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I. Abstract

The general synopsis of the experiment performed was to investigate basic concepts of acoustic measurements. This included understanding the extents of data acquisitions, aliasing, and quantization. These ideas promoted conclusions regarding frequency domain analysis of signals in addition to filters and the purposes of them.

From the laboratory experiment, it was concluded that a band-pass filter with the addition of the Fast Fourier Transform can generate responses from a dynamic sound signal. The electret microphone element produced voltage signals which was converted into amplitude responses with respect to time and frequency. While performing the experiment, analytical values for the quantization error was used to calculate the resolution of the DAQ board to be 12 bits. Furthermore, the increasing of the decimation factor related to a muffled and lower pitched playback sound.

II. Introduction

The experiment to be conducted is in relations to sound measurements, data acquisitions, and filtering. The overarching goal of the experiment is to grasp a basic understanding of these terminologies and successfully applying them. One should be able to identify the limitations of these data acquisition methods, such as aliasing and quantization, as well as understanding what the purposes of filters are in relation to frequency domain analysis of signals. All of these will be achieved under basic concepts of acoustic measurements.

These ideas are important for engineers to understand. A large majority of data types are dynamic, meaning that they change over time. Understanding how to implement data measurements through frequency domain analysis is crucial when developing complex relationships. A successful experimentation would be fully understanding the purposes of filters as well as being able to understand the methods and results of data acquisition methods.

III. Theory

To experiment with sound, a basic understanding of microphones is needed. For the following experiment, electret microphone elements are used. These devices can convert sound waves in forms of pressure, into an electrical signal. An example of an electret microphone element is shown in Appendix A, Figure 1. The device alters the spacings between an internal capacitor when it is under vibration. This is then converted into an electrical signal.

Just as beneficial the data acquisition system is to a modern laboratory, these systems are also limited in certain aspects. The term ‘quantization’ refers to the method of converting infinite ‘smooth’ values into discrete, finite values. This loss in accuracy can be related to the resolution of the analog-to-digital converter (ADC). This error is calculated with the following relationship:

$$\varepsilon = \frac{\Delta V}{2^n} \tag{1}$$

In addition to the quantization error, ACD's often face an issue of aliasing. This occurs when the measured signal's frequency is less than half of the sampling frequency (known as the Nyquist frequency). This phenomenon will deteriorate the quality and structure of the original signal, possibly producing inaccurate frequencies and graphs. An example of this can be seen in Appendix B, Figure 2.

A special transformation that is used frequently in dynamic response is the Fast Fourier Transform (FFT). The FFT can convert a conventional amplitude response with respect to time, into a more desirable amplitude and frequency relationship. This is mathematically processed through representing an output signal as multiples of harmonic waves with varying amplitudes and phase shifts.

Used heavily in this experiment and through many dynamic response systems, a band-pass filter can clear out undesired outputs. For example, a band-pass filter can remove signals outside of a certain frequency to ensure background noises are not present. Although this idea seems straightforward, in the real world, this will not be possible. Only an ideal band-pass filter can remove all undesired responses. All band-pass filters used have some form of cutoff; A representation of this can be seen in Appendix C, Figure 10.

IV. Procedure

The experiment is relatively straightforward once the equipment is set up and tested. The electret microphone element is used to collect the desired sound inputs. The setup is shown below in Figure 3 and can be easily created on a breadboard.

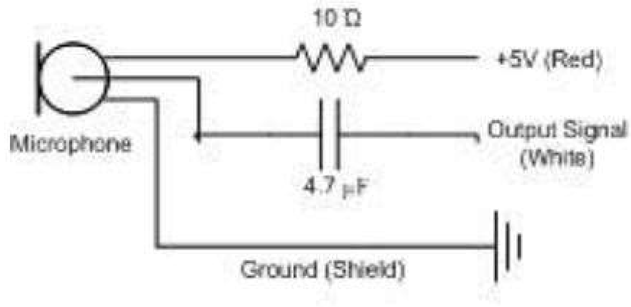

Figure 3: Electret Microphone Circuit

The LabView VI file is used to process and collect the dynamic data. Only a few settings are altered throughout the experiment; The decimation factor setting, sampled signal setting, and the ability to turn on and off a band-pass filter is present.

The experiment is then a simple cycle between providing an audio sample for the microphone and observing the outputs. The outputs will be either in an audial, playback, or through a graph created in the VI file.

