

Digital Signal Processing Laboratory



Introduction

As you now understand, electrical signals originating from physiological processes of the body can be measured using electronic equipment. Each of these biopotentials has different amplitude and frequency characteristics that make them distinct from other measurements. For example, the ECG signal appears very different from the EEG, EOG, or EMG signals. However, sometimes recordings are done where one physiological signal may contaminate the other due to the proximity of the recording electrodes.



In order to extract the desired signal from a contaminated signal, processing must be performed. Digital signal processing (DSP) can be used to extract meaningful information from recorded physiological signals. Digital signal processing methods are simply digital models of analog signal processing methods. As compared to analog signal processing, digital signal processing provides a person with the flexibility to easily change the parameters of the filter. If this type of processing were done with analog electronics, one would have to rebuild or replace parts in a circuit each time a parameter change was desired. Digital signal processing allows the computer to change the filter parameters and provide instant results!

This laboratory will cover some techniques and background on how signal processing can be applied to separate out physiological signals that might be interfering with each other. This will require understanding of Fourier analysis, some basic filters such as highpass and lowpass, and how these basic analog filters can be converted into their digital equivalents.

Equipment required:

- CleveLabs Laboratory Kit
- CleveLabs Course Software
- MATLAB® or LabVIEW™

Background

Frequency Domain Analysis

In the first laboratory session, we analyzed signals in the time domain. However, only utilizing time domain processing is somewhat limiting if we wish to process these signals to eliminate artifacts. The more practical method is to do frequency domain analysis. There is a bi-directional relationship between the time domain and frequency domain. Suppose a sound generator is used to generate a 60 Hz tone. If you looked at the electrical signal output of the tone generator on an oscilloscope, you would see a continuous periodic signal that has a frequency of 60 Hz. This means that only a single frequency is in that signal, 60 Hz. Plotting the same signal in the frequency domain would reveal a peak at 60 Hz, and zero elsewhere (Fig 1). This transformation is reversible. Given a plot of the frequency of a signal, it can be converted into a time-domain signal.

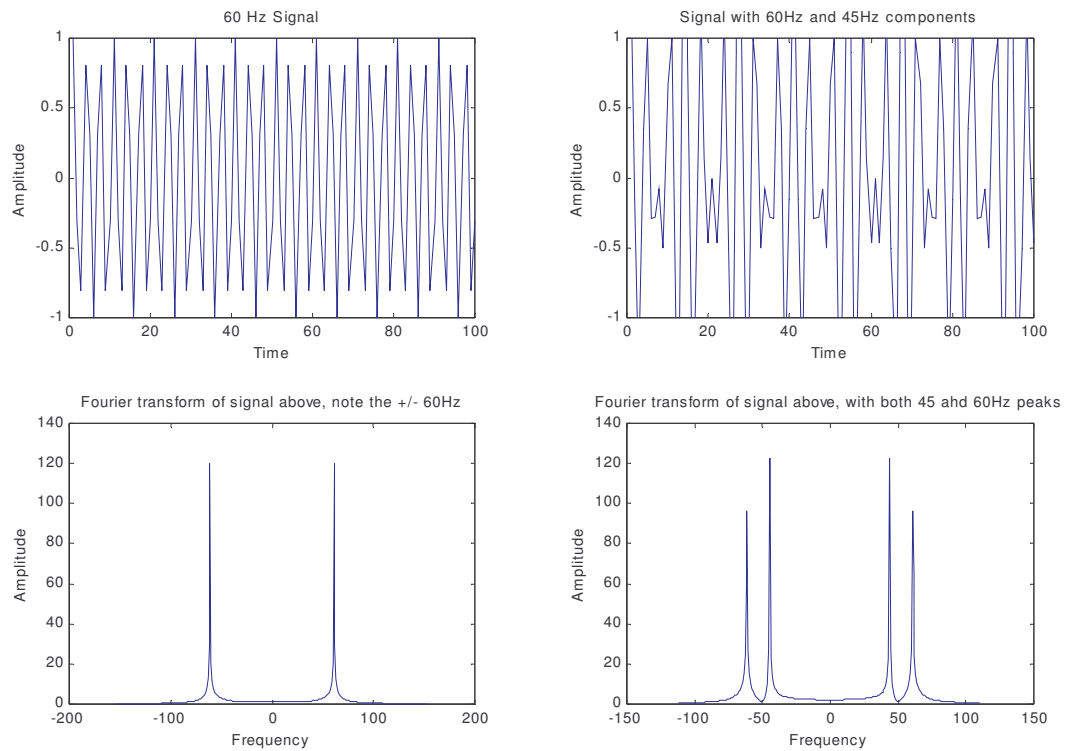


Figure 1: Example of Fourier transforms of two signals. Notice that there are peaks at both positive and negative frequencies. This negative frequency is the same as the positive frequency due to a trigonometric relationship. So, there really are not two frequencies, they are one and the same.

