Introduction

The quantification of electrical energy can be accomplished using a digital multimeter, analog oscilloscope, or a PC based digital data acquisition system. The oscilloscope and PC are capable of displaying traces that vary with time. Inverting the period of a signal will result in the determination of frequency. However, these instruments are limited to measuring signals that are fairly simple to quantify. As the signal becomes more complex, the ability to accurately determine the frequency components of the signal decreases. A dynamic spectral analyzer can be used to analyze the frequency components of a signal which may not be distinguishable with either the oscilloscope, digital multimeter, or the PC based digital data acquisition system.

The objective of this experiment is to introduce a means of analyzing signals using a dynamic spectral analyzer. In doing so, the conversion of time to frequency domain data using Fast Fourier Transformations (FFT) will be examined. Also, errors inherent with FFT and methods used to reduce these errors will be studied.

Pre-Lab Analysis

1. A spectrum analyzer will be used to acquire frequency domain data between 0 and 4000 Hz. What is the minimum sampling rate required? At what increment of time will the signal be acquired?

2. Define the term leakage, when used to express errors associated with frequency domain digital data acquisition.

3. Explain two methods that may be used to reduce leakage.

4. What purpose does an anti-aliasing filter serve? What error may they cause?

Background

The Dactron Photon multichannel dynamic signal analyzer is capable of displaying signals in the frequency domain by transforming time domain traces using Fast Fourier Transformation (FFT). Prior to performing FFT on the signal, analog to digital conversion (ADC) must be accomplished. Nyquist sampling theory states that the sampling rate must be at least two times the frequency of the signal in order to obtain a true representation of the signals frequency.
\[ S_R > 2(f_{\text{max}}) \]

where \( S_R \) is the sampling rate and \( f_{\text{max}} \) is the highest frequency being measured. The error associated with sampling at a low rate is called aliasing. The analyzer is equipped with anti-aliasing filters that reduce errors associated with sampling high frequency signals. Although anti-aliasing filters can be beneficial, caution is required when using them.

Fast Fourier Transformation of a time domain trace is required in order to properly analyze complex signals. Figure #A is a time domain trace that is comprised of two sine waves. The frequencies of this signal cannot be easily determined.

![Figure #A Time domain trace](image)

However, if the same signal is transformed from the time domain to the frequency domain, the determination of the frequency components that comprise the signal can be made much more effectively. Figure #B is a frequency domain trace which was generated by performing a FFT on the signal shown in figure #A.

![Figure #B Frequency domain trace](image)
The transformation from the time to the frequency domain is based on the Fourier Transform. This Fourier Transform is defined as follows:

\[ S_x(f) = \int_{-\infty}^{\infty} x(t)e^{-j2\pi ft} \, dt \]

where \( X(t) \) = the time domain representation of the signal \( X \)

\( S_x(f) \) = the frequency domain representation of the signal \( j = \sqrt{-1} \)

Although the signal being analyzed is analog, which is continuous, the signal is digitized prior to performing the transformation. This digitized signal is comprised of discrete quantities; therefore an approximation of the true Fourier Transform may be obtained by performing numerical integration. The approximation of the true integral is called the Discrete Fourier Transform. Because the signal can only be evaluated at discrete points the transform becomes:

\[ S_x(m\Delta f) = \int_{-\infty}^{\infty} x(t)e^{-j2\pi m\Delta ft} \, dt \]

where \( m = 0, \pm 1, \pm 2, \ldots \)
\( \Delta f = \) the frequency spacing

Also, the computation of an integral is required. This can be done by computing the area under the curve defined by the function. The area under the curve will be obtained using the rectangular rule. The transform will now be expressed as follows:

\[ S_x(m\Delta f) \approx \Delta t \sum_{n=-\infty}^{\infty} x(n\Delta t)e^{-j2\pi m\Delta fn\Delta t} \]

where \( \Delta t = \) the time interval between samples

Finally, the signal must be sampled from \(-\infty\) to \(+\infty\). However, this is not practical and the transformation must occur over a limited amount of time. The resulting transformation is called a Discrete Fourier Transformation which can be expressed as follows:

\[ S_x'(m\Delta f) = \Delta t \sum_{n=0}^{n-1} x(n\Delta t)e^{-j2\pi m\Delta fn\Delta t} \]
Introduction to PHOTON II Analyzer