V. Results and Discussion

From the first sampling rate of 10 kHz and with the number of samples set at 50000, the Nyquist frequency is calculated as the following:

$$f_N = \frac{f_s}{2} = \frac{10000\ Hz}{2} = 5000\ Hz \tag{2}$$

This Nyquist frequency suggests that a standard singing or whistling pitch will not fall below this value of 5000 Hz, as the average whistle from a human is in the range of 500 – 5000 Hz (Nilsson, Bartunek, Nordberg, Claesson, 2008).

When an average pitch for a whistle is introduced to the microphone element, the voltage signal changes with respect to the sound volume. When a louder volume is introduced, the voltage was larger respect to a softer volume. This makes intuitive sense as the larger vibrations increases the gaps between the capacitance inside the microphone. This can be visualized in Figure 4.

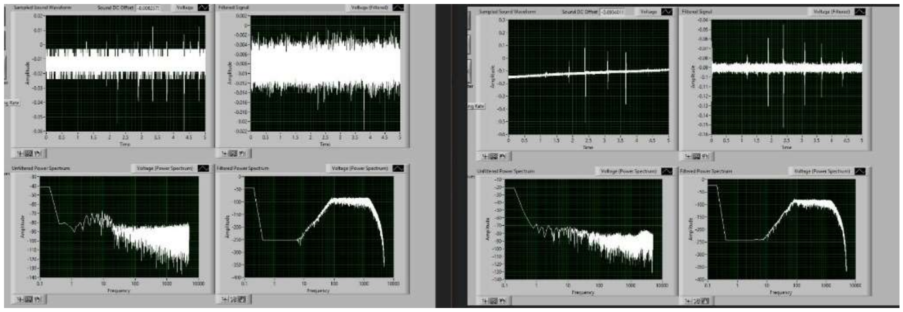

Figure 4: A soft (left) signal compared to a loud (right) signal

In addition, a standard 440 Hz, concert "A", was vocally attempted to verify the accuracy of the signal. The accuracy is close, as seen in Figure 5, showing the output signal.

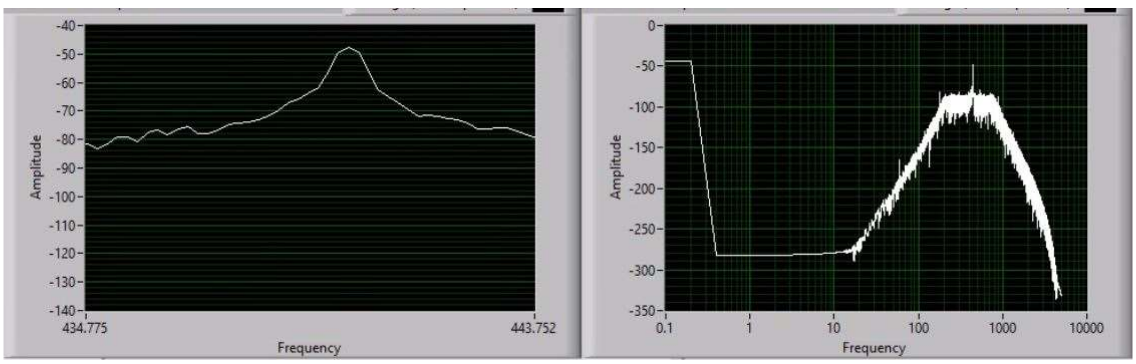

Figure 5: Concert "A" 440 Hz signal

It is observed that if a lower sampling rate is used, larger amounts of 'noise' will be detected in a frequency response. The larger the sample rate is, the more accurate the results will be. However, this usually will require more expensive or larger equipment. This is due to the decrease in quantization.

While observing the display of the voltage signal, visible quantization is identifiable. Seen in Figure 6, a steady value of 0 is being approximated about small values.

