Northeastern University Department of Electrical and Computer Engineering

Final Project
Signal Analysis and Filtering

EECE 2520 Prof. Purnima Ratilal Makris

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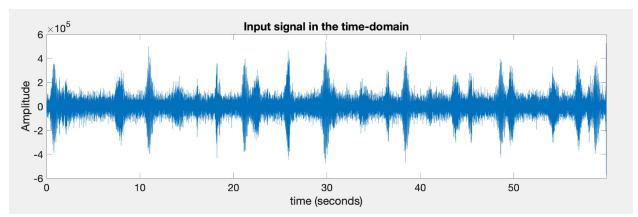
Fall 2020 Semester December 9, 2020

Introduction and Objectives

The goal of this semester final project was to utilize the concepts learned throughout the linear systems course to manipulate signals in the time and frequency domains within a real-world application. With input data regarding simultaneous vocalization signals of fin whale song calls and humpback whale song calls, the object of this assignment was to isolate the lower-frequency fin whale song calls from the higher-frequency humpback whale song calls. This was accomplished through the use of low-pass filters. MATLAB was used to perform the calculations and plot various time and frequency domain data.

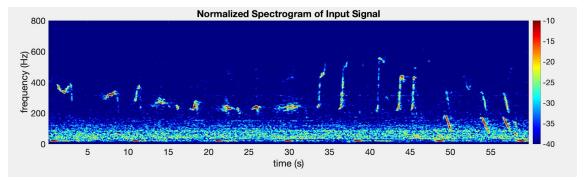
Results and Analysis

<u>Part I</u>: The goal of this portion of the project was to understand the duration and frequency content of the calls from the fin whale and the humpback whale. Utilizing the input data, which was 1600 Hz. time-domain signal, the data was plotted in MATLAB.



a)

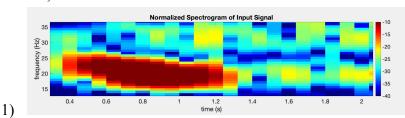
i) The preceding is a plot of the input signal in the time-domain, with the vertical axis being the amplitude of the signal and the horizontal axis stating the time, in seconds.



b)

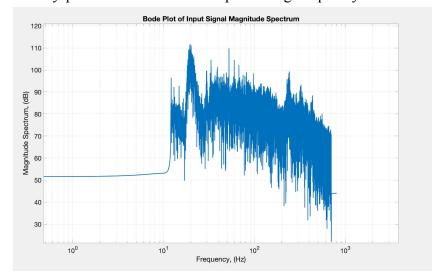
i) The preceding figure is a plot of a normalized spectrogram image of this same time-domain signal data. However here, the vertical axis is the frequency of the marine mammal sound, while the horizontal axis is the time, in seconds.

ii) Zoomed in, the fin whale call can be visualized as:



