Analog vs. Digital Filtering of Data

By Sabrina Miller, ECE '20

Introduction

My senior design project, Smartbell, requires data collection from an accelerometer. The data is intended to eventually be classified into type of movement and number of reps via a machine learning algorithm. In order for proper classification, much of the noisiness must be eliminated, otherwise it may be hard for the machine learning algorithm to differentiate between movements and characterize number of reps. This report aims to explore the difference between analog and digital filtering of data, to establish their pros and cons, and to decide on the method that best suits Smartbell.

The Fundamentals of Raw Data

What is Noise?

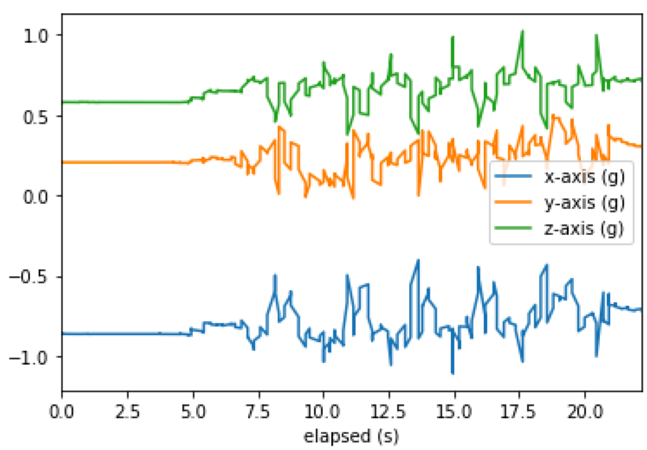

Figure 1. Raw x, y, and z axis deadlift data from accelerometer

Fig. 1 shows raw accelerometer data from a set of five deadlifts. The x-axis data gives a fairly clear sense of the number of reps completed, if looking at the tallest peaks of the data. However, the surrounding data is noisy. Noise is characterized by meaningless data points – the only data that are useful are the peaks above a certain threshold, as those represent sharp fluctuations in acceleration. In order to properly count the number of deadlifts performed, the smaller amplitude fluctuations must be minimized.

Dealing with Noise: Filters

When dealing with noisy data, there are various ways of going about filtering that data. The two overarching categories in which one can filter data are analog filtering and digital filtering. Analog filtering involves physical hardware that alters analog signals before they are passed off to other components to be processed. Digital filtering involves passing analog data to a processor that then runs code to digitally filter the data.

Digital Filtering

Advantages

The advantages to digital filtering are numerous. The most apparent is that digital filters require less hardware, as they are done on a processor. This makes them very versatile and applicable in any system with a processor. In addition to lowering the cost, extra hardware has the added disadvantages of being affected by external factors such as temperature, humidity, and general wear and tear. Advanced and specialized filters can be applied to data, so long as the processing power exists to perform the filtering methods. Digital filters are software programmable, which makes them easy to bring up and test. Being able to quickly program and prototype digital filters also contributes to their versatility (Taranovich).

Disadvantages

The standard disadvantage of a digital filter is that digital filters are significantly slower than analog filters (Smith). Digital filters introduce additional latency into a system, as the analog data that comes out of the hardware must be processed on a computer before it is filtered as desired. It is also difficult handle large frequency ranges with digital filters. The sampling rate to capture one cycle at 0.01 Hz must be extremely high (20 million points). This is a very large amount of data for just one cycle, let alone multiple.

Analog Filters

Advantages

Analog filters have the main advantage of speed. Filtering with hardware means that the signal coming out of the physical filter is the final signal. Analog filters also provide a greater dynamic range for frequency. It is relatively easy to design a frequency filtering circuit with an operational amplifier (op amp) that can handle signals that have frequencies between 0.01 Hz and 100 kHz (Smith).

Disadvantages

Analog filters require physical space, so they must be used sparingly if space is an issue. As with any physical hardware, if there is an issue with its design, it is much harder to fix once a product is deployed, as hardware cannot be altered over the air.

Types of Analog Filters

There are three types of basic frequency filters: low-pass, high-pass, and bandpass. These three filter types can be implemented both in analog and digital. Low-pass filters are intended to filter out low frequencies, high-pass filters are intended to filter out high frequencies, and bandpass filters are intended to filter out frequencies below and above a certain frequency range. When it comes to implementing analog frequency filters, the first thing to consider is the type of components that will be used to create the filter. There are two main categories of frequency filters: active and passive. Passive filters use passive components, meaning the filtering components used require no external power (other than the power provided by the signal being filtered itself). Examples of such components are resistors, inductors, and capacitors. Active filters use components that require external power, such as op amps (O'Leary).

Passive Filters

Passive filters can be useful in providing a simple method to do simple filtering. The ideology behind passive filters rests in the way inductors and capacitors respond to changes in frequency.

Impedance	$Z_C = \frac{1}{j2\pi fC}$	$Z_L = j2\pi fL$	$Z_R = R$

Figure 2. Impedance of various electrical components, respectively: capacitor, inductor, resistor (Lee)

The impedance of a resistance is entirely real, so resistors (theoretically) act the same no matter what frequency the signal they are dealing with is at. The impedances of inductors and capacitors are entirely imaginary, so their behavior differs greatly depending on the frequency of the signal they are dealing with.

$$\lim_{f \to \infty} \frac{1}{j2\pi fC} = 0$$
$$\lim_{f \to 0} \frac{1}{j2\pi fC} = \infty$$
$$\lim_{f \to \infty} j2\pi fL = \infty$$
$$\lim_{f \to 0} j2\pi fL = 0$$

Figure 3. Behavior of passive components at 0 and infinite frequency

As the frequency on a capacitor approaches infinity, the impedance becomes 0. This means that at high frequencies, capacitors act as a short circuit. The same is true for inductors at low frequencies. This can be abstracted to say that capacitors act as low-pass filters and inductors act as high-pass filters. Using these components as such is especially helpful in filtering out small fluctuations in DC signals.