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The basis behind these frequency transformations is the Fourier transform. The Fourier transform is an extremely powerful mathematical tool that allows scientists and engineers to analyze the frequency components of signals. Many of the signal processing techniques that are covered in this laboratory will employ frequency-based tools. The mathematical basis of the Fourier transform is the following:

$$F(\omega) = \int_{-\infty}^{\infty} f(t)e^{-j\omega t} dt$$

$F(\omega)$ denotes the frequency domain signal, where ω is the frequency in radians. Now, to convert back from the frequency domain to the time domain, the inverse Fourier transform is used.

$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} F(\omega)e^{j\omega t} d\omega$$

When signals are represented in the frequency domain, the function is typically capitalized, thus the $F(\omega)$ and $f(t)$.

The Fourier transform only exists for signals that satisfy the following three properties:

1. Absolutely integrable
2. Has a finite number of discontinuities
3. Finite number of minimum and maximum points.

Simply stated, the above three properties mean that the signal must be bounded and contain a finite number of discontinuities.

Frequency analysis is particularly useful in the design of filters. Filters are used to emphasize frequency bands that are important and de-emphasize frequency bands that are not a part of the desired signal. As you have learned, 60 Hz noise is evident in almost all physiological recordings due to the interference from surrounding electrical systems. Without frequency domain analysis, there are no tools that can be used to eliminate this noise. However, if a filter can be designed to block out this 60 Hz noise quality of the physiological recording will be improved. Since this noise may obscure the signal being recorded, a 60 Hz notch filter is used to eliminate this artifact. A notch filter is an extremely narrow band filter that does not allow a band of signals to pass through. For example, a 60 Hz notch filter is frequently used to block out 60 Hz noise.

Filters

As mentioned earlier, DSP techniques implemented on the computer are actually based on mathematical models of analog hardware filters. There are four main types of filters of which you should be aware (Fig 2). These include lowpass filters, which allow low

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frequencies to pass through, highpass filters, which allow high frequencies to pass through, bandpass filters, which allow signals within a certain frequency band to pass through by combining a high and lowpass filter, and bandstop filters, which prevent certain frequency bands from passing through.

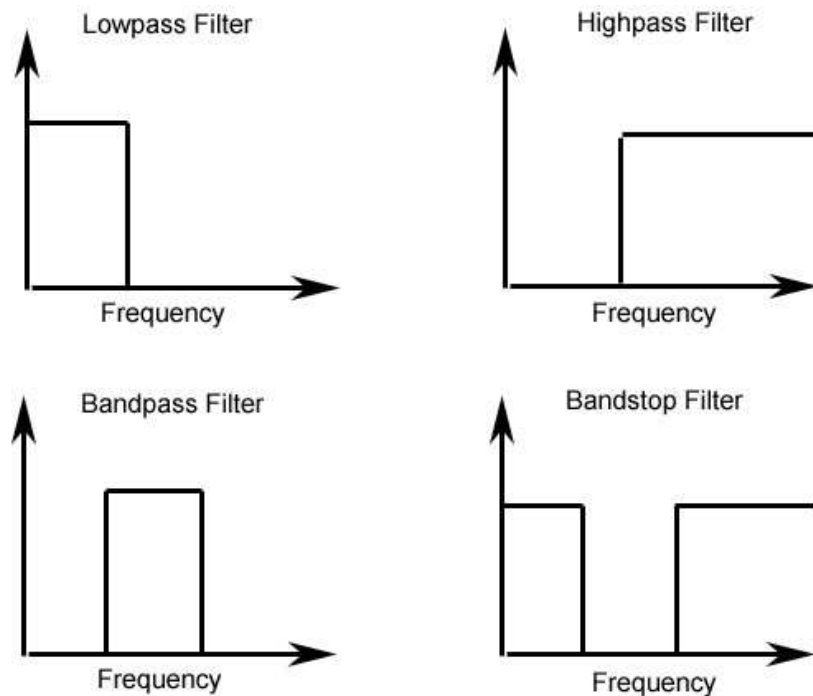


Figure 2: Diagram of four basic filter types. Ideal filters would have the steep cutoffs as seen in the diagram, however, real-world filters do not have perfect frequency cutoffs.