Note:
1. The text formatted in **bold** and *Italic* is an icon in the RT Pro Photon software, e.g. *(Y Axis Format)*
2. The text formatted in *(BOLD CAPTION)* is the setting up of the hardware, e.g *(SUMMING BLOCK OUTPUT)*

**Assignment #1: PHOTON II Set Up**

1. Open the *RT Pro Photon* analyzer from the programs menu or from the desktop icon. In the new project window select:

   **New Realtime Processing**
   - **Signal Analysis & Waveform source**

   to open a new project.

The Photon project Graphic User Interface is shown in Figure 1.

![RT Pro Photon GUI](image)

**Figure 1: RT Pro Photon GUI.**
Look around the GUI to get familiarized with the program. The channel status window shows the properties of the signals of different channels. The main window shows the waveform of time and frequency domains of the input signals. The tabs on the right side of the GUI can be used to setup the properties for the measurement of the input signal and for the waveform source. Figure 2 shows the measurement and waveform tabs.

2. Select **Setup → Engineering Units...**
   Select appropriate units in the Engineering Units window by following Figure 3 and click **OK**.

![Figure 2: Measurement and Waveform Source Tabs.](image)

![Figure 3: Engineering Units Window.](image)
3. Select *Setup → Measurement Request → Signal Setup...*
   Alternately, clicking on the *Signals* button in the Measurement tab links to the same window.

In the Signal Setup window select the check boxes in the two tabs as shown in Figure 4 (a) & (b) and click **OK**.

![Signal Setup window](image)

(a) *Auto Channel Signals Tab.*

(b) *Cross Channel Signals Tab.*

Figure 4: Signal Setup window

4. Right click on the frequency domain plot in the main window of the GUI and select **Contents** in the popup.

   In the Contents window (Figure 5) change the *Display Unit* to **EUpk** (Engineering Units peak). Also change the *Y Axis Format* to **Mag** (Linear Magnitude). Click
OK. The Y Axis Format can also be changed by selecting one of the icons in the toolbar on top.

Open the Contents window for the time domain plot and make sure the Y Axis Format is set to Real.

![Contents Window](image)

**Figure 5: Contents Window.**

5. Click on the green Start button in the Measurement tab to start acquiring the data from the Photon Analyzer. Notice that the analyzer is set to acquire data continuously. Since there is no signal being input into the analyzer, the displayed trace is simply low-level noise. Note the signal properties in the Channel Status window.

6. Click on the numbers on the X Axis on any one of the plots in the main window to open the Change X Limit window (Figure 6). The limits of the X axis can be changed to zoom into a particular region of the plot. Click OK.
7. Click on the button on the bottom left of the plots in main window to select or unselect the channels that are displayed.

8. Select **Cursor → Add Normal Cursor**. To add a cursor on one of the selected plots in the main window. Drag the cursor using the mouse or by using the left and right arrow keys on the keyboard. This is how to record the pertinent values.


![Image of Change X Limit window](image)

**Figure 6: Change X Limit window.**

![Image of Report Setup window](image)

**Figure 7: Report Setup window.**
10. Select **Report → Quick Report** to generate a report in word format. The ActiveX format cursors can be added to the waveform plots to record the pertinent information. This will not work if macros are not enabled in Microsoft Word.

**Assignment #2: Acquisition of a Pure Tone**

The time domain signal being analyzed is converted to the frequency domain by performing a Discrete Fourier Transformation. In order to correctly produce the true frequency, the time domain signal must be periodic in the sample window. When analyzing a pure tone such as a sine wave, the signal is periodic if the amplitude and phase are equal at the beginning and end of the sample window. If the signal is not periodic, a leakage error will occur resulting in a distortion of the frequency domain trace.

![Figure 8: Periodic and Non-periodic signal window](image)

**Procedure:**

1. Set the frequency band of measurement to 1000 Hz by selecting from pull down menu next to **Span → 1000 Hz** in the Measurement tab. Set the number of spectral **Lines** to 100.

