ELEC 3004/7312: Digital Linear Systems: Signals & Control!
Prac/Lab 2: An Introduction to Spectra and Filtering
Revised April 1, 2014

Pre-Lab

The Pre-laboratory exercise for this laboratory is to:
 ➢ Revise the effects of aliasing
 ➢ Look at the spectra of common signals
 ➢ Briefly look at how FFT works

Laboratory Completion & Extra Credit Points

Laboratory Completion:
Please work **together** on the lab in groups of 2-3.

No extra credit points:
This is an "express lab" and worth up to +2 extra credit points.

Experiment 2: An Introduction to Spectra and Filtering

We will be extending what you did in Lab 1b :Sampling and Reconstruction, with a view to seeing how we can analyse signals which are more complicated than a sinusoid.
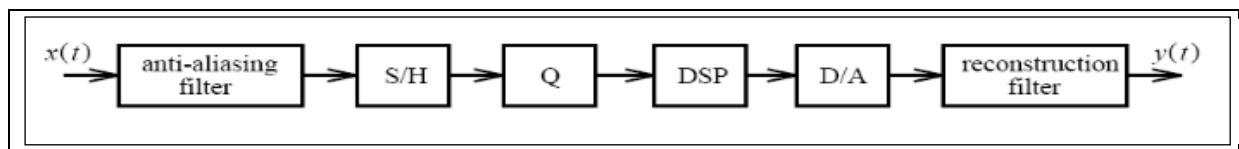

Figure 1. A block diagram of a practical DSP system

We will start by studying the spectra of unfiltered signals such as square, triangle and random noise. Then we will look at how the filters in the DSP system affect what each section does, as well as look at how aliasing can be visualised. In Lab 1b you were asked to find the native sampling frequency of the system by increasing the input frequency until your output frequency went to nearly DC. Some of you may not have found the correct one, but we'll show how you can do it with the spectrum analyser and wave generator on your oscilloscope.
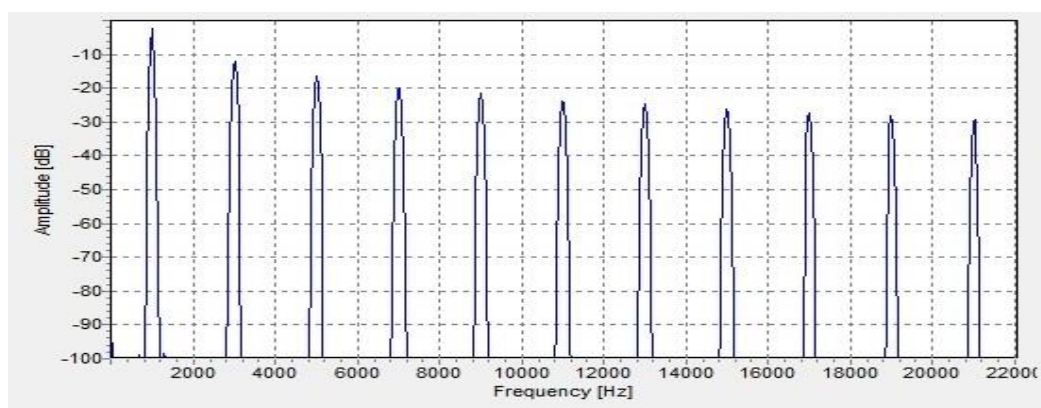

Figure 2

http://en.wikipedia.org/wiki/File:Spectrum_square_oscillation.jpg

Equipment
1. Arduino Due 32-bit development board + USB cable/s
2. PMOD-CON4 RCA board/s (optional)
3. Oscilloscope with function generator
4. 2 x RCA male to BNC male cable approx 0.5 - 1 m
5. 1 x E36 DAC Filter board and 1 x E36 ADC Filter board (UQ designed)
6. RCA-3.5mm adaptors + powered speakers + 3.5mm stereo to 2 RCA male Y-cable
7. BNC male to BNC male approx. 0.5-1m, RCA-BNC adaptor, and T-piece F-M-F

Preparation
Read the following documents to get an idea of the system we will implement:
1. Spectrum of Square Wave
   http://en.wikipedia.org/wiki/Square_wave