Understanding how to analyze hardware filters requires knowledge of the concept of complex impedance. As you know, resistors have certain impedance. Ideally, the impedance of a resistor is independent of the frequency of the signal passing through it. Capacitors and inductors also have impedances associated with them. The only difference here is that capacitors and inductors have complex impedances. In other words, their impedance is dependant on the frequency of the signal passing through them. Understanding complex impedance is a power tool for characterizing hardware filter circuits.

Recall the fundamental equations for a capacitor and inductor:

$$i = C \frac{dV}{dt}; \text{ The current through a capacitor}$$

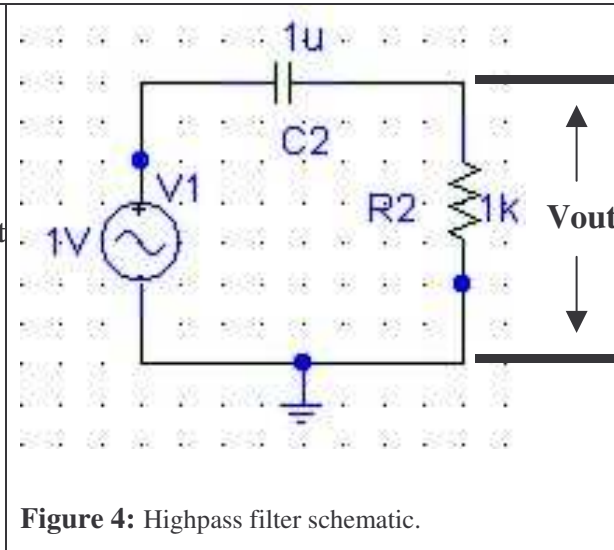
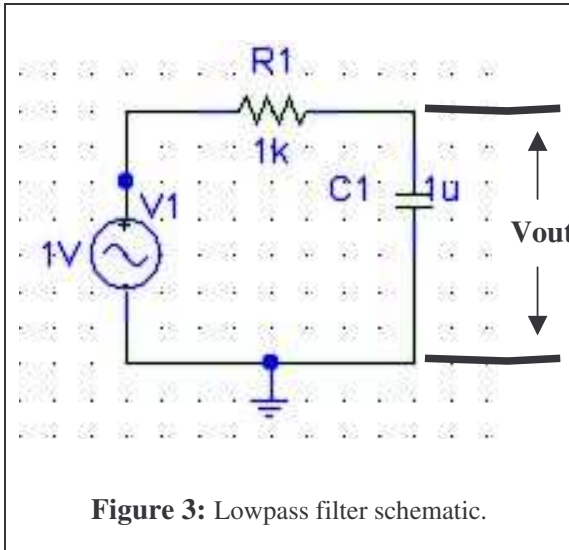
$$V = L \frac{dI}{dt}; \text{ The voltage across an inductor}$$

Where C and L are the capacitance or inductance values.

Now, suppose an operator s exists, which acts like $\frac{d}{dt}$. Substituting s in the equations above yields: $i = CsV$, $V = LsI$. Ohm's Law states that $R = V/I$. The complex impedance Z_C of a capacitor is then $Z_c = \frac{1}{Cs}$, and the complex impedance Z_L of an inductor is $Z_L = Ls$.

You may be wondering what s means. One can substitute $j\omega$ for s , and now you have a relationship between frequency and impedance. So, when replacing $j\omega$ in the equations above, as frequency ω increases to infinity, the complex impedance of a capacitor goes towards zero, and the complex impedance of an inductor goes towards infinity. And as an additional note, Z is used to denote complex impedance instead of R , to avoid confusion.

So now, we can treat inductors and capacitors as resistors with complex impedance. The first circuits to analyze with this approach are the first order lowpass and highpass filters (Figure 3 and 4). These circuits contain both an R and a C element. The lowpass filter is measured across the capacitor while the highpass filter is measured across the resistor.