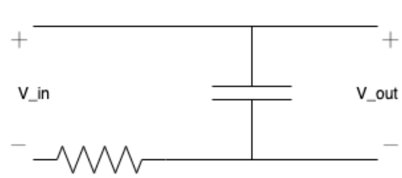

Figure 4. Passive low-pass filter

Fig. 4 shows a passive low-pass filter. As the frequency of V_in increases, the impedance of the capacitor will decrease, meaning the capacitor will begin to act more and more like a short circuit. Thus, the voltage drop across the capacitor will approach zero.

Active Filters

There are many active frequency filter configurations, and many of them use op amps. Op amps have the advantage of having (theoretically) infinite input impedance and zero output impedance. This essentially means that the op amp does not act as a load.

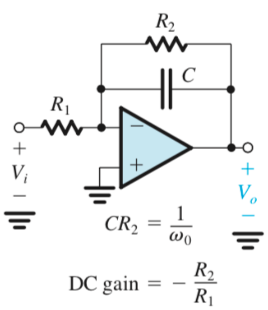

Figure 5. Active low-pass filter (Sedra)

Fig. 4 shows a realization of an active low-pass filter with amplification. Depending on the capacitor and resistor values used, it works similarly to the passive filter in Fig. 3. However, the advantages (and disadvantages) of an active filter apply.

Two examples of other common types of active filters are the Butterworth and Chebyshev filters. A Butterworth filter is a monotonically decreasing low-pass filter. There are no ripples in the passbands, meaning that it is a good filtering option for signals that require precise levels across the entire passband (Kugelstadt).

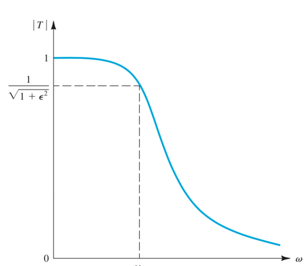

Figure 6. Magnitude response of a Butterworth filter (Sedra)

Chebyshev filters display an even ripple pattern in the passband and monotonically decrease in the stopband. They are designed to have a more steep rolloff after the cutoff frequency. Cutoff frequency is typically defined as the frequency at which the magnitude of the transfer function is attenuated beyond a certain amplitude.

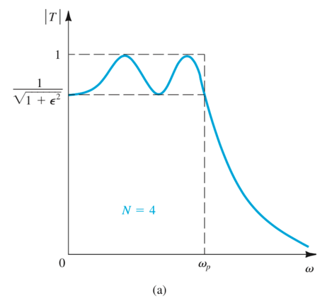

Figure 7. Magnitude response of an even-order Chebyshev filter (Sedra)

Note that the filter shown in Fig. 7 is even-ordered, meaning that at a frequency of 0, there is maximum magnitude deviation.

Conclusion: Which is Better?

For the purposes of our senior design project, digital filters seem to be the best option for filtering our data. This is because the filtering is not time sensitive. The data can be filtered after it is collected, and a finely tuned digital filter will be helpful in getting the clean data that is desired. Additionally, extra hardware would increase the size of the sensor that is designed and it is not necessary for the filter to be able to handle a large frequency range, as the data we are collecting is not subject to frequencies much above a few kHz. We can adjust the sampling rate to be fast enough to avoid data loss, but slow enough to avoid saturation of data in the form of collecting too many data points for our processors to handle.

A Note on Digital Filtering

Digital frequency filters require sampling in the time domain. All the magnitude responses that have been presented in this paper are shown in the frequency spectrum. To transform something in the time domain to the frequency domain, a Fourier transform is typically used. Once a signal is in the frequency domain, it can simply be multiplied by an ideal low-pass or high pass filter to achieve the desired filtered status. Once this is done, the signal has to go back to the time domain, but this is a different issue.

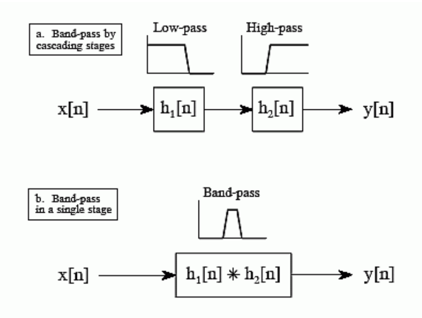

Figure 8. Construction of a digital band-pass filter

References

1. Kugelstadt, Thomas. “Active Filter Design Techniques.” *Op Amps For Everyone*, edited by Ron Mancini, Texas Instruments, 2002, pp. 16–1-16–23.

2. Lee, Chuan-Zheng. *Circuits in the Frequency Domain*. Stanford University, 2 Aug. 2017.

3. O'Leary, Sean. “Active vs. Passive Filters.” *Celtic Engineering Solutions*, 7 Dec. 2016.

4. Oppenheim, Alan V., and Alan S. Willsky. *Signals & Systems*. Prentice Hall International, Inc. , 1983.

5. Sedra, Adel S., and Kenneth C. Smith. *Microelectronic Circuits*. Oxford University Press, 2015.

6. Smith, Steven W. *The Scientist and Engineer's Guide to Digital Signal Processing.* California Technical Pub., 1997.

7. Taranovich, S. (2013, July 15). *Integration Choices: Analog Filters vs. Digital Filters*. Retrieved December 19, 2019.