Design Project – Digital Filters and Their Applications

Introduction to Matlab

A widely used linear-phase FIR filter design is based on the Parks-McClellan algorithm, which results in an optimal FIR filter with an equiripple weighted error. It makes use of the Parks-McClellan optimal equiriple optimization algorithm and is available in Matlab as the function "firpm". This function can be used to design any type of single band or multiband filter. In its basic form, the command *b*=firpm(*N*,*F*,*A*,*W*) returns a vector ***b*** of length *N*+1 containing the impulse response coefficients of the desired FIR filter in ascending powers of z^{-1}. *F* is the vector of specified frequency points, arranged in increasing order, in the range 0 to 1 with the first frequency point being 0 and the last frequency point being 1. The sampling frequency is assumed to be 2 Hz. The desired magnitudes of the FIR filter frequency response at the specified band edges are given by the vector *A*, with the elements given in equal-valued pairs. The desired magnitude responses in the passband(s) and the stopband(s) can be weighted by the vector *W*. The order *N* of the FIR filter to meet the given specifications can be estimated using "firpmord" command. It can be used in the following form: [*N*, *F*, *A*, *W*]=firpmord(*F*,*A*,*Dev*,*Fs*). To plot the frequency response in dB and phase response, Matlab command "freqz" can be used.

Example:

a lowpass filter with passband edge at 1500Hz, stopband edge at 2000Hz, the passband and stopband ripples are 0.01 for both bands, the sampling frequency is 8000Hz. We can use firpmord to estimate the filter length.

```
>>F=[1500, 2000];
>>A=[1,0];
>>Dev=[0.01,0.01];
>>fs=8000;
>>[N,Fi,Ai,W]=firpmord(F,A,Dev,fs);
```

To find the filter coefficients, use firpm command:

```
>>h=firpm(N,Fi,Ai,W);
```

To plot frequency response : use freqz;

```
>>freqz(h,1);
```

To find more information on these two commands, type "help firpmord" or "help firpm" in Matlab.

For the design of IFIR filter, you can use "ifir" command in Matlab. Type "help ifir" in Matlab for details.

Design Project (Report due: Dec. 1 at 12 midnight)

Submission details:

You need to submit a project report and all m files used in your experiment. The project report should consist of following sections.

1. Introduction: a brief statement (few sentences) mentioning what you did in the project and your results.
2. Equipment: please list equipment and software used in the project.
3. Results and discussion: this is the main part of your report. Please include your codes, and plots. Note: the code must be well documented. If you were asked to compare experimental results to the calculations, or to explain parts of your experiment, it should be presented in this section.
4. Conclusion: state what you learn from this project, objectives you achieved, and any difficulties you met.

The report together with all m files should submit to Moodle: https://learn.lassonde.yorku.ca.

You will need to provide your login credentials to access Moodle. Once you have logged in, select the course: EECS 3602 E EN LECT 1 Systems and Random Processes. The lab report submission is under "Lab" tag.

Files to be submitted:

1. Your report.
2. All M-file used in the project.
3. Please submit one zipped file or published file, which contains your report and all m files.

Grading details:

75% of your lab grade is for completing all project requirements correctly. 15% is for clear writing and good presentation, and well-documented code; 10% is for extra work or analysis or going beyond the project requirements.

The project contains two parts. In part 1, you design filters using Matlab commands and plot the frequency response. In part 2, you use the filter(s) to remove noises in the given ECG signal.

Part 1: FIR Filter Design

Q1. Design a narrowband FIR filter using the interpolated finite impulse response (IFIR) technique. The filter should meet the following specifications:
    Passband edge: 3600 Hz

Stopband edge: 3200 Hz
Sampling frequency: 8000 Hz
Pass-band ripple <= 0.01
Stopband ripple <= 0.0001

Design the filter using Matlab command: "ifir"
(a) Find an optimal interpolation factor, M.
(b) Find bandedges of bandedge shaping filter $H_a(z)$ and masking filter $H_{Ma}(z)$.
(c) Calculate the filter lengths of bandedge shaping filter and masking filter.
(d) Design the filter using "ifir". What are the final filter lengths for bandedge shaping filter and masking filter? Are they the same as calculated values in (c)?
(e) Plot the frequency response of interpolated bandedge shaping filter, masking filter, and overall filter.