2. Spectrum of Triangle Wave
   http://en.wikipedia.org/wiki/Triangle_wave

Figure 4 Arduino Due

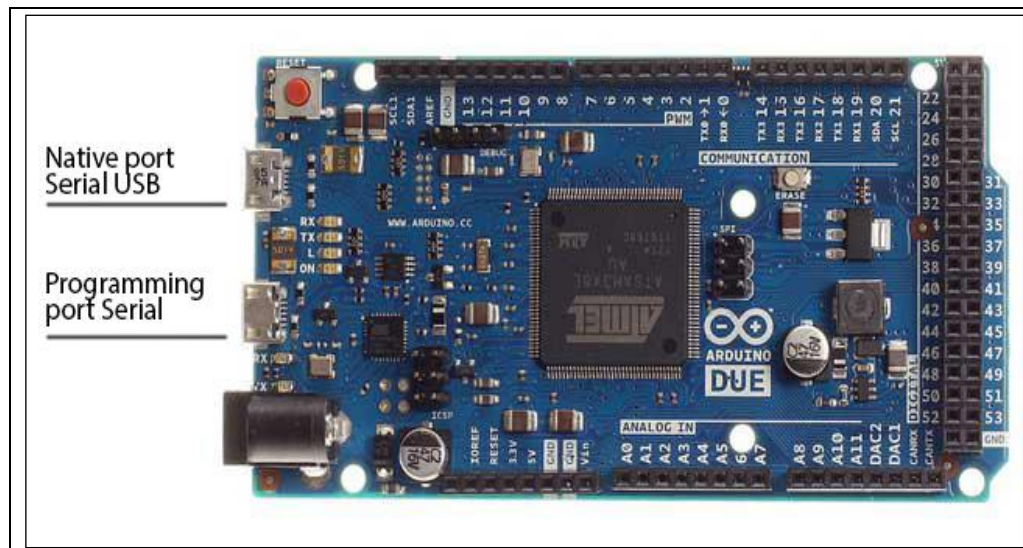*Figure 3 DAC Filter board*
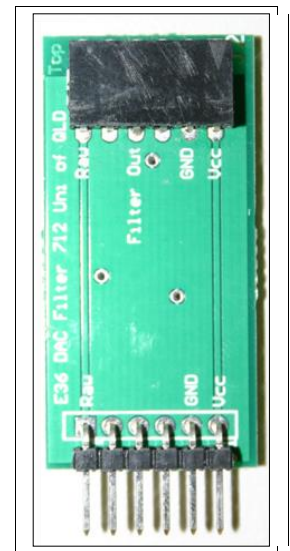

Figure 5   PMODCON4
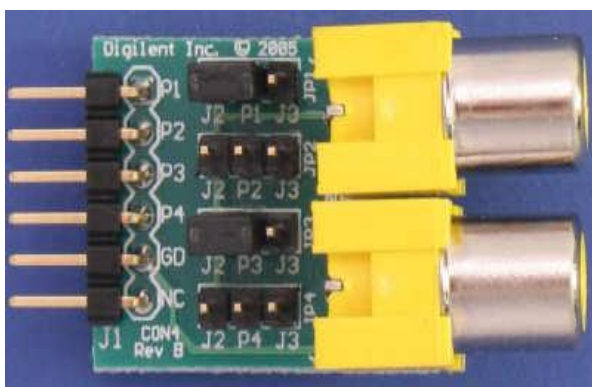

Figure 6   USB MicroB cable

Procedure:

<div style="background:black;color:white;text-align:center">Lab 2a: Looking at spectrum of a sine wave</div>

Most of your connections will be similar to what you did in Lab 1.

1. Start the Arduino 1.5.6-r2 IDE using the desktop icon, or navigate to where it exists in Program Files.

2. Open the ELEC3004_2014_Lab1b file

3. Fit both Filter boards and PMODCON4 boards

4. Connect the USB to Micro-B USB cable between the PC and the Due on the Programming Port as shown in Figure 4. Windows may give you a message about needing to load a driver. If so let this complete.

5. Setup the software to communicate with the Arduino Due in the same manner as you did for Lab 1

6. Download the software to the Arduino board

7. Attach a BNC male-male cable with the BNC t-piece to the output of the Wave Gen, so that you have the output of the Wave Gen going to CH 1 of the scope, as well as into the ADC Filter board via the PMODCON4 and a BNC to RCA cable