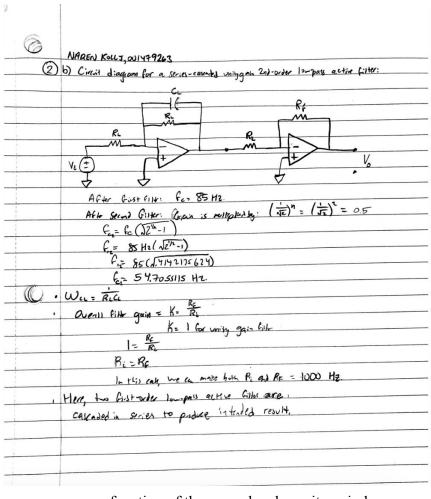
- (a) The duration of the fin whale call is around one second. From the preceding figure, the range is from 0.4 seconds to 1.2 seconds, so this duration is 0.8 seconds in total.
- (b) The lower frequency range for the fin whale call is approximately 15 Hz. This makes sense, as the call is centered around 20 Hz.
- (c) The higher frequency range for the fin whale call is approximately 25 Hz. This makes sense, as the call is centered around 20 Hz.
- (d) The bandwidth is the difference between the higher and lower frequencies. Thus, the bandwidth of the fin whale call is approximately 10 Hz.
- iii) The humpback whale, in contrast, have different duration and especially frequency ranges for each of the calls.
 - 1) The typical duration of the humpback whale call is one second. However, there are calls that last from less than one second to calls that last nearly two seconds.
 - 2) The lower frequency range for the humpback whale call is approximately 85 Hz. This makes sense, as the lowest frequency a humpback whale can emit sounds at is 50 Hz.
 - 3) The higher frequency range for the humpback whale call is approximately 560 Hz. This makes sense, as the highest frequency a humpback whale can emit sounds at is 700 Hz.
 - 4) The bandwidth is the difference between the higher and lower frequencies. Thus, the bandwidth of the humpback whale call is approximately 475 Hz.
- c) Using MATLAB's audio player function at normal speed, the input signal sound was listened to. It was characterized by a deep rumbling sound, the sound of the ocean, punctuated by short, high-frequency sounds. These short sounds were determined to be the song calls of the humpback whales. The fin whale calls could not be heard, as the frequency was too low, and were consumed by the higher-frequency sounds.
- d) Using MATLAB's audio player function at twelve times the normal speed, there wre chirping sounds as well as repeated 'beeping' sounds. Due to the higher speed, the chirping sounds were determined to be the higher-frequency sound of the humpback whale, while the repeated beeping sounds were determined to be the song calls of the fin whale. Compared to the sound played at normal speed, the higher frequency sounds were found to be much higher, while the lower frequency sounds of the fin whale were

- determined to be audible higher and thus perceivable as well. Increasing the speed of the audio thus increased the frequency perceived of the marine mammal sounds as well.
- e) Utilizing the fft (Fast Fourier Transform) and fftshift MATLAB functions to perform Fourier transforms on the time-domain signal input data, a Fourier transform for the input signal was determined. The following is a Bode plot of this data, with the vertical axis representing the input signal Fourier transform magnitude, and the logarithmically-plotted horizontal axis representing frequency in Hz.:



<u>Part II</u>: The second part of this final project assignment was the design of a second-order unity gain low-pass active filter to eliminate the higher frequency humpback whale call, thus providing only the fin whale calls. This is possible because low-pass filters allow signals with frequencies below the cutoff frequency to pass, and they also dampen any signals above the cutoff frequency. This filter was designed by cascading two first-order filters unity gain low-pass active filters in series.

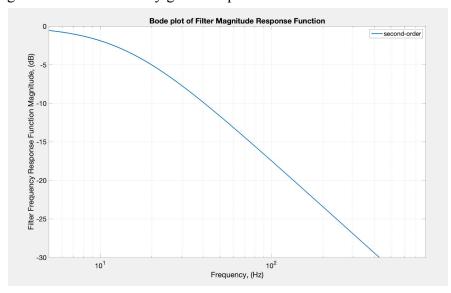
- a) In order to eliminate the humpback whale song call, it was necessary to determine the minimum frequency of the humpback whale call. From Part I, this was found to be 85 Hz. Thus, the cutoff frequency was 85 Hz. The half-power frequency with, *fhp*, was calculated for the low-pass filter. This is an important value, as it is the frequency when the power is half of its peak. Using the formula $fhp = fc(\sqrt{2^{1/n} 1})$, with fc at 85 Hz. and n=2, the half-power frequency was calculated to be 54.705515 Hz.
- b) The following is a diagram illustrating the circuit diagram for a series-cascaded second-order unity gain low-pass active filter, with the cut-off frequency of each individual first-order low-pass filter operational amplifier circuits used, as well as sample resistor values in order to have the filter design maintain a unity gain (next page).



c) The frequency response function of the second-order unity gain low-pass active filter is:

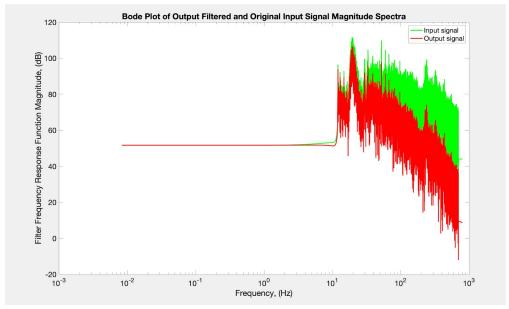
i)
$$H = (-1 * 2 * \Pi * fhp) \div (s + 2 * \Pi * fhp)$$

d) The following is a Bode plot, displaying the magnitude frequency response function of the designed second-order unity gain low-pass active filter:

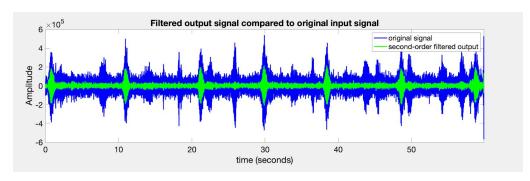


<u>Part III</u>: The third portion of this project was integrating the previously designed second-order unity gain low-pass active filter into the frequency domain. This was used to further eliminate the higher frequency humpback whale song call, in order to isolate the fin whale song call. Then, higher-order low-pass filters were designed in MATLAB to fine-tune the higher frequency elimination.

- a) The output signal was represented as a Fourier transform, to place it in the frequency domain from the time domain. This was accomplished through the use of the MATLAB fft (Fast Fourier Transform) and fftshift functions.
- b) The following is a Bode plot of the output signal magnitude spectrum, as well as the plot of the input signal magnitude spectrum obtained in part I(e). They are both plotted on the same graph for ease of reference.

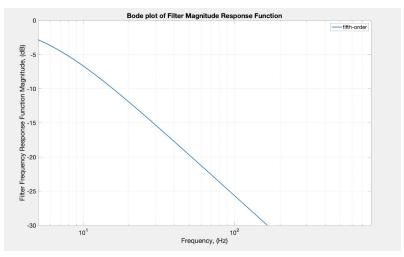


- ii) As it can be seen, the output signal represents only a portion of the input signal magnitude, due to higher frequencies being filtered out through the second-order unity gain low-pass active filter. This enables the elimination of higher-frequency signals, in this case, the humpback whale song calls.
- c) Next, the time-domain output signal was calculated using the MATLAB functions ifft (Inverse Fast Fourier Transform) and ifftshift. This enabled the output signal to be represented in the time-domain from the frequency-domain. The following is a plot of the time-domain output signal as well as the time-domain input signal. As it can be seen, much of the input signal was filtered out when producing the output signal. However, the humpback whale song calls are still present in the data, and must be filtered out further in order to isolate the fin whale song calls (next page).



d) The output signal sound was then played at twelve times the regular speed. It was found to be much easier to hear the fin whale sounds. As much of the higher frequencies of the humpback whale were filtered out by the second-order unity gain low-pass active filter, the repeated lower frequency 'beeps' of the fin whale song call was heard with much more clarity. However, there was still 'chirping' from the humpback whale song calls, which needed to be eliminated to produce a more isolated output signal with the fin whale song calls.

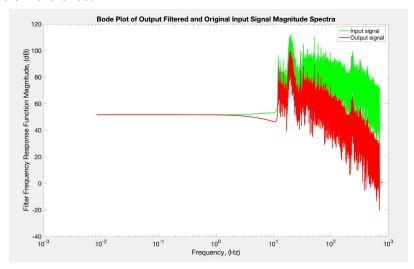
- e) While using the second-order unity gain low-pass active filter provided for a more filtered output signal, it was not sufficient to completely eliminate the humpback whale song calls. Thus, it was still hard to listen to the fin whale song calls. In order to adequately hear the fin whale, the humpback whale song call frequencies needed to be further eliminated. This was accomplished through designing higher-order low-pass filters; Additional filters continued to eliminate the higher frequencies by lowering the cutoff frequency used by the circuit. Third-order and fourth-order unity gain low-pass filters were then utilized, however, these did not eliminate the humpback whale sounds in a sufficient manner. A fifth-order unity gain low-pass active filter provided the sufficient performance required to filter out the higher-frequency humpback whale song calls, allowing for the fin whale song calls to be heard more clearly. Utilizing the equation in part II(a), $fhp = fc(\sqrt{2^{1/n} 1})$, with fc at 85 Hz. and n=5, the half-power frequency was calculated to be 32.77721182 Hz.
 - i) The following is a Bode plot, displaying the magnitude frequency response function of the designed fifth-order unity gain low-pass active filter:



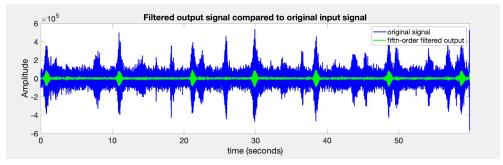
1)

1)

- 2) As displayed, the Bode plot approaches a lower frequency here, compared to the Bode plot of the fifth-order unity gain low-pass active filter. This was due to lower cutoff frequency allowing for less frequency magnitude response.
- ii) The following is a Bode plot of the output signal magnitude spectrum using the fifth-order unity gain low-pass active filter, as well as the plot of the input signal magnitude spectrum obtained in part I(e). They are both plotted on the same graph for ease of reference.



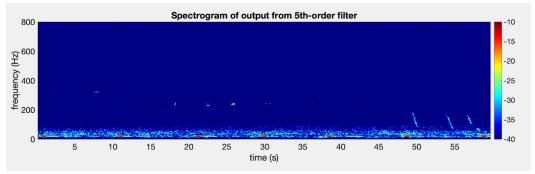
- 2) As displayed in the plot above, the output signal is filtered more compare to the second-order filter and thus represents a smaller fraction of the original input signal. That means more of the higher frequencies are filtered out when processed through this fifth-order filter, allowing for further isolation of the lower frequencies of the fin whale call.
- iii) The following is a plot of the time-domain output signal from the fifth-order low-pass active filter as well as the time-domain input signal. As displayed, the output signal is filtered more, allowing for isolation of the lower frequencies.



iv) The follwoing figure is a plot of a normalized spectrogram image of the fifth-order low-pass active filtered time-domain data. As in the spectrogram above, the vertical axis is the frequency of the marine mammal sound, while the horizontal axis is the time, in seconds (next page).

1)

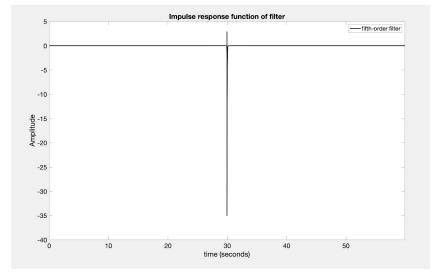
1)



- 2) Compared to the original time-domain signal input spectrogram, this visual of the fifth-order filtered signal data demonstrates the power of the low pass filter in eliminating higher frequencies to isolate lower frequencies. Here, a majority of humpback whale song call frequencies have been removed, leaving only the fin whale frequencies below.
- f) With 1k Ohm resistors used in the filter design, the capacitance value needed to achieve the half-power width specified in part II(a) for the fifth-order filter designed was determined through the equation in part II. This value was 3 F.

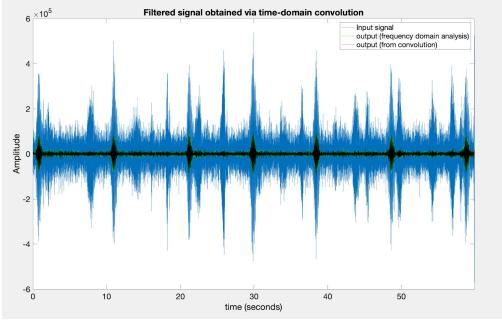
<u>Part IV</u>: The final portion of this assignment was utilizing another form of filtering within the time-domain: convolution.

a) Utilizing the MATLAB functions fftshift, ifft, and ifftshift, the fifth-order unity gain low-pass active filter's impulse response function was calculated using the built-in inverse Fourier transform functions. The plot of this impulse response function is below:



b) The output signal was determined through the convolution of the previous impulse response function and the original input signal. This was calculated through the MATLAB function conv (convolution of vectors), and can be viewed in the plot below (next page):

i)



ii) As viewed in the plot above, the output signal determined through convolution (in black) matched the output signal determined through analysis within the frequency domain (in green).