ME104 Lab 2: Aliasing and Data Acquisition with Oscilloscope

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1 Objective and Procedure

The first part of this lab aimed to examine aliasing, which refers to the omission of high-frequency signal components during data acquisition when the sampling rate is limited. In general, the highest frequency captured in data acquisition, f_{max}, is given by

$$f_{\text{s}} > 2f_{\text{max}} \tag{1}$$

where f_{s} is the sampling rate. In this section of the lab, we explored how sampling at rates beneath the Nyquist frequency, $2f_{\text{max}}$, can lead to poor results and aliasing. To examine this phenomenon, a sinusoid with a peak-to-peak amplitude of 2V and a frequency of 25HZ was generated via the waveform generator built into the laboratory oscilloscope. The signal was then received by a NI USB 6211 DAQ module and registered by a LabVIEW virtual instrument at varying sampling rates.

The second part of the lab introduced the use of the oscilloscope for acquisition of mixed, or noisy, periodic signals. A composite signal with several sinusoidal components was generated from the DAQ module using a provided Virtual Instrument. The signal was sampled by the oscilloscope and post-processed in MATLAB.

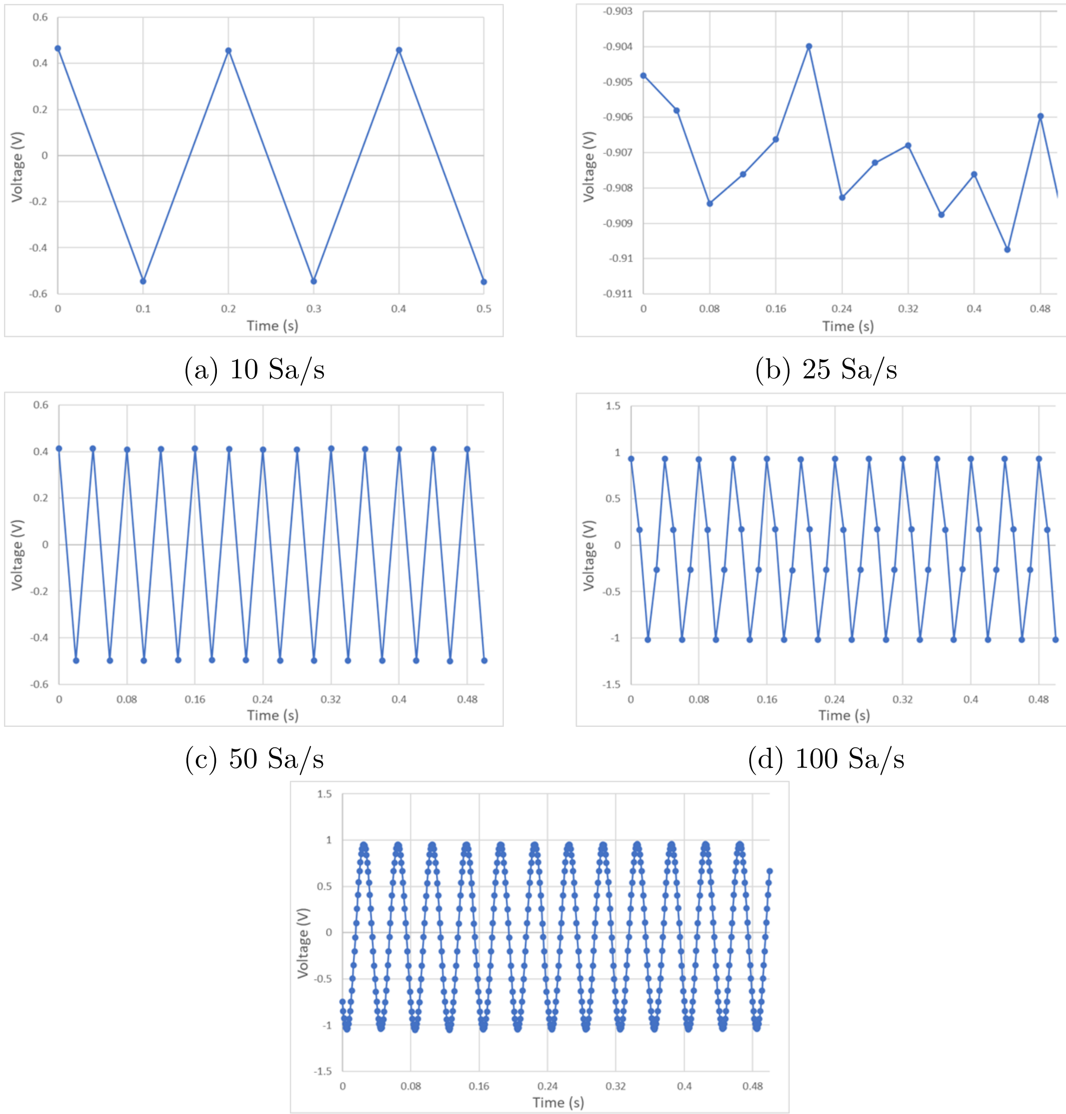