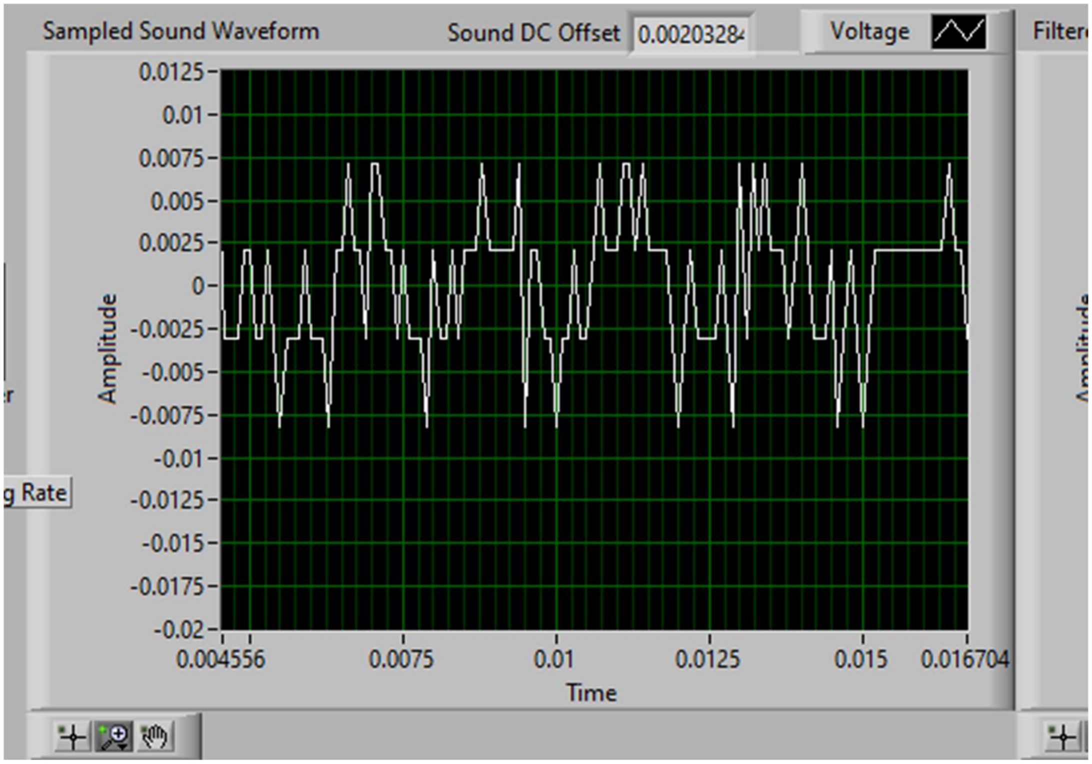

Figure 6: Observing quantization

From Figure 6, the calculation of the minimum voltage resolution from the display using equation (1) is done. Knowing that the change in voltage is 10 Volts, and the smallest increment in amplitude is observed to be 0.0025, the resolutions is calculated as follows:

$$0.0025 = \frac{10}{2^n} \tag{3}$$

$$n = 11.965 \tag{4}$$

With an n value of 11.965, the approximation can be made that the DAQ board has a resolution of 12 bits.

Another form of error in the electret microphone element can be observed through a windy day. As the electret microphone element is dependent on the vibrations supplied, wind will cause extreme muffles and spikes in the output signal. A human voice will likely not be heard when using this electret microphone element on a windy day. The frequency response reflects this in a large amount of 'noise' present.

While attempting to understand the usages of a filter, a band-pass filter is enabled. When enabling a band-pass filter, the plot in Figure 7 shows clearly that the effects of the filter is evident.