8. Attach another BNC to RCA cable to the RAW output of the DAC board via a PMODCON4

9. Setup the Wave Gen to 1kHz sinewave, 3Vpp, 1.5 V offset

10. Press the Math button on the scope and change the mode to FFT

11. Adjust the settings so that the Vertical units are Volts RMS, not decibels

12. Select CH 1 as your input source for the FFT

13. Make other adjustments until you see something like the image below. Note the information under Acquisition.
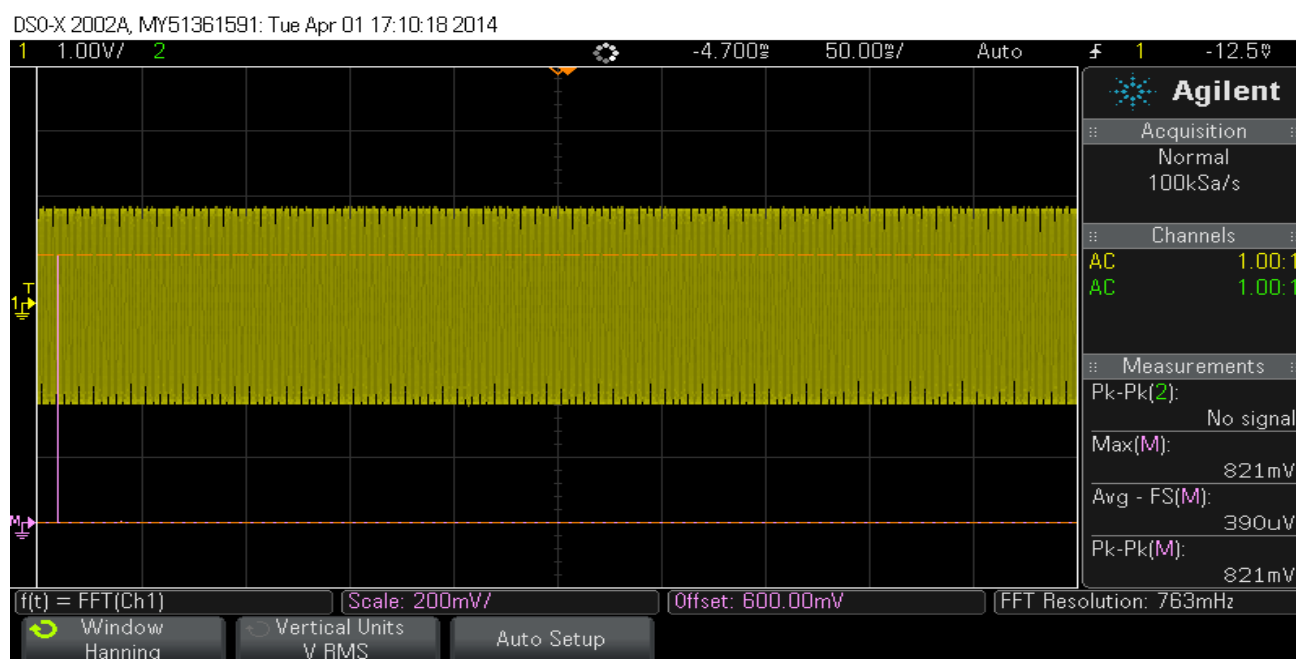

Figure 7: Spectrum of 1kHz sinewave

What you should now see in Figure 7 is a single line representing the spectrum of a sinewave with no harmonics.

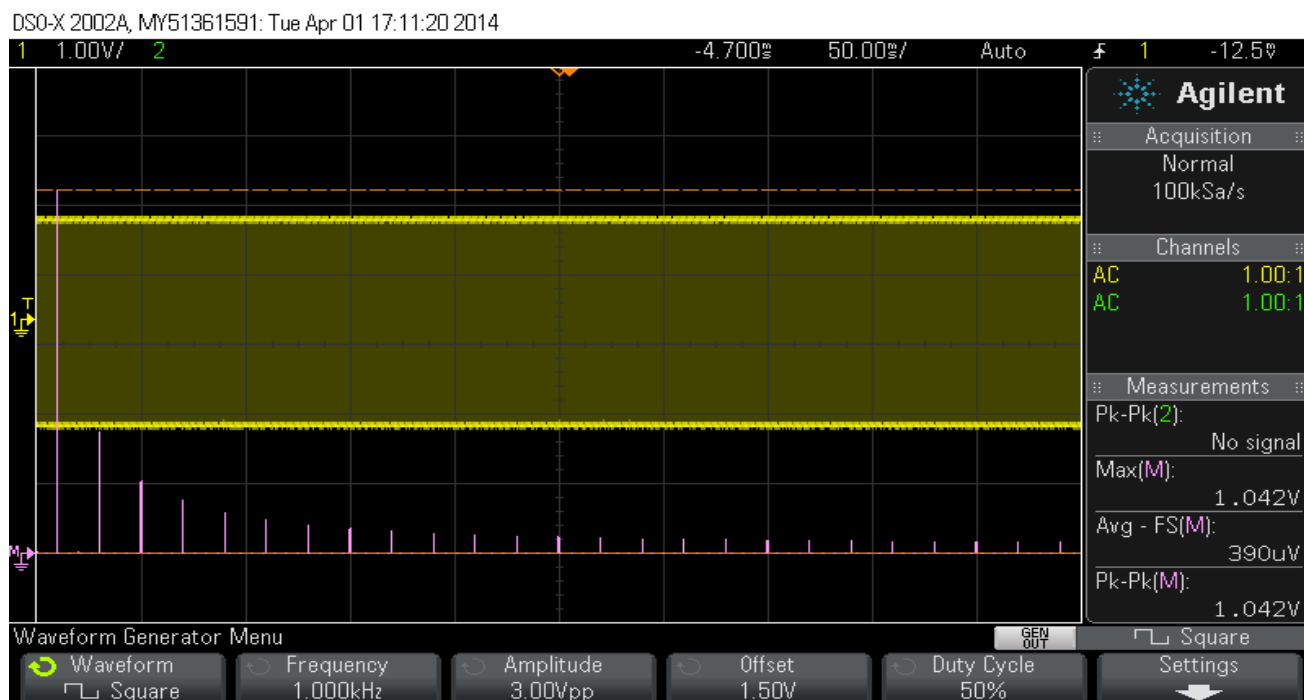If you change the settings to a squarewave you should see the following:

Figure 8

The spectrum for a squarewave consists of the fundamental at a relative amplitude value of 1, followed by 3 times the frequency at 1/3rd that fundamental amplitude, then 5 times the frequency at 1/5th the fundamental amplitude, then 7 at 1/7th etc until we are just measuring noise.