2 Results

2.1 Part 1: Aliasing

In part 1 of the lab, we generated a signal following equation 2 with the oscilloscope and sampled it using the DAQ.

$$f(t) = 1V \cdot \sin(50\pi t) \tag{2}$$

Sampling rates of 10, 25, 50, 100, and 1000 Sa/s were implemented and the number of samples was adjusted to record 500 ms of the signal for each sampling rate. The procedure for finding the number of samples is outlined in section 5. The data presented in figure 1 shows our sampled data using a first-order reconstruction with lines connecting our sampled points.



(a) 10 Sa/s

(b) 25 Sa/s

(c) 50 Sa/s

(d) 100 Sa/s

(e) 1000 Sa/s

Figure 1: Data collected via LabVIEW for a sinusoid (A=1 V, f=25 Hz) at different sampling rates using a first-order reconstruction. Aliasing occurs for sampling frequencies below 50Hz.

2.2 Part 2: Data Acquisition with Oscilloscope

In part 2 of the lab, we sampled a noisy signal from the DAQ by using the oscilloscope. The oscilloscope was set to collect data at 500 KSa/s, and the results were saved to a .csv (comma separated value) file. A screenshot from the oscilloscope is shown in figure 2.

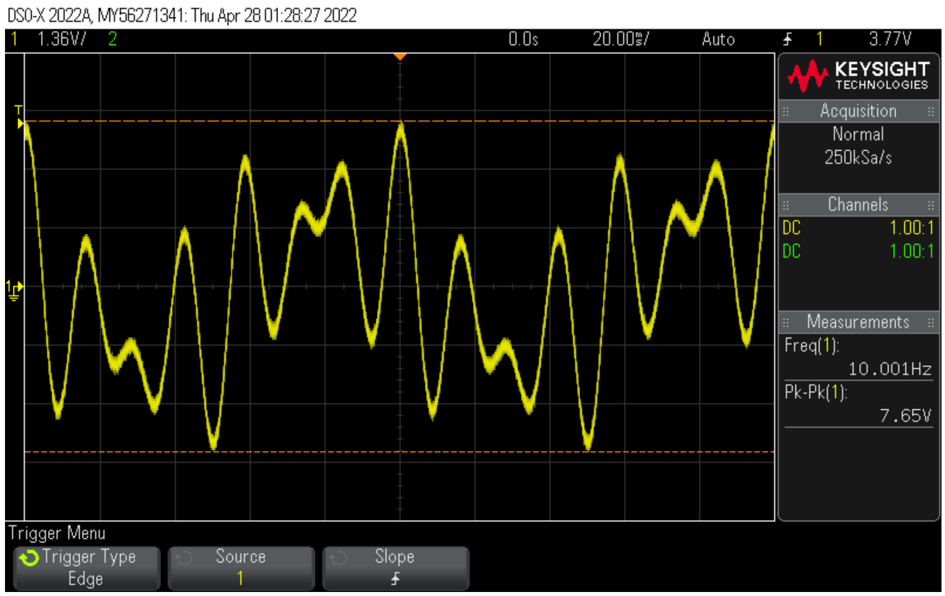


Figure 2: Input signal from the DAQ registered by the oscilloscope at 500KSa/s. Note that the attenuation factor was incorrectly set to 10.