Q2. Design an FIR filter using the Frequency-Response Masking technique. The filter should meet the following specifications:
Normalized passband edge: 0.194
Normalized stopband edge: 0.2
Passband ripple: 0.01
Stopband ripple: 0.001

Design the filter according to the procedures given in Chapter 8.
(a) Calculate the optimum interpolation factor, M_{opt}, using formula given in Chapter 8.
(b) Find the parameters listed in the following Table.

M	Case A or B	Bandedges of $H_a(z)$	Bandedges of $H_{ma}(z)$	Bandedges of $H_{mc}(z)$	L_a	L_{Ma}	L_{Mc}	L_{Total}
M_{opt} -2								
M_{opt} -1								
M_{opt}								
M_{opt} +1								
M_{opt} +2								

where L_a, L_{Ma}, L_{Mc}, L_{Total} are the filter lengths of bandedge shaping filter, upper branch masking filter, lower branch masking filter, and the sum of L_a, L_{Ma} and L_{Mc}, respectively.
(c) Determine the optimum interpolation factor based on the Table.
(d) Design the bandedge shaping filter, upper branch masking filter, lower branch masking filter use "firpm" with the bandedges found in the Table and taking 85% of the given ripples for each filter.
(e) Write a Matlab program to find the frequency response in linear scale. (Hint: you can develop the program based on the frequency response formula for Type I and Type II filters given in Chapter 7.).
(f) Use the program in (e) to find the frequency responses of all 3 filters and combine them to form the overall filter.

(g) Zoom in to passband and stopband regions to verify whether the frequency response of the overall filter meets the given specifications. If not, re-design the subfilters until the overall filter meets the given specifications.
(Note: You need to plot the 3 frequency responses for the overall filter in linear scale: (1) overall filter; (2) passband details; (3) stopband details.)

Part II Remove noises in ECG signal.

1. Brief Introduction of ECG Signal

Electrocardiography (ECG) is a technique to capture the electrical activities of a heart against time. The heart is an organ that pumps blood throughout the body in a rhythmic fashion called a heartbeat. Electrical impulses captured in an ECG test provide rich information about the biological activities of the heart, which helps medical professionals to identify heart problems.

Early studies in the 1950s and 1970s have shown that the ECG signal is generated from the depolarization and repolarization processes in the heart cell generating a propagation of dipole wavefront across the tissue of the heart. Figure 1 illustrates a single cycle regular Lead II ECG signal drawn on a standard ECG paper. The major deflections are labeled in alphabetical order, namely, P wave, QRS complex, T wave and U wave.

The P wave is relative to the atrial depolarization. Following that, the QRS complex corresponds to the ventricular depolarization. Note that the atria repolarization also occurs during the QRS complex but its amplitude is insignificant in the ECG waveform. The T wave refers to the repolarization of the ventricles.