The fundamental is at approximately 1.04 volts. If we move the cursor down to the next harmonic (3 times the frequency), we should measure approximately 340mV, as you will see below.
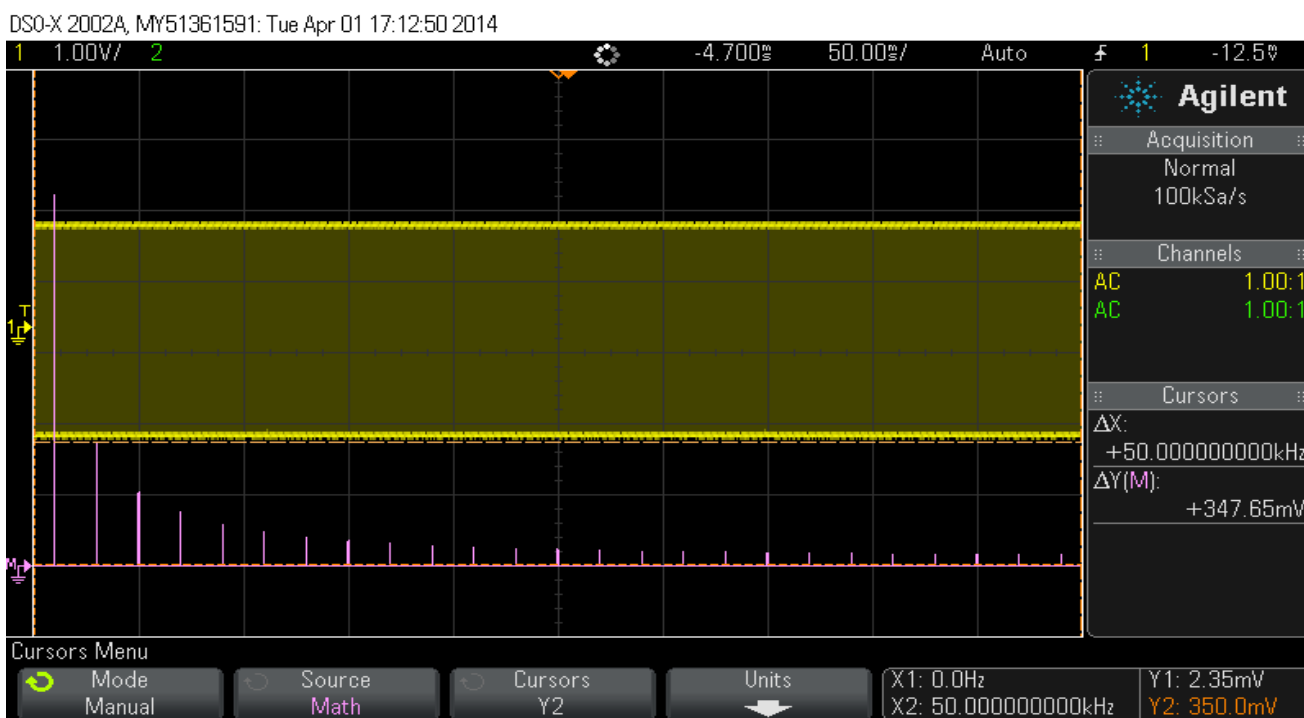