3 Analysis

The oscilloscope data from part 2 was post-processed in MATLAB. First, the discrete Fourier transform of the data was found via the fast Fourier transform, and the two-sided amplitude spectrum was obtained by normalizing the transformed amplitudes by the number of samples. This spectrum was then converted to a one-sided amplitude spectrum by multiplying all non-DC elements by 2. Finally, the power spectrum was evaluated using

$$\begin{aligned} \text{One-Sided Amplitude RMS Spectrum} &= \frac{\text{One-Sided Amplitude Spectrum}}{\sqrt{2}} \quad \text{(For non-DC input)} \\ \text{One-Sided Power Spectrum} &= (\text{One-Sided Amplitude RMS Spectrum})^2 \end{aligned} \tag{3}$$

The MATLAB code used to carry out this process is appended to the report. Figure 3 shows the one-sided power spectrum as calculated in MATLAB and table 1 shows the dominant frequencies of the signal.

4 Discussion

4.1 Part 1: Aliasing

The results in figure 1 demonstrate the impact of aliasing at low sampling rates. Given the input frequency of 25 Hz, we can use the sampling theorem (equation 1) to conclude that the sampling rate must be greater than 50 Hz (Nyquist frequency) to fully capture the

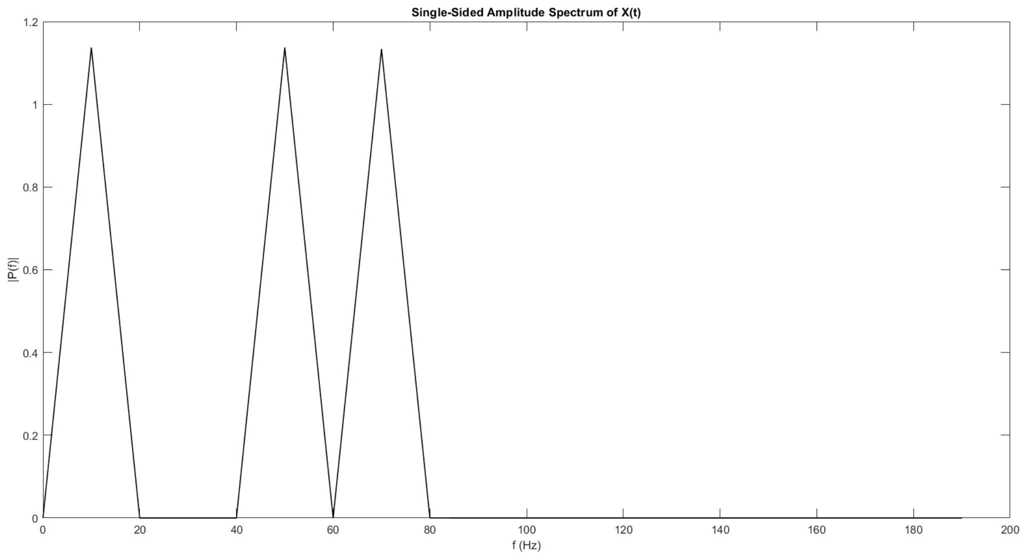


Figure 3: Amplitude spectrum of the input signal found from the discrete Fourier transform of the acquired amplitude. The graph extends to 1000 Hz, but is roughly zero for all frequencies beyond those shown.

input signal. As expected, figures 1a, 1b, and 1c are not able to fully capture the input signal. It can be seen in figure 1a that the period is 0.2 seconds, and the frequency is 5 Hz. The peak-to-peak amplitude is visibly smaller than 2 V as well. Clearly, the signal that is regenerated via this first-order reconstruction is only capturing the low frequency content of the input.

Similarly, figure 1b shows a peak-to-peak amplitude in the order of 0.01, which is far smaller than the input amplitude. This phenomenon happens when the sampling rate and the system frequency coincide. At this sampling rate, the measurement system is registering the same segment in each cycle of the sinsuoid, so the peak-to-peak amplitude becomes exceedingly small. As the sampling rate is increased from 25 Sa/s to 50 Sa/s in figure 1c, the impact of aliasing is reduced and the data shows the same frequency as the input signal. However, the signal is still not fully captured as evident by the peak-to-peak amplitude.