First we will analyze the lowpass filter (Figure 3). When presented with this circuit, we first solve for the transfer function of this circuit. The transfer function, V_{out}/V_{in} , of this circuit is $\frac{Z_C}{Z_C + R1}$. Substituting values, this becomes $\frac{1}{RCj\omega + 1}$. It is obvious that for low frequency values, this transfer function approaches one, and for high frequency

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values, this transfer function becomes zero, thus making it a lowpass filter, since low frequency values will pass, but high frequencies are attenuated. Solving for the transfer function of the highpass filter is left as an exercise for the student.

Now, how do we determine when the filter begins to attenuate high frequency signals? This requires the computation of the cutoff frequency. Looking at the transfer function, this is when the value of ω is equal to $\frac{1}{RC}$. This is because when ω is equal to RC in the transfer function, the value of V_{out}/V_{in} becomes $\frac{1}{\sqrt{2}}$.

Ideally, we would want filters to pass or block signal components at the exact specified cutoff frequencies. However, we do not live in an ideal world, nor do we have access to ideal filter components. Realizable filters attenuate over a range of frequencies rather than dropping off to 0 at a specific frequency.

To characterize how fast the filter is able to attenuate signals at the cutoff frequency, we perform a measurement called a Bode plot. The Bode plot illustrates frequency on the x-axis and the attenuation of the signal on the y-axis. On the y-axis, 1 refers to no attenuation while 0 refers to no amplitude at that frequency. Bode plots are extremely useful in visualizing the frequency response of a filter. Experimentally, this can be performed by measuring the output of a circuit at different frequencies, then plotting these values on a semilog scale.

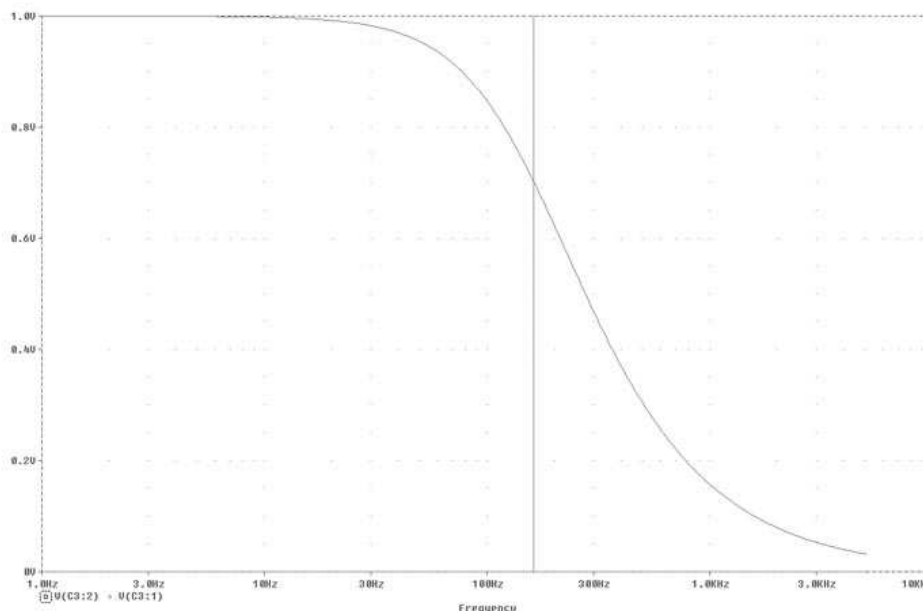


Figure 5: Lowpass Bode Plot for the RC circuit above. The cursor shows the 3dB point to be at 161 Hz, even though the cutoff frequency was designed for 100 Hz. See that low frequencies are passed up to about 100 Hz, and then attenuated for frequencies above that.