Figure 9

We can now repeat for 5 times the frequency, so we should see approximately 200mV.
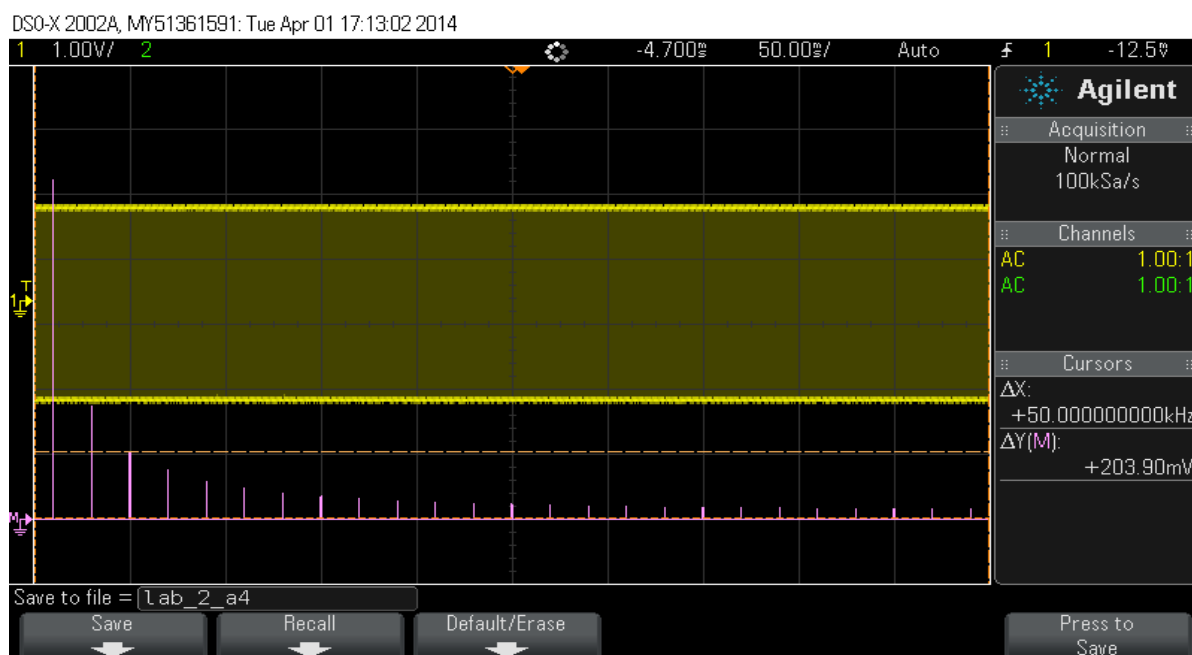

Figure 10

You could repeat this for all the harmonics to satisfy your curiosity. If you change the timebase with the **Horizontal** knob, you can zoom in a bit.
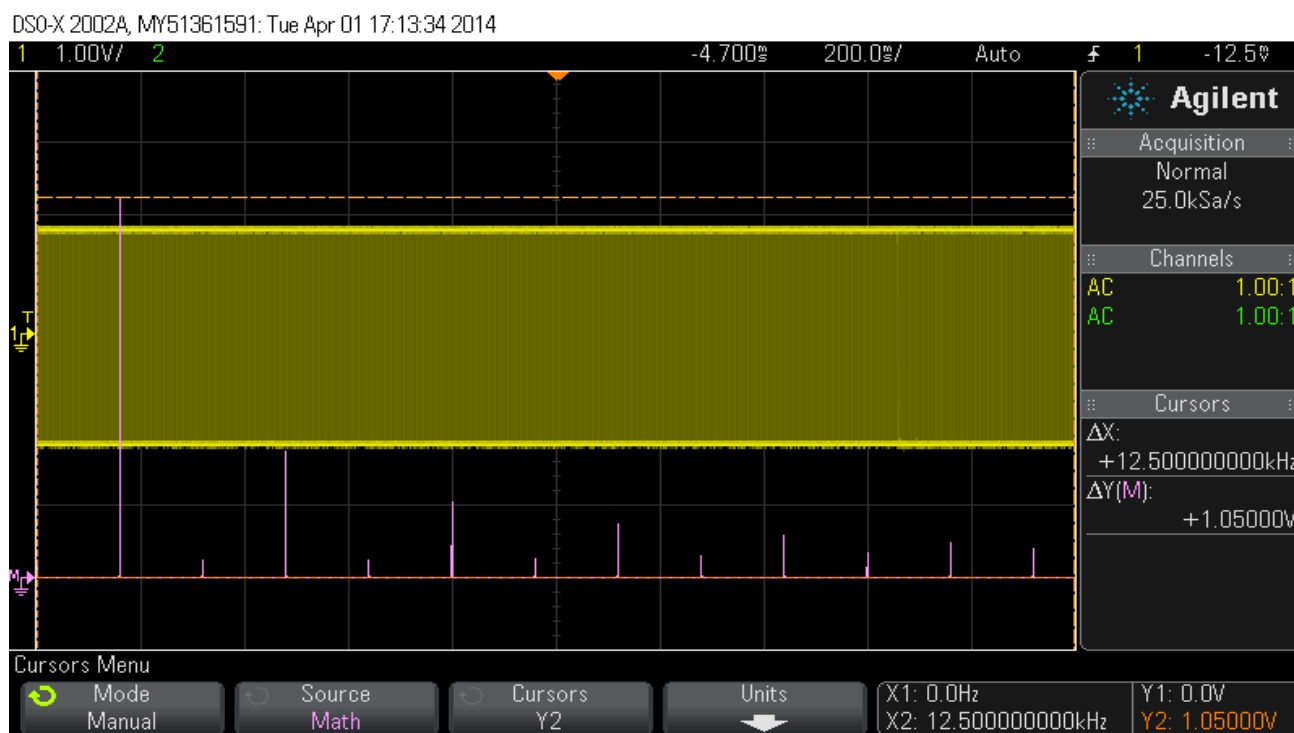

Figure 11

Finding the sample frequency:

We can now look at the output of the sampling process. If you change your FFT input to CH 2, you can see the spectrum of your sampled signal. You may recall that the output of the sampler will consist of the input signal plus repeating combinations of the equations (F1+F2) and (F1-F2), where F1 is the sampling Frequency Fs, and F2 is the input frequency. This is why a filtered sampler output would approach DC when F2 is very close to F1. You can use Raw and Filtered inputs as well as Raw and Filtered outputs.