2. Set the **Window** to **None**.

3. Select **Waveform Source → Sine** as the output signal. Set the amplitude to **1 Volt** and frequency to **150 Hz**.
4. Connect the **OUTPUT** of the Photon II analyzer to its **Channel 1** using a BNC cable.

5. Click on the green **Start Source** button to start generating a pure tone sine wave.

6. Click on the green **Start** button on the Measurement tab to start measuring the signal from Channel 1.

7. Add a Normal Cursor by clicking **Cursor → Add Normal Cursor** and record all pertinent information for post-lab data analysis.

8. Click on the **Average** button in the measurement tab to open the Average window. Select **Average → Type Linear** and set the **Average Frame Number** to 10.
9. Now change the source frequency to 155 Hz and record all the pertinent information for post-lab data analysis.

10. The leakage that is present in the displayed waveform can be reduced with the use of a window. Select Hanning from the pull down menu next to the Window icon in the measurement tab. Note the effects a window has on the leakage that occurs when performing a FFT on a non-periodic signal.

11. Now apply a Flat Top window to the signal. Plot the resulting trace. Take note of all pertinent information.

**Assignment #3: Acquisition of a Pure Tones and Random Noise**

The dynamic signal analyzer is a powerful instrument when used properly. As with all digital data acquisition, the trace displayed by the analyzer may not always be an accurate indication of the actual signal being acquired. Also, the signal displayed may be comprised of desired components as well as undesired background noise. The ability to determine a good measurement from a poor one is an essential part of making viable conclusion based on experimental results. The objective of this assignment is to acquire a signal that is comprised of a pure tone and random noise.

**Procedure:**

1. Connect the OUTPUT from the function generator to Channel #1 of the Photon analyzer.

2. Set the display to Log Magnitude in the Main Frequency Window. Adjust the amplitude tuner on the function generator to generate an 800 Hz sine wave. Use
the multi-meter to adjust the function generated to an amplitude of 5 Vpp (Volt Peak to Peak). Make sure the attenuation button on your function generator is off. Then double check the peak value using the Channel Status window. The Peak value location is framed with red in Figure 11.

3.

![Figure 11: Channel Status Window---Peak Value Monitoring](image)

4. Set the frequency Span of the analyzer to be from 0.0 Hz to 8 KHz. Change the number of Lines to 3200.

5. Set the Window on the Measurement tag to None using the dropdown menu.

6. Turn the Average function on, allowing the analyzer to complete 10 Linear Averages of the input signal. Examine the resulting trace, note that multiple frequencies are displayed.

7. Connect the Internal Source of the Phonon Analyzer and output from the function generator to INPUT #1 and INPUT #2 of the SUM BLOCK. Connect the SUMMING BLOCK OUTPUT to CHANNEL #1. (Refer to the figure 12)
8. Set the Function Generator to generate a 100 Hz sine wave with amplitude of 5 Vpp. Create a 10 Hz sine wave of amplitude 0.1 Vpp with the Photon Analyzer’s Waveform Source. Set the frequency range of the analyzer from 0.0 Hz to 1000 Hz with 200 Lines. Allow 10 Averages and plot the result.

9. Set the Function Generator to generate a 100 Hz sine wave with amplitude of 5 Vpp. Alter the source of the analyzer to Pseudo Random at the Waveform Source window. Set the level to 0.1 Volt RMS. Set the frequency Span of the analyzer to go from 0.0 Hz to 1000 Hz with 200 Lines. Allow for 10 Averages, record any pertinent information and plot the spectral trace. Turn Averaging Off and plot the resulting spectral trace. Make sure to check the Stop at Frame Number box and type in a low number such as 10.

10. Increase the random noise level to 1 Volt RMS. Record any resulting information and plot the spectral trace both with and without averaging.

**Assignment #4: Filter Characterization**

The characteristics of a filter can be determined through frequency response measurements. A random excitation can be applied to the filter as an input signal and the output response measured. These time signals can be transformed to the frequency domain and the ratio of output to input (aka, frequency response function) can be measured. The cutoff frequency can be determined from the Bode plot (dB Magnitude vs Log frequency). This assignment will identify the characteristics of a filter.
Procedure:

1. Hook up the **Output** of the **Photon Analyzer** to the **Input** of the **Filter** as well as **Channel 1** of the **Signal Analyzer**. Hook up the **Output** of the **Filter** to **Channel 2** of the **Photon Analyzer**. See Figure 13.