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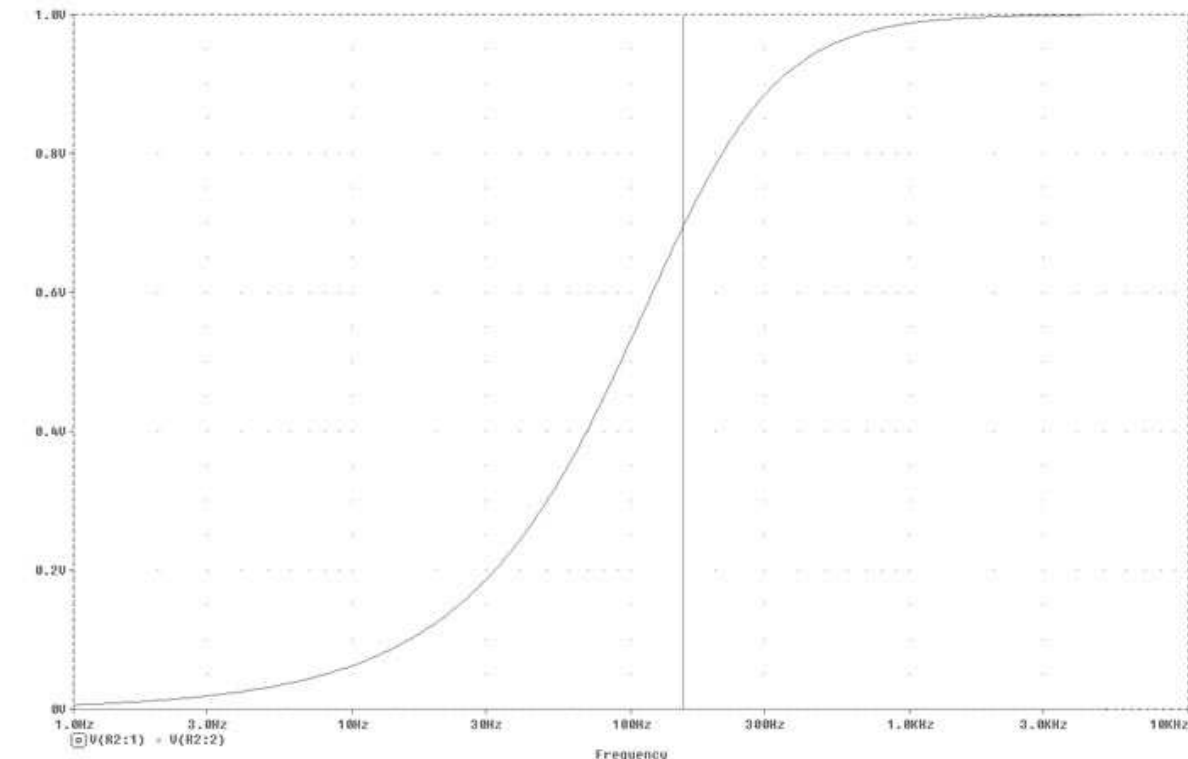


Figure 6: High pass Bode plot for a first order RC circuit. The cutoff frequency here is 100 Hz, but the 3dB point is at approximately 160 Hz.

Decibels, or dB, relate the output power of a circuit to the input power. Decibels are a logarithmic scale, so that high values of gain or attenuation don't require very large numbers. Computing between voltage and decibels is done through the following equation:

$$dB = 10 \log \left(\frac{V_{out}^2 / R}{V_{in}^2 / R} \right).$$

Some may observe that V^2/R is a measurement of power. Now,

performing some math, the R term drops out since it is present in both the numerator and denominator. Furthermore, using the exponent property of logarithms, this equation

becomes $dB = 20 \log \left(\frac{V_{out}}{V_{in}} \right)$. So, using this equation, when the amplitude of a signal is

$$\frac{1}{\sqrt{2}} V_0, \text{ the dB value is } -3\text{dB}.$$

At this 3dB attenuation point, the power is reduced by 1/2, and is sometimes referred to as the half-power point.

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These filters that have been mentioned up to this point are considered first-order filters. They are relatively simple to construct and analyze. However, they are not always the best filter for the application. First order filters have a slow cutoff, and there may be some applications where a filter with a steeper frequency cutoff is desired. In those instances, higher order filters requiring several R, C and L components can be designed, and are more complex to design and analyze. Higher order filters will have steeper roll-offs. This means that the frequency associated with the 3 dB point will be much closer to the cutoff frequency used to design the circuit. These higher order filters do a better job of approximating the ideal filter, since they will only emphasize a tighter frequency band with a steeper roll-off.

Table 1: List of commonly used higher order filters.

Name	Advantage	Disadvantage
Butterworth	Maximally flat in the passband (passband ripple is zero)	Roll-off is not very steep compared to other filters
Chebyshev	Monotonic roll-off, steeper than the Butterworth	Some passband ripple.
Elliptical	Steeper roll-off than Chebyshev filter.	Non-monotonic roll-off, ripple in both the passband and stopband.
Bessel	Designed for linear phase	Slow roll-off compared to other filters above.