Figure 12

In this image above, we have chosen CH2, the sampler output via the Raw output of the DAC Filter board. The span of the screen is 250kHz horizontally, so we have 25kHz per division. The input frequency has been increased to 10 kHz and is now back to a sinewave. The two lines you see just to the right of centre are F1+F2 and F1-F2.

Count the number of divisions to the centre of these two lines and you will find the sampling frequency F1, also known as Fs of course. What frequency is it? You will see that most of the power is in the input signal. Don't worry about trying to find an exact relationship between your input voltage and the displayed absolute spectra voltages. What matters is the relative voltages at related frequencies

Filtered output:

If you now move the CH 2 cable to the filtered output of the DAC Filter, you will see the higher frequencies disappear.
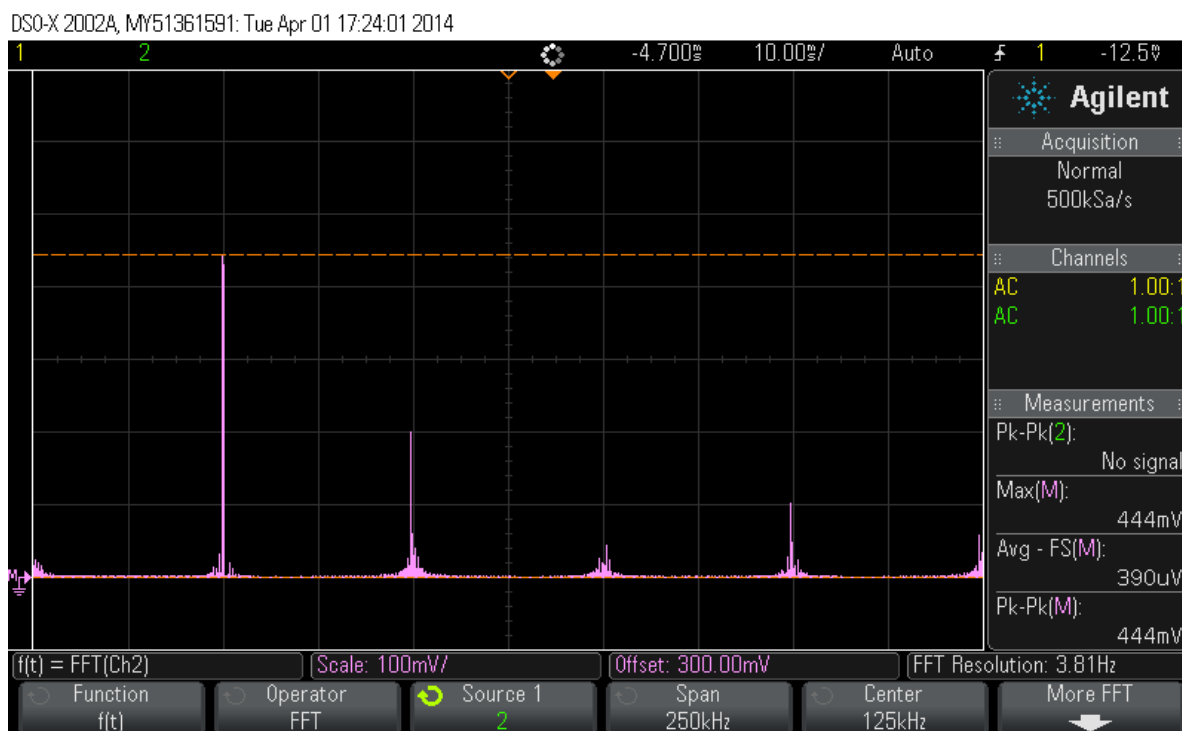