![Diagram of equipment configuration](image)

**Figure 13: Assignment #4 and #5 Equipment Configuration**

2. Select a bandwidth with a 2.0 KHz upper frequency; a lower frequency bandwidth may be necessary depending on the actual filter used. Set the frequency span of the analyzer by selecting the **Measurement** tab. In **Span** scroll down until **2000 KHz** is reached. Select **800 Lines**.

3. On **Measurement** tab click on **Window** and select **None**.

4. Press the **Waveform Source** tab. Select **White Noise** in **Waveform** frame. Set the amplitude to **1 Volt RMS**. Click on the **Start Source** button.

5. Make sure that the two channels signals can be observed. If not, click on the left lower corner of the window and check both signals.

6. Measure the frequency response function with 20 or more averages such that a suitable measurement is obtained (the coherence function should be used as an indication of a suitable measurement). Click on the **Measurement** tab and click on **Average**. Click on the **Settings** tab, then select **Linear** as the **Average Type**. On **Average Frame Number** type **40**. Select **Frequency** in the **Average Domain** frame. Check **Stop at frame number** and type **40**.
7. Measure the frequency response function with 20 or more averages such that a suitable measurement is obtained (the coherence function should be used as an indication of a suitable measurement). In the main menu select Window, New window and click in the left lower corner of the new window check COH2,1(F) to view the coherence.

8. Plot the dB magnitude vs. Log frequency of the function. Right click in the frequency plot then select CONTENTS. In Y AXIS FORMAT select dBMag and check Log in X AXIS TYPE frame.

9. Right click on the left bottom corner of the pane and select EXPORT TO EXCEL, ALL SIGNALS. Save the data in Excel for Future Analysis.

Assignment #5: Sinusoidal Filter Excitation

A sine wave input to the filter will have its amplitude modified depending on the frequency of the sine wave and the specific filter characteristics. The ratio of the output amplitude to the input amplitude will be determined in this portion of the lab.

Procedure

1. Using the same equipment configuration as assignment #4. Multiple sinusoidal input frequencies will be investigated. These will be distributed across the cutoff frequency of the filter.

2. Select a fixed sine wave of amplitude 1 Volt peak for all measurements to be collected. Select the Waveform Source tab. Select Sine from the waveform frame. Type the desired frequency on Frequency (Hz). Type 1 as the Amplitude in Amplitude text box. Click on Start Source.

3. In the Frequency Spectrum pane, Right Click and select Contents then in Y AXIS FORMAT select Magnitude. Then click on the left bottom corner of the pane and Uncheck G2,2(f).

4. Measure the input and output spectrum amplitudes at each of the frequencies selected. Press in the toolbars. Now record the amplitude using the cursor (You can Right Click on the cursor and select Track To Peak).

5. Click on the left bottom corner of the pane then check G2,2(f) and Uncheck G1,1(f). Record the Output Data.

6. Follow the same procedure for the remaining frequencies.
Post-Lab Analysis

1. Explain why a dynamic signal analyzer is used to make measurements. Define the term leakage as it relates to spectral analysis measurements.

2. Why is leakage present in frequency domain measurements?

3. List methods that could be used to reduce leakage. Explain the limitations of each method. Refer to plots generated during the lab exercise.

4. Which type of window will best preserve the frequency of a signal?

5. Why would the logarithmic scale of the analyzer be used?

6. If a pure tone at 800 Hz was generated with the function generator and measured with the analyzer, why did the analyzer generate multiple spectral lines in assignment #3?

7. Referring to the plots generated in assignment #3, discuss why the random noise levels were decreased after averaging.

8. Discuss how the level of noise can affect the accuracy of a spectral analysis of a signal.

9. Plot the frequency response function of Assignment #4 for the filter characteristics measured.

10. Identify the cutoff frequency.

11. Plot the ratio of the output to input power spectrum from Assignment #5. Note any differences or similarities to the plot in Assignment #4.

12. List recommendations that should be followed to properly perform a spectral analysis of a time domain signal in order to minimize measurement errors.