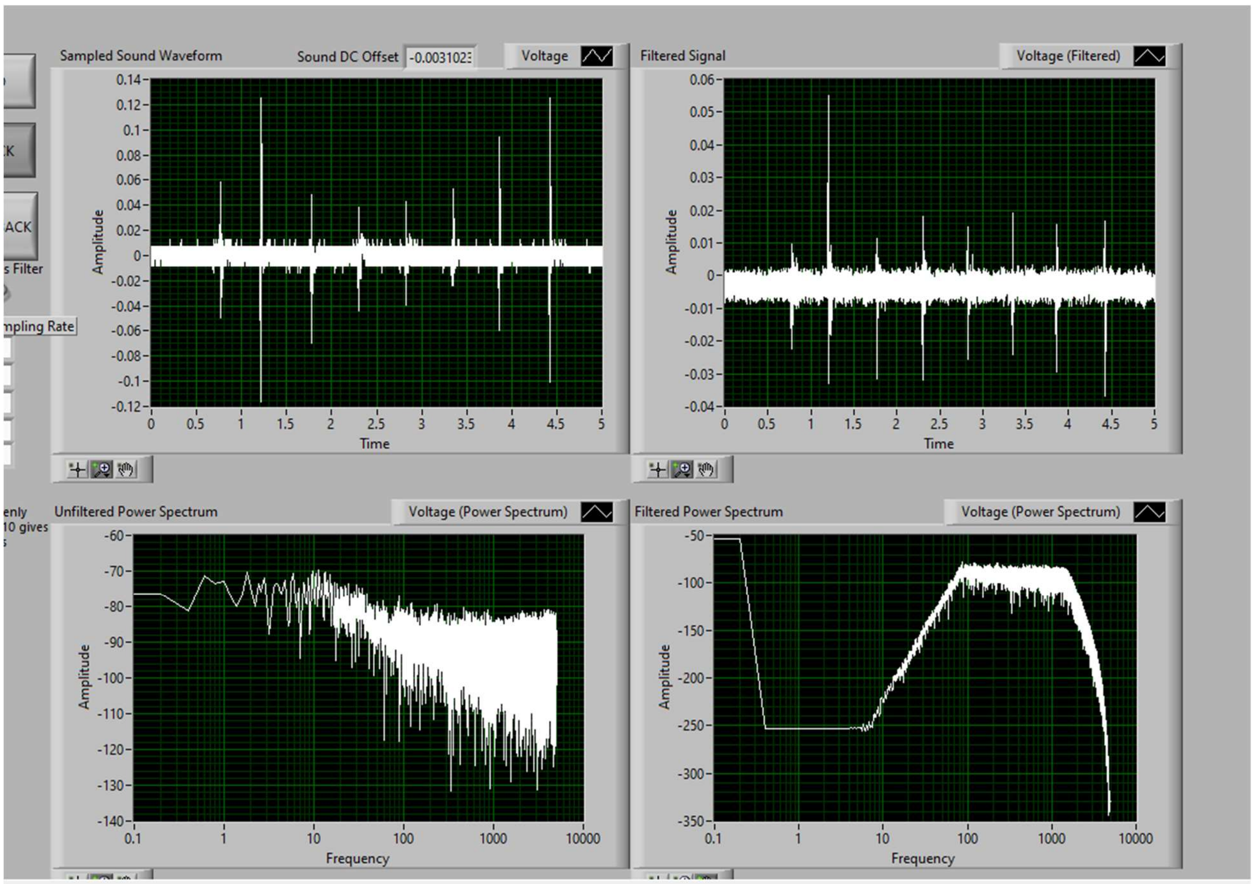

Figure 7: Addition of a band-pass filter

From Figure 7, the noticeable different values of frequencies are compared to frequencies without using a filter. This result could vary depending on the type of filter used. For example, if an ideal band-pass filter is used, the frequency would cut off immediately without a drop off in the filtered power spectrum, unlike the one seen in Figure 7.

However, a band-pass filter is not required to produce similar results. Instead of a band-pass filter, a combination of a low-pass filter and high-pass filter can be used to achieve the same results. This is since a band-pass filter filters out responses from a given range, and a low-pass filter and high-pass filter can be combined to produce a similar effect.

Even without playing back this signal, another way to determine if a high frequency is present is with the usage of a Fast Fourier Transform (FFT). The FFT can transform the amplitude and time signal values into a desirable amplitude and frequency comparison.

Another adjustment that can be made is the cut-off frequencies. While changing these values, a noticeable change in the playback is evident. As seen in Table 1, the low cut-off and high cut-off frequency relates to a description while clapping.

***Table 1**: Qualitative Descriptions of Varying Cut-off Frequencies*

Low cut-off (Hz)	High cut-off (Hz)	Qualitative Description
1	5000	More background noise present.
1	3000	The sound begins to clear up.
100	3000	This is the clearest.
100	1000	The sound begins to muffle.
300	1000	The sound is more muffled, but clap is still audible.

The "Decimating Factor" is also able to be adjusted. This refers to the number to data points taken from the original signal. As seen in Figure 7, a table shows the qualitative description of the audio playback when adjusting the decimation factor.

***Table 2**: Qualitative Descriptions of Varying Decimation Factors*

Decimation Factor	Qualitative Description
1	This sounds exactly same as 100 (low) / 3000 (high) in Table 1.
2	The sound provides a slightly different noise.
5	The sound is quieter and muffled.
8	The sound is now really muffled.
10	The clapping sound is starting to disappear.

Combining Table 1 and Table 2, Table 3 is created. This allows for a verification of the effects of aliasing. After reaching a decimation factor of approximately 50, the play back becomes nearly inaudible.

Table 3: *Qualitative Descriptions of Varying Cut-off Frequencies and Decimation Factors*

Low cut-off (Hz)	High cut-off (Hz)	Decimation Factor	Qualitative Description
1	5000	1	The sound is very similar, there is little to no difference.
1	3000	1	The sounds the same to the previous settings. Not too many notable differences.
600	3000	1	The pitch of the play back becomes lower and deeper.
600	3000	5	The tone becomes very low and deep.
600	3000	10	The tone is extremely low and deep.