As the sampling rate passes the 50Hz mark, the signal is fully replicated by the discrete data. Here, the use of a sinc function in place of first-order reconstruction would improve the results of reconstruction. Finally, figure 1e is resolute enough that even a first-order reconstruction creates a pattern that is visually similar to the input signal.

4.2 Part 2: Data Acquisition with Oscilloscope

The power spectrum highlights the three dominant frequencies of the input signal. These were 10, 50, and 70 Hz, and represent the core frequencies in the composite signal that

Dominant Frequency (Hz)	Power Spectrum Amplitude (V_{RMS}^2)
10	1.13708
50	1.13726
70	1.13377

Table 1: Dominant frequencies of noisy signal based on the obtained power spectrum.

was sampled. As evident from the amplitude magnitudes in table 1, the three components contributed roughly equally to the power composition of the signal. The plot also suggests minimal DC-bias (0-frequency content), which is desirable.

All the noise in the signal was represented in the power spectrum as roughly zero amplitude at all other frequencies up to 1000 Hz in figure 3. These frequencies were cropped from the graph to improve clarity.

5 Sample Calculations

5.1 Part 1: Number of Samples

To find the number of samples necessary to capture 500 ms of a signal, we multiply the frequency by 0.5 s, following equation 5.

$$\text{Number of samples}\,[\text{Sa}] = \text{Sampling rate}\,\left[\frac{\text{Sa}}{\text{s}}\right] * \text{Desired time}\,[\text{s}] \tag{4}$$

For 10 Hz, Number of samples $N = 10\frac{\text{Sa}}{\text{s}} * 0.5\ \text{s} = 5\ \text{Sa}$. The number obtained was then rounded up if not an integer to ensure 500ms is captured. Finally, n was increased by 1 to account for the errors in the startup transient period. So for 10 Hz, $n = 10 * 0.5 + 1 = 6$.

5.2 Part 2: Power Spectrum from FFT

Taking the n-th non-zero component of the FFT to have amplitude $A_n = 1$ [V] and the number of samples to be $L = 100$ [Sa], we follow this procedure to obtain the power spectrum.

$$\begin{aligned} \text{One Sided Amplitude Spectrum Input}\,[\text{V}] &= \frac{1\,[\text{V}]}{100\,[\text{Sa}]} * 2 \\ \text{One Sided Power Spectrum Input}\,[\text{V}] &= \left(\frac{0.02\,[\text{V}]}{\sqrt{2}}\right)^2 = 1 * 10^{-4}\,\left[\text{V}_{\text{RMS}}^2\right] \end{aligned} \tag{5}$$

6 Appendix

6.1 MATLAB Code for Part 2 Post-Processing

```
%% Take a .csv file with two columns and produce a
one-sided power spectrum of the data
clear; clc; clf;

%% Input file processing
filename = 'scope_1_1.csv'; %Name the file to be imported
preInput = readtable(filename); %Import csv file using readtable
Input = table2array(preInput);
t = Input(:,1); %Store the time data
X = Input(:,2); %Store the magnitude data
T = t(50,1)-t(49,1); %Sampling period: find from data
Fs = 1/T; %Sampling frequency: calculate from sampling period
preL = size(t); %Length of signal
L = preL(1);

% FFT
Y = fft(X); %Take the FFT of the amplitude vector
P2 = abs(Y/L); %Find two-sided amplitude spectrum
P1 = P2(1:L/2+1); %Find one-sided amplitude spectrum from above
P1(2:end-1) = 2*P1(2:end-1);
P0 = zeros(L/2+1,1);
P0(1) = P1(1); %Find one-sided amplitude rms spectrum from above
P0(2:end-1) = (1./sqrt(2))*P1(2:end-1);
P = P0.^2; %Find the power spectrum
f = Fs*(0:(L/2))/L; %The frequency domain

% Plotting
%Plot the power spectrum against the frequency domain
plot(f(1,1:20),P(1:20,1),'k', 'Linewidth', 1)
title('Single-Sided Amplitude Spectrum of X(t)')
xlabel('f (Hz)')
ylabel('|P(f)|')
```