These filters also have phase effects as well. Recall from circuits that phase relates to how a certain signal is delayed. For example, the sine wave is an example of a cosine wave that has a phase delay of 90 degrees or $\pi/2$. The phase of a filter is determined by looking at the denominator of the transfer function. Now, note that there are two terms in the denominator, a real term and an imaginary term ($j\omega$). The analysis of the phase angle is done in the imaginary plane, where the x-axis represents real values, and the y-axis represents the imaginary values. Recall the generic form of the lowpass filter transfer function was $\frac{K}{\alpha + j\omega}$.

The phase $\theta(\omega)$ is determined by taking the angle of $K - \arctan(\omega/\alpha)$. Since K is always real, the angle of K is 0 when K is positive, or ± 180 when K is negative. So, the phase for the low pass filter illustrated above is as follows. When ω is small, the phase is 90 degrees. When ω is equal to the filter cutoff, then the phase is 45 degrees, and for large ω , the phase becomes 0 degrees.

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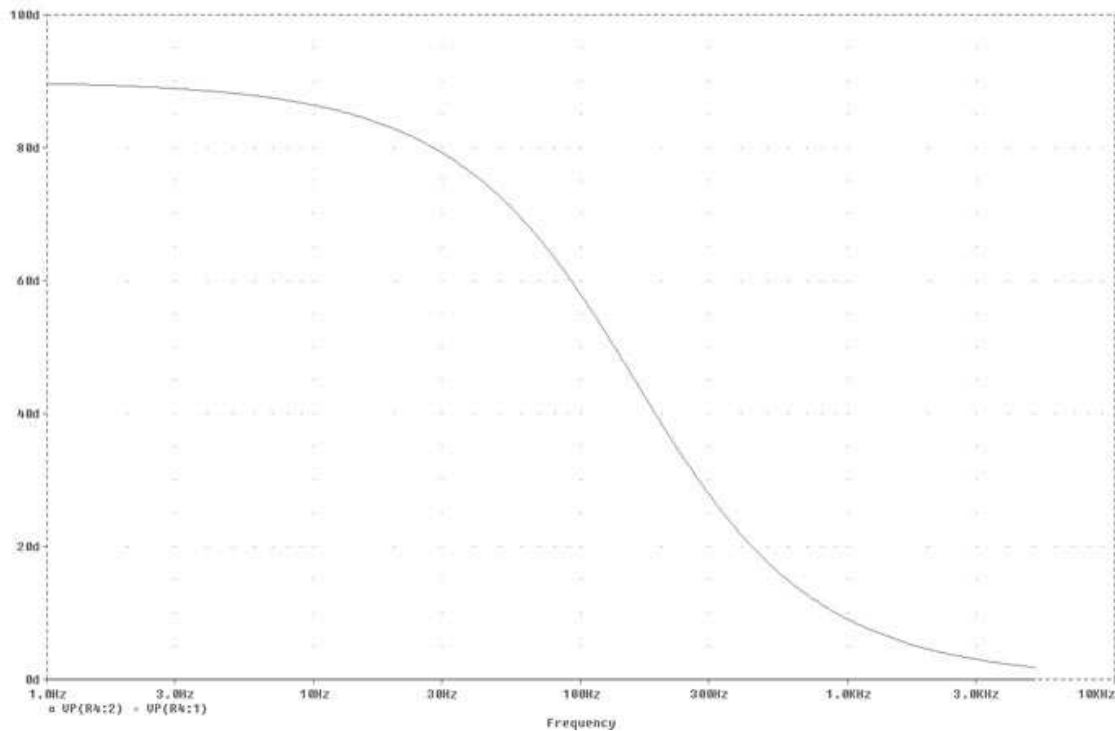


Figure 7: Phase Bode plot of the first order low pass filter.

Digital Filtering

Digital signal processing is a very powerful tool for signal analysis. The analog filters described above are actual hardware circuits. Therefore, if changes in the cutoff frequency or filter order are desired, parts must be replaced. Digital signal processing allows filters to be based on the mathematical rules of these filters. This allows the implementation of these filters in software, thus giving instant flexibility in filter design. In the BioRadio software, there are some filter settings that are based on a higher order digital filter known as a Chebychev filter. In order to design these filters for the computer, we create an array with filter coefficients to mimic the frequency response of the hardware filter. These filter coefficients are then convolved with the digitally sampled signal. The result is an array containing values of the filtered signal.

