**PURPOSE**

The purpose of the lab is for the student to become familiar with the use of the Tektronix TDS-3054 oscilloscope with the TDS3FFT module to analyze signals in the time domain and the frequency domain. Each student will learn how to properly set up the scope FFT to measure the frequency content of a signal and should understand the advantages and limitations of FFT spectral analysis.

# Reference Sources

Couch example 2.4 and example 2-12, Tektronix user manual for the TDS3FFT module.

Tektronix TDS5054B manual, Tektronix\_TDS3012B fft module manual

Tektronix website for additional information and application notes:

http://www.tek.com/search/apachesolr\_search/TDS-3054

**PRE\_LAB:**

Be able to answer the following questions in lab. Be sure to include in your report the theoretical values you calculate to the values you measure in lab.

* What is the voltage measurement accuracy of the TDS-3054 oscilloscope? (scope manual or data sheet) +/- 2%
* Use the Fourier Transform to determine the signal rms amplitudes and frequencies you expect to see in the frequency domain. (The FFT measures in Vrms) for:
  + A 2V p-p 2.5 KHz sinewave 2Vp-p @ 2.5 kHz (.707 Vrms)
  + A 2V p-p 2.5 KHz squarewave (calculate the values to the 10th harmonic)
    - 1.2732 Vrms @ 2.5 kHz.4244 Vrms @ 7.5 kHz
    - .2546 Vrms @ 12.5 kHz
    - .1819 Vrms @ 17.5 kHz
    - .1415 Vrms @ 22.5 kHz
* What is the relationship between a Complex Fourier coefficient, cn, of a waveform and what is seen on a spectrum analyzer displaying rms volts/Hz? (hint text 2-103, 2-106, fig 2-11) n\*f is the nth harmonic of the waveform, c\_n is the magnitude in Vrms/Hz
* What is the rms amplitude of a 1 volt peak sinusoid in dBV (dB relative to 1volt)? -3dB
* Define Nyquist Sample Rate. The Nyquist sample rate is the minimum sampling frequency to successfully recreate an original waveform.

**METHOD:**

After an initial lecture concerning signals, spectrum analyzers, and the FFT, the student will:

**Sinewave in the Frequency and Time Domain.**

1. Use a signal generator to get a 2V p-p 2.5 KHz sinusoid. Observe the signal with an oscilloscope:
   1. 1ms/div and 500mV/div
   2. Acquisition mode: Sample (Acquire, menu; Sample)
   3. Horizontal Resolution 500 points (Acquire, menu; Horz res; Fast Trig) (note that there are 10,000 points for the “normal” acquisition mode)
   4. FFT on: (Math; FFT)
   5. FFT scale: 10 dBV (Math, FFT, scale; dBV)
   6. FFT window: Hanning (Math, FFT, window; Hanning)
   7. FFT horizontal scale: 2.5KHz/div

* What is the sample rate of the scope given the above settings? (Acquire, menu) 50 kSamples/s
* What is the maximum frequency that can be analyzed without aliasing?25 kHz
* Use the scope markers to determine the amplitude in dBV and the frequency of the input signal. How do these compare to the theoretical values? -3dBV, 2.5 kHz
* What is the frequency resolution of the FFT for these settings? How could you increase the resolution? Delta F = Fs/N = 50 kHz/500 = 100 Hz

1. Change the acquisition mode to average. (Acquire, menu; mode; average)

* How does the FFT display change? Return the acquisition mode to sample. Reduces the noise/distortion components

1. Increase the scope input channel vertical sensitivity in steps to 100 mV/div.

* Why does the spectrum display change? Return the vertical sensitivity to 500 mV/div. The input signal hits the gain rail of the scope (clipping), so the FFT no longer sees a sinusoid. Instead, it sees a waveform with 2 input frequencies and their harmonics

1. Increase the frequency in 1 KHz steps to 37.5 KHz and then to 62.5 KHz. Use the internal frequency counter of the oscilloscope to measure the frequency of the time domain signal.

* Why doesn’t the counter read correctly for frequencies above 25 KHz? (Hint: what is the sample rate?) The Nyquist rate of the scope is 50 kHz, so the fastest frequency possible to convert is 25 kHz without aliasing.
* show the mathematical relationship between fin, fout and fsample when fin>fnyquist f\_out = f\_in – f\_sample\*n

1. Set the signal generator to 12.5 KHz sinewave. Adjust the FFT display to center the signal on the horizontal axis. Set the frequency span for 250 Hz/div and the vertical scale for 10dB/div. Change the FFT window from Hanning to Rectangular, Hamming, and Blackman-Harris.