Figure 13 Filtered sampler output

Figure 14 Freq in = 50kHz

If we now increase the input frequency to 50kHz, sinewave, you'll see the display above. In Figure 12, the voltage for the input signal was 528mV, whereas here it is 444mV due to more of the total signal being spread across the spectrum.
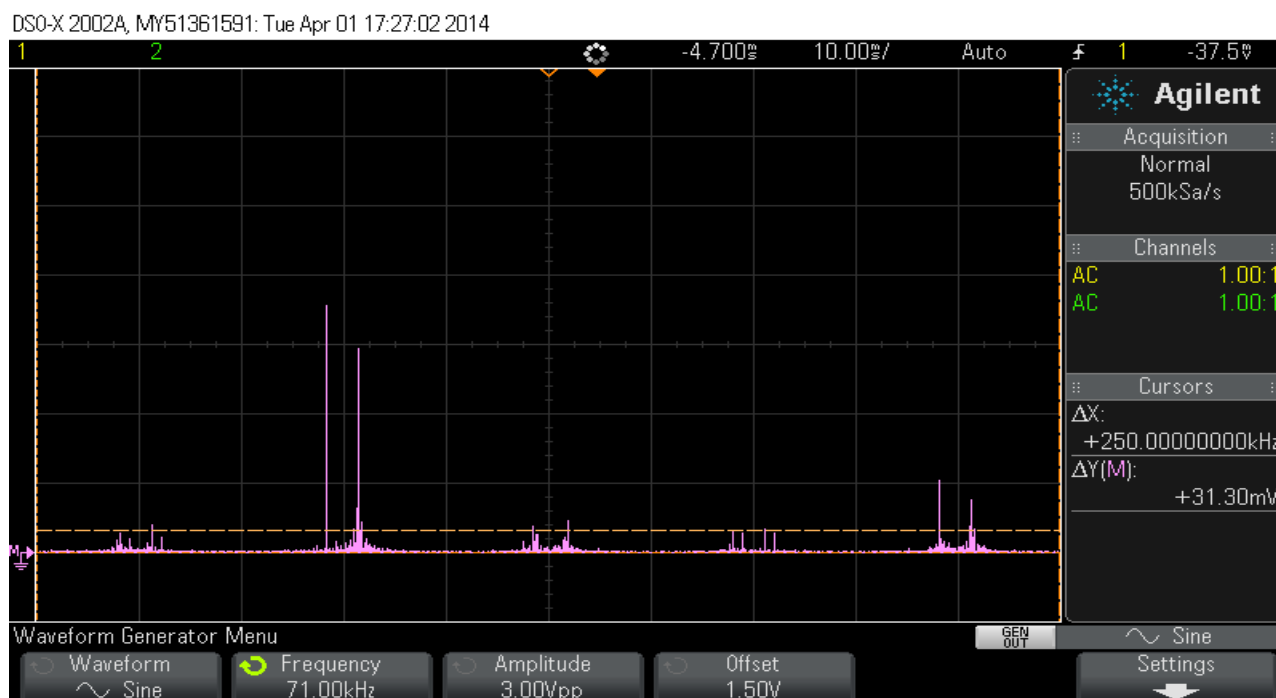

Figure 15: Input frequency raised to 71kHz

Now the input frequency is nearly at the Nyquist frequency, so F1-F2 is almost equal to F2. Note also the results at F1+F2.

Figure 16: F-in almost equal to Nyquist

Here we have hit the Nyquist frequency, so F1-F2 overlaps F2.