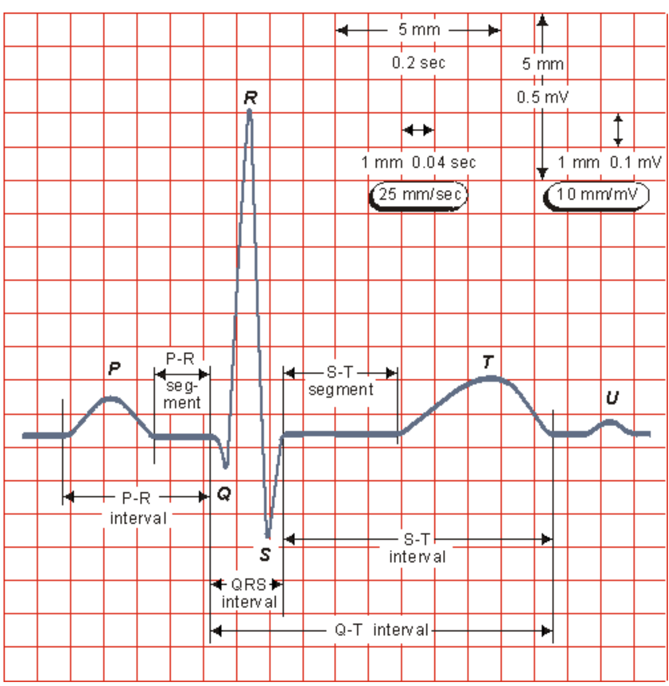

Figure 1: The regular ECG signal in one cardiac cycle

These waves and segments are measured and commonly used by medical professionals for diagnosis and evaluation. A more detailed mapping of heart activity leading to the overall ECG cardiac cycle is shown in Figure 2. Heart rate is measured by the time interval between two successive R peaks.

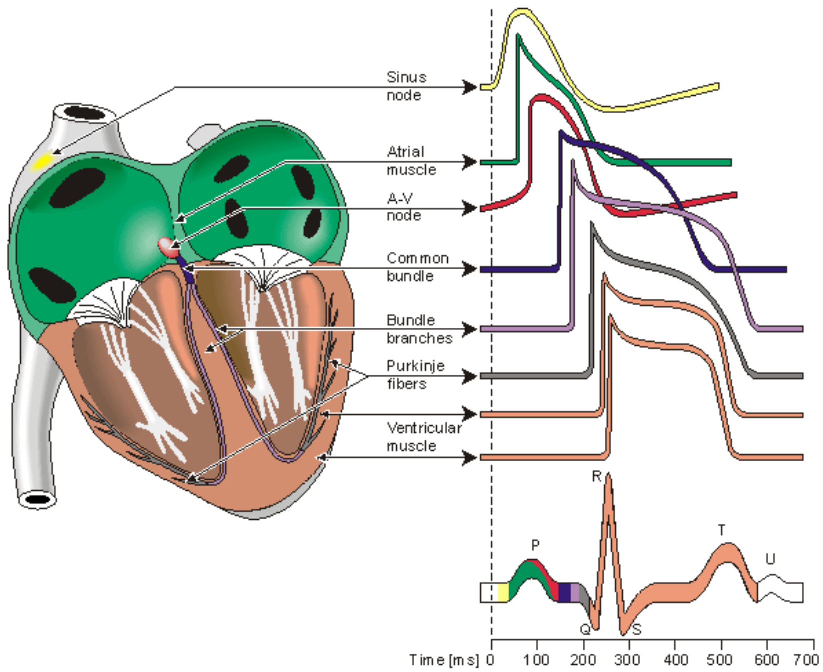

Figure 2: The firing of signals from various heart location forming the ECG one cardiac cycle

ECG signal is often corrupted with various forms of noises and artifacts. Hence, it is often difficult to automatically identify important signal features. Pre-processing steps might help improve the signal to noise ratio (SNR) of the ECG signal, so as to make feature detection techniques more effective and accurate. The most commonly found noise types observed in the ECG signal electrode contact noise, motion artifacts, respiration noise, and powerline noise. The respiration noise induces a drift in the baseline of the measured ECG signal, which is often a sinusoidal component at the frequency of respiration, i.e. 0.15 to 0.3Hz. The powerline noise is at frequency 50/60Hz.

The ECG data given in the project (download from course website) is corrupted by respiration and powerline noises. Your task is to design two FIR filters to remove these noises, i.e. one filter to remove respiration noise, and another filter to remove powerline noise. This can be done by using following filters:

Highpass filter:
Stopband edge: 0.17Hz
Passband edge: 1.0Hz

Lowpass filter:
Passband edge: 40Hz

Stopband edge: 54Hz

The passband and stopband ripples are: 0.01 and 0.001, respectively, for both filters.
The sampling frequency is 360Hz.

Apply these two filters to the ECG signal. Plot the original noisy ECG signal and filtered ECG signal.