+4*fs prime, 'k--', 'LineWidth', 2);
357 grid on;
358 box on;
359 \times \lim([-70 \ 70]);
360 xticks(-70:10:70);
361 ylim([-120 0]);
362 yticks(-120:20:0);
363 xlabel('Analog Frequency $f$ [Hz]');
364 ylabel('$|X^\prime(f)|$ [dB]');
365 legend('$|X^\prime(f)|$', '$\pm f 1$', '$\pm f 2$', '$\pm f 3$', 'Location', ✓
'northeast');
366 title('Logrithmic Magnitude of $X^\prime(f)$');
367
368 exportgraphics(gcf, 'Figure5-PartIV-Trial#1.png', 'ContentType', 'image');
369
370 % Trial #2: Use passband/stopband weight ratio = 100. -----
371 Weight3 = [100 1]; % Define weight ratio vector [passband stopband].
372 h3 = firpm(N coefficients-1, f parameter, a parameter, Weight3)'; % Filter design.
374 % Calculate the frequency response of the filter.
```

```
375 h3 padded = padarray(h3, [NFFT filter-size(h3,1) 0], 'post'); % Pad the filter to the \checkmark
same length as NFFT.
376
377 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
magnitude.
378 H3 magnitude = 20*log10(abs(fftshift(fft(h3 padded.*RectangularWindow, ✓
NFFT filter))));
379 H3 magnitude = H3 magnitude - max(H3 magnitude); % Normalize the result.
381 % Plot the result.
382 figure('Position', [0, 0, 1800, 1000]);
384 subplot (2, 2, 1);
385 hold on;
386 plot(0:1:N coefficients-1, h3, 'b', 'LineWidth', 2);
387 grid on;
388 box on;
389 \times lim([0 70]);
390 xticks(0:10:70);
391 ylim([-0.04 0.16]);
392 yticks(-0.04:0.04:0.16);
393 xlabel('$n$');
394 ylabel('$h(n)$');
395 title('Impulse Response $h(n)$');
397 subplot (2, 2, 2);
398 hold on;
399 plot(f_filter_fft, H3 magnitude, 'b', 'LineWidth', 2);
400 xline(fc passband, 'r--', 'LineWidth', 2);
401 xline(fc_stopband, 'k--', 'LineWidth', 2);
402 grid on;
403 box on;
404 xlim([-500 500]);
405 xticks(-500:100:500);
406 ylim([-100 0]);
407 yticks(-100:20:0);
408 xlabel('Analog Frequency $f$ [Hz]');
409 ylabel('$|H(f)|$ [dB]');
410 legend('$|H(f)|$', 'Passband', 'Stopband', 'Location', 'northeast');
411 title('Logrithmic Magnitude of $H(f)$');
412
413 subplot (2,2,3);
414 hold on;
415 plot(f filter fft, H3 magnitude, 'b', 'LineWidth', 2);
416 grid on;
417 box on;
418 xlim([0 40]);
419 xticks(0:10:40);
420 ylim([-0.01 0.002]);
421 yticks (-0.01:0.002:0.002);
422 xlabel('Analog Frequency $f$ [Hz]');
423 ylabel('$|H(f)|$ [dB]');
424 title('Passband Ripples of $H(f)$');
425
426 % Low pass filter the complex sequence in the time domain.
427 y3 complex = filter(h3, 1, x complex);
```

```
428
429 % Desample the complex filtered sequence y(n) by a factor of 8.
430 x3 prime = y3 complex(1:factor downsample:end);
432 % Window the data points (n = 256-511) of the signal x'(n).
433 x3 prime 256 KB = x3 prime(NFFT+1:2*NFFT) .* KaiserBesselWindow;
435 % Perform 256-point FFT on the windowed signal and calculate the logarithmic \checkmark
436 X3 prime 256 KB magnitude = 20*log10(abs(fftshift(fft(x3 prime 256 KB, NFFT))));
437 X3 prime 256 KB magnitude = X3 prime 256 KB magnitude - max ✔
(X3 prime 256 KB magnitude); % Normalize the results.
438
439 % Plot the results.
440 subplot(2,2,4);
441 hold on;
442 plot(f prime fft, X3 prime 256 KB magnitude, 'b', 'LineWidth', 2);
443 xline(f(1)-f0+fs prime, 'r--', 'LineWidth', 2);
444 xline(f(2)-f0, 'g--', 'LineWidth', 2);
445 xline(f(3)-f0, 'k--', 'LineWidth', 2);
446 xline(-f(1)-f0+3*fs prime, 'r--', 'LineWidth', 2);
447 xline(-f(2)-f0+4*fs prime, 'q--', 'LineWidth', 2);
448 xline(-f(3)-f0+4*fs prime, 'k--', 'LineWidth', 2);
449 grid on;
450 box on;
451 \times lim([-70 70]);
452 xticks(-70:10:70);
453 ylim([-120 0]);
454 yticks (-120:20:0);
455 xlabel('Analog Frequency $f$ [Hz]');
456 ylabel('$|X^\prime(f)|$ [dB]');
457 legend('\$|X^{\text{prime}}(f)|\$', '\$|pm f 1\$', '\$|pm f 2\$', '\$|pm f 3\$', 'Location', \checkmark
'northeast');
458 title('Logrithmic Magnitude of $X^\prime(f)$');
460 exportgraphics(gcf, 'Figure6-PartIV-Trial#2.png', 'ContentType', 'image');
461
```