There are two main types of digital filters: FIR and IIR filters. FIRs, or finite impulse response filters, are designed without using feedback from the output, i.e., the output of the filter has no impact on the next sample that is filtered. These FIR filters have linear phase and variable steepness, depending on the filter order. IIRs, or infinite impulse response filters, use feedback so that the filtered output has an effect on the next value. IIR filters do not have linear phase, but the advantage to using them is that fewer filter coefficients are used for an equivalent performing FIR filter. IIR filters are commonly used to approximate higher order analog filters such as the Chebychev, Bessel, or Butterworth filters.

Biopotentials Interference

Digital filtering tools can be used to remove the undesired biopotential from the desired one. Other signals contained within the signal one is attempting to measure are known as artifact. Recall that the electroencephalogram (EEG) is a measurement of the activity of the brain. More specifically, the signal that is measured on the scalp originates from the post-synaptic potentials of the neurons in the brain. When these neurons fire synchronously, the EEG appears as a signal with a certain frequency. Brain waves can be used to determine when a person is sleeping, awake, or having a seizure. The electro-oculogram (EOG) is a measurement of the electric field generated by the eye. The EOG is used in sleep studies to help characterize when a person is in REM sleep. The EOG can also be used to determine the direction of a person’s gaze. Finally, the electromyogram (EMG) measures the number of muscle fibers depolarizing and can be used as an indicator of muscle force. EMG can often be higher amplitude than the other biopotentials in the body.

Signal	Typical Frequencies (Hz)	Typical Amplitude (uV)
EEG	8 - 13 (α)	20-100
	13 - 22 (β)	5-10
	0.5 - 4 (Δ)	20-100
	4 - 8 (θ)	10
EOG	DC-100	10 – 5000
EMG	2-500	50 – 5000

Table 2: Typical amplitudes and frequencies for the signals we will be measuring in this laboratory are shown above.

In another example, muscles in the face and neck are used for chewing, talking, and maintaining the posture of the head. EMG often times creates large artifact in the EOG and EEG signals. For example, a patient may be in an epilepsy monitoring unit with scalp EEG electrodes placed on the head. If a seizure starts occurring while they are eating lunch, there is going to be a large amount of EMG artifact polluting the EEG signal. This is obviously undesired, so the engineer must understand what frequencies the EMG contains, and filter those frequencies out while minimally affecting the EEG data.

Experimental Methods

Experimental Setup

1. Using the BioRadio Configuration Wizard, program your BioRadio transmitter and receiver to the existing configuration file “LabDSPBasics”.
2. After the unit has been programmed successfully, connect the Test Pack to the transmitter.
3. Run the Course software. From the “Engineering Basics” lab set, select the “Digital Signal Processing” laboratory session and click on the “Begin Lab” button.

Procedure and Data Acquisition

1. Make sure the receiver is properly connected to the serial port on the computer and is powered on. Make sure your transmitter is still connected to the test pack. Turn the transmitter ON.
2. Click on the green “Start” button.
3. Click on the “Test Pack Data” tab. You should see the BioRadio Raw Test Pack Data plot. Make sure that the time scale is set to 1 second.
4. You should see the +/-150uV, 10Hz square wave begin scrolling across the screen. Report this screen to a new report file and call the report “LabDSP”.
5. Save a few seconds of data to file and name the saved data file “DSPtestpack”
6. Next, click on the tab labeled “Spectral Analysis”.

Spectral Analysis

Each laboratory after this one will have a similar spectral analysis screen. Therefore, this section will be explained in greater detail here than it will be in consecutive labs. The Spectral Analysis tab allows you to perform digital filtering on the BioRadio signal and then view that signal in either the time or frequency domain. Clicking on the subtabs in the main spectral analysis tab makes the selection of the “Frequency Domain” or “Time Domain” plots. The spectral analysis is completed one channel at a time. The “Channel to Process” can be selected in the top right corner of the screen.

The spectral analysis tab allows you to specify the filtering parameters. You can plot raw or filtered data by turning the switch one way or the other. If you have selected “Filtered Data” then the other filter parameters have an effect. The filter that is used in all spectral analysis in this laboratory course is a Butterworth Filter. You can define the type of filter, the highpass cutoff, lowpass cutoff, and order of the filter.

Finally, you can also set spectral parameters of the signal that apply to the frequency domain plot. You can specify a log or linear plot, any type of windowing to be completed, and also the display unit you wish to use. Please note that the spectral analysis is completed over each data collection interval period.