VI. Conclusions

From the experiment regarding dynamic sound data acquisition, a few basic methods and terminologies were understood. Quantization is determined to be the loss in accuracy when converting from an analog device into a digital response, and this error can be calculated with equation (1). Aliasing closely ties into quantization as aliasing occurs when the digital response is inaccurate to the actual response due to inadequate sampling rates. The band-pass filter was also determined to selectively pick data which falls into the measuring range, reducing any undesired noise.

It was concluded that a band-pass filter with a low cut-off of 100 Hz and a high cut-off of 3000 Hz provided the clearest playback sound when performing the experiment. Furthermore, a larger decimation factor generally deepens the playback tone. When an approximate decimation factor of 50 was achieved, almost no notable sound is recognized.

VII. References

B. Rasmussen, M. Adams, B. Wang, C. Lee. "LAB 06 SOUND MEASUREMENT, DATA ACQUISITION, AND FILTERING LAB INSTRUCTIONS" Texas A&M University, 2020.

M. Nilsson, J. S. Bartunek, J. Nordberg, I. Claesson. "Human Whistle Detection and Frequency Estimation" (2008) Congress on Image and Signal Processing, Sanya, Hainan, 2008, pp. 737-741.

Pollock, D. S. (2018, August). The rectangular frequency response of the ideal bandpass filter [Digital image]. Retrieved from https://www.researchgate.net/figure/The-rectangular-frequency-response-of-the-ideal-bandpass-filter-defined-on-the-interval_fig2_326995679

VIII. Appendices

A) Below is Figure 1, an electret microphone element. This is a representation of the device used in the experiment.

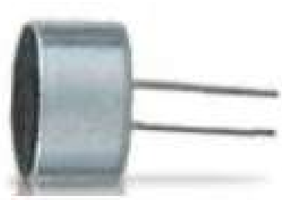

Figure 1: Electret Microphone Element (Rasmussen, Adams, Wang, Lee, 2020)

B) Below is Figure 2, showcasing the effects of aliasing. The red curve shows the original, true signal, while the blue shows the inaccurate, aliased signal. The original curve shows a very high energy signal, while the blue curve shows an elongated period. This occurs when the observed signal has a frequency below the Nyquist frequency, which is exactly half of the sampling frequency.

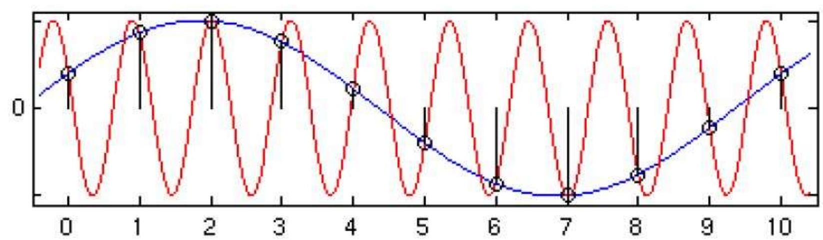

Figure 2: Effects of Aliasing (Rasmussen, Adams, Wang, Lee, 2020)

C) Below is Figure 10, showcasing the effects of an ideal band-pass filter compared to a realistic band-pass filter. The straight-line box signal represents the ideal filter, showing that only signals in that range will be allowed through. However, the curved signal shows the actual response when using a band-pass filter, with the same range, in real life.

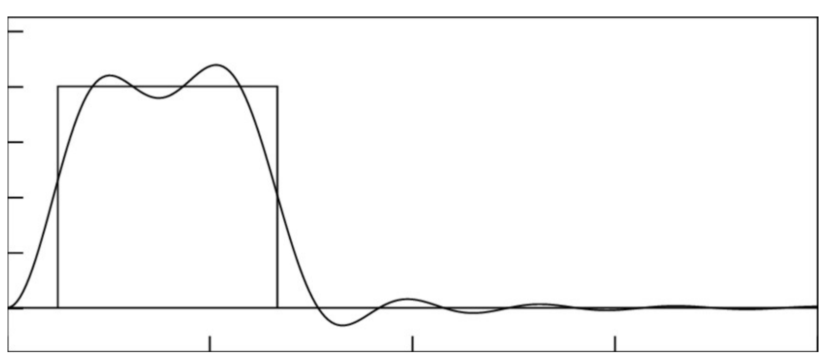

Figure 10: Band-Pass Filter (Pollock, 2018)