* Which window results in the best frequency resolution? (>dV/df) Rectangular
* Which window gives the best amplitude accuracy? All about the same
* Describe the purpose of windowing in your report. (Tektronix booklet for FFT module)

Windowing allows you to sample a more representative waveform. For many cases, we are not sampling perfect sinusoids, and the ends of our sample may not align correctly as they should with a continuous waveform. This inconsistencies introduce high-frequency inaccuracies to our FFT. Windowing allows us to smooth out these inconsistencies.

**Square wave in the Frequency and Time Domain.**

1. Change the input signal to a 2V p-p square wave.

* Record the frequencies and linear magnitudes of the 5 spectral components visible.
  + 896 mV\_rms @ 2.5 kHz
  + 300 mV\_rms @ 7.5 kHz
  + 180 mV\_rms @ 12.5 kHz
  + 130 mV\_rms @ 17.5 kHz
  + 96 mV\_rms @ 22.5 kHz
* Make a table and compare the values recorded to the theoretical values you calculate.

1. Change the input frequency to 2.4 KHz.

* What are all of the additional spectral components due to? Because 2.5 kHz is an integer divisor of the sample rate (50 kHz), the aliased harmonics fold onto the existing harmonics. 2.4 kHz is not an integer divisor of the sample rate, so the aliased harmonics do not fold onto the existing harmonics.
* How would you eliminate this effect?

**Report:**

Turn in the completed report form next week at the beginning of lab

Grading will consist of:

* Prelab question answers
* Clarity and completeness of graphs
* Answers to the lab questions. Your analysis aspects of data collected (theory vs. measured results)

**Be sure your answers to the lab questions are easy to find!**

* **Attach your pre-lab work to this report.**

**Sinewave in the Frequency and Time Domain.**

1. Use a signal generator to get a 2V p-p 2.5 KHz sinusoid. Observe the signal with an oscilloscope:

* What is the sample rate of the scope given the above settings?

**50 kS/s**

* What is the maximum frequency that can be analyzed without aliasing?

**25 kHz**

* Use the scope markers to determine the amplitude in dBV and the frequency of the input signal. How do these compare to the theoretical values?

|  |  |  |
| --- | --- | --- |
| Frequency | Calculated [Vrms] | Measured [Vrms] |
| 2.5 kHz | .707 Vrms | .707 Vrms |

* What is the frequency resolution of the FFT for these settings? How could you increase the resolution?

**ΔF = Fs/N = 50 kHz/500 = 100 Hz**

**Increase the number of samples, or decrease the sampling frequency.**

1. How does the FFT display change?

**Reduces the noise components.**

1. Why does the spectrum display change?

**The input signal hits the gain rail of the scope (clipping), so the FFT no longer sees a sinusoid. Instead, it sees a waveform with 2 input frequencies and their harmonics**

1. Why doesn’t the counter read correctly for frequencies above 25 KHz?

**The Nyquist rate of the scope is 50 kHz, so the fastest frequency possible to represent is 25 kHz without aliasing.**

* show the mathematical relationship between fin, fout and fsample when fin>fnyquist

1. Change the FFT window from Hanning to Rectangular, Hamming, and Blackman-Harris.

* Which window results in the best frequency resolution? (>dV/df)

**Rectangular**

* Which window gives the best amplitude accuracy?

**Blackman-Harris**

* Describe the purpose of windowing in your report. (Tektronix booklet for FFT module)

**Applying a window function to the source waveform record changes the waveform so that the start and stop values are close to each other, reducing FFT waveform discontinuities. This results in an FFT waveform that more accurately represents the source signal frequency components. The ‘shape’ of the window determines how well it resolves frequency or magnitude information.**

**Square wave in the Frequency and Time Domain.**

1. Record the frequencies and linear amplitudes of the 5 spectral components visible.

* Make a table and compare the values recorded to the theoretical values you calculate. Discuss the cause of any error you see.

|  |  |  |
| --- | --- | --- |
| Frequency | Calculated [Vrms] | Measured [Vrms] |
| 2.5 kHz | 1.2732 | 0.896 |
| 7.5 kHz | 0.4244 | 0.3 |
| 12.5 kHz | 0.2546 | 0.18 |
| 17.5 kHz | 0.1819 | 0.13 |
| 22.5 kHz | 0.1415 | 0.96 |

1. Change the input frequency to 2.4 KHz.

* What are all of the additional spectral components due to? Explain where they come from.

**Because 2.5 kHz is an integer divisor of the sample rate (50 kHz), the aliased harmonics fold onto the existing harmonics. 2.4 kHz is not an integer divisor of the sample rate, so the aliased harmonics do not fold onto the existing harmonics.**

* How would you eliminate this effect?

**Use a low pass filter.**