1. Make sure that your data collection interval is set to 100ms. Then click on the Frequency Domain subtab and make sure the filter parameters are set to “Raw Data”. Notice where the peaks occur in the frequency domain. Report this screen.
2. Change the data collection interval to 500ms and then repeat step 1.
3. Click on the Time Domain subtab.
4. Turn on the “Filtered Data” switch, set the filter type to lowpass, and select a lowpass cutoff of 10Hz.
5. Report this plot to your report file.
6. Change the filter to a highpass filter with a cutoff of 10Hz and report this plot
7. Click on the main tab labeled “Processing and Application”.

Processing and Application

1. Click on the tab labeled “Processing and Application”.
2. Under the Display Graph Controls, click on the “Display Raw Signal” switch. Also, make sure that the time scale of the Processed Data Display is set to 1 second. Your data collection interval should still be set to 500ms. Report this plot.
3. Under processing applications, set the noise type to white and the amplitude to 100uV. Report this plot.
4. Change the noise type to sine, set the frequency to 60Hz, and the amplitude to 45uV. This will simulate what 60Hz noise will do to a signal. Notice how a new peak occurs in the frequency domain at 60Hz. Report this plot.
5. Save about ten seconds of this data to file under the file name “60HzNoise” for analysis later. When data is saved to file in this laboratory, the data saved includes “Channel 1 Raw”, “Channel 2 Raw”, and “Processed Data”. We will analyze the processed data channel later.
6. Now turn on the lowpass filter and set it to 20Hz. Examine what happens to both the time and frequency domain plots. Report this plot.
7. Turn the filtering off. Now set the mathematics function to “Derivative” to examine the derivative of the signal. Examine what happens to the time and frequency domains. Notice how the 60Hz noise becomes amplified. Report this plot.
8. Now set the mathematics function to “Integral” and examine the time and frequency domains. Notice how the 60Hz noise has become smoothed. Report this plot.
9. Set the Noise to none. Set the processing to normalize and then to rectification. Change the scale to show the normalized signal. Report these plots.

10. Try different combinations of processing and filtering and examine the effects in the frequency and time domains. Note any interesting findings. Please be aware that for the processing box, processing always occurs in the following order: noise added, processing, then mathematics.

Data Analysis

Note any interesting observations that were made in the processing and application section of this laboratory.

Review all of your screen captures in your report and explain why the signal appears as it does in each of the plots based on the parameters that you had turned on.

Discussion Questions

1. Often times in biomechanics, transducers are used to record the angle of a joint during motion. From your analysis of what happened to 60Hz noise when the derivative and integrals were taken, explain why even a small amount of noise in the signal may prohibit someone from calculating the angular velocity and acceleration of the joint using the angle data from the transducer.
2. Explain the difference in the spectral analysis plots when the data collection interval was increased.
3. Explain why complete elimination of noise artifacts in signals can be so difficult to remove.
4. Explain how a bandpass or bandstop filter can be constructed by combining a lowpass and highpass filter together. If a bandpass filter for 10-25 Hz is desired, what cutoffs are necessary for the HP and LP filters? And for a bandstop filter of 50-60 Hz?
5. In the Background, walk through each of the steps and show why the transfer function for the lowpass filter is the one listed there. What is the cutoff frequency of this filter?
6. Thinking back to your physics classes, why is it that the capacitor has zero resistance for infinite frequency and the inductor has infinite resistance for infinite frequency?

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7. Solve the transfer function for the first order highpass filter and show that it indeed does pass high frequencies and attenuates low ones.
8. Using MATLAB, complete and plot an FFT analysis of the saved data file "60HzNoise". The analysis should be performed over the Processed Data Channel.
9. Plot the phase for a first order highpass filter with a cutoff frequency of 50 Hz. What are the R and C values required here for this desired cutoff frequency?
10. In this laboratory, you learned that digital filtering is a useful tool to filter digitally acquired data. What are some drawbacks to using digital filtering? In which instances would it be better to have an analog filter instead of a digital filter?
11. Why would a filter with linear phase be more ideal for DSP filtering than one that is not? What is the tradeoff between using a linear phase filter vs. a filter with non-linear phase?
12. Design and draw a simple, first order lowpass filter with realistic filter component (resistor, capacitor) values, given a voltage source of 5V, with a cutoff frequency of $\omega_c = 30\text{Hz}$. Design and draw a simple, first order highpass filter with the same restrictions, but $\omega_c = 13\text{Hz}$. Recall that simple filters can be cascaded to make specialized filters to meet the need of a specific application. If you were to cascade the two simple filters you just made, what application (pertaining to this laboratory) would you apply it to?

References

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