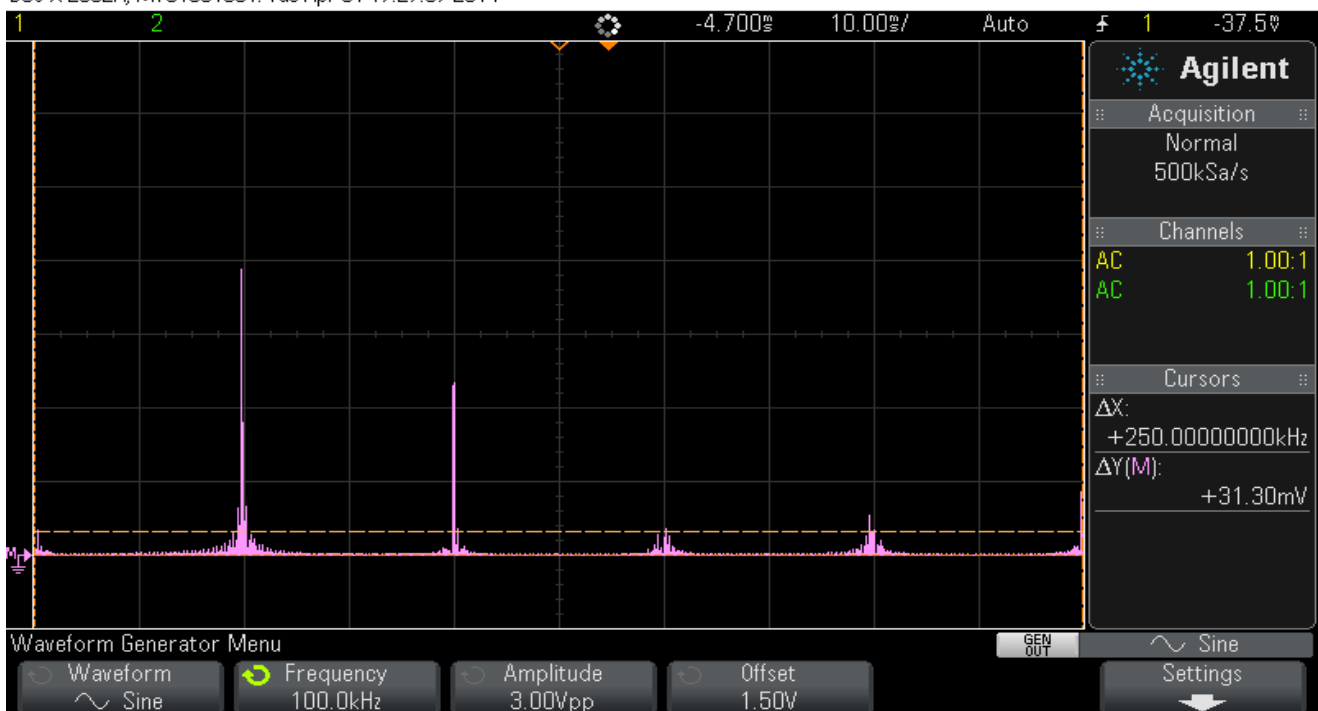**Figure 17: Freq in = 100 kHz**

Now, F2 = 100kHz, so we have passed the Nyquist frequency. You should try this as well and keep increasing the frequency. You'll observe the input line F2 going the right, and F1-F2 moving to the left

Effects of filters:

Output filter

Basically we swept the input frequency from about 1 kHz to beyond the sampling frequency. If you adjust the delayMicroseconds() code you can change the sampling rate and observe similar effects but with a shift in the F1 point.

Repeat this process for the filtered output and see what happens to the spectral lines.

Input Filter:

Compare the spectrum of filtered input for sinewave, square, triangle and noise, by changing the code to sample A3 instead of A5. For example, what spectra do you expect if you sample a filtered 4kHz sinewave? Now, how about a 4kHz squarewave? Try each signal and observe filtered in, raw out, then filtered in AND out.

Effect of output filter on a sampler Raw output with input frequency above the filter cutoff frequency.
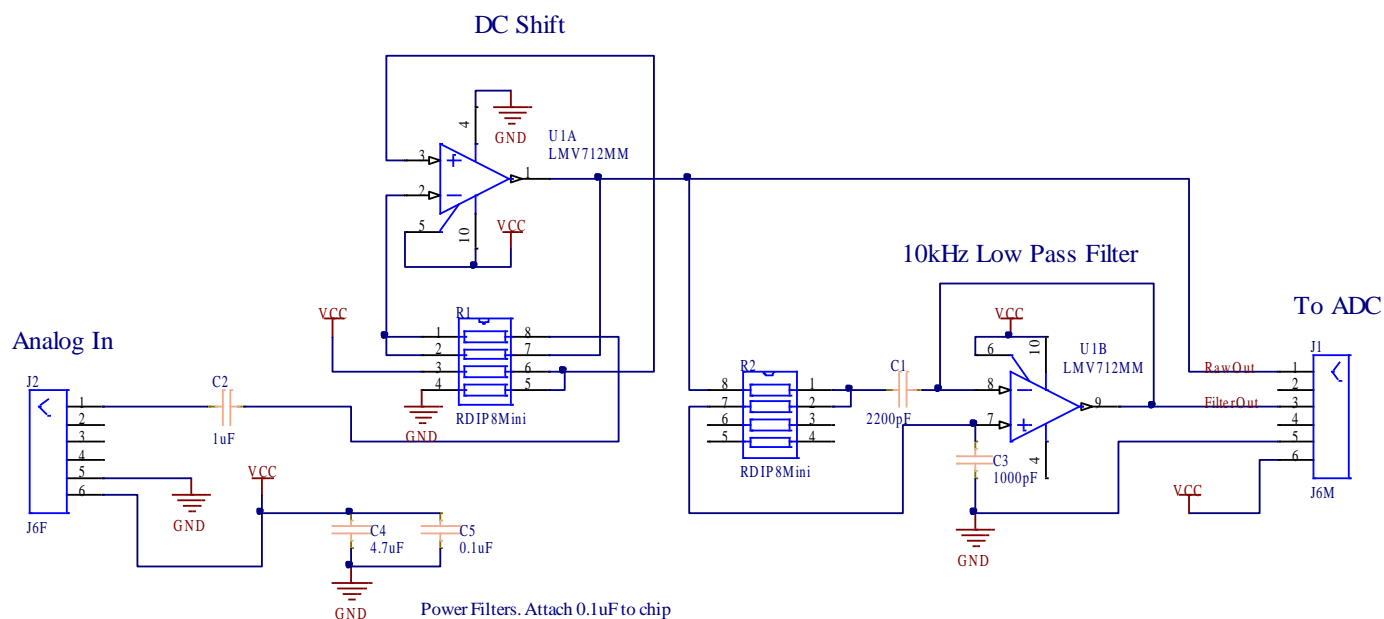
If you use the Raw input and Filtered output, what do you observe as you sweep the input sinewave from 1 kHz to about 500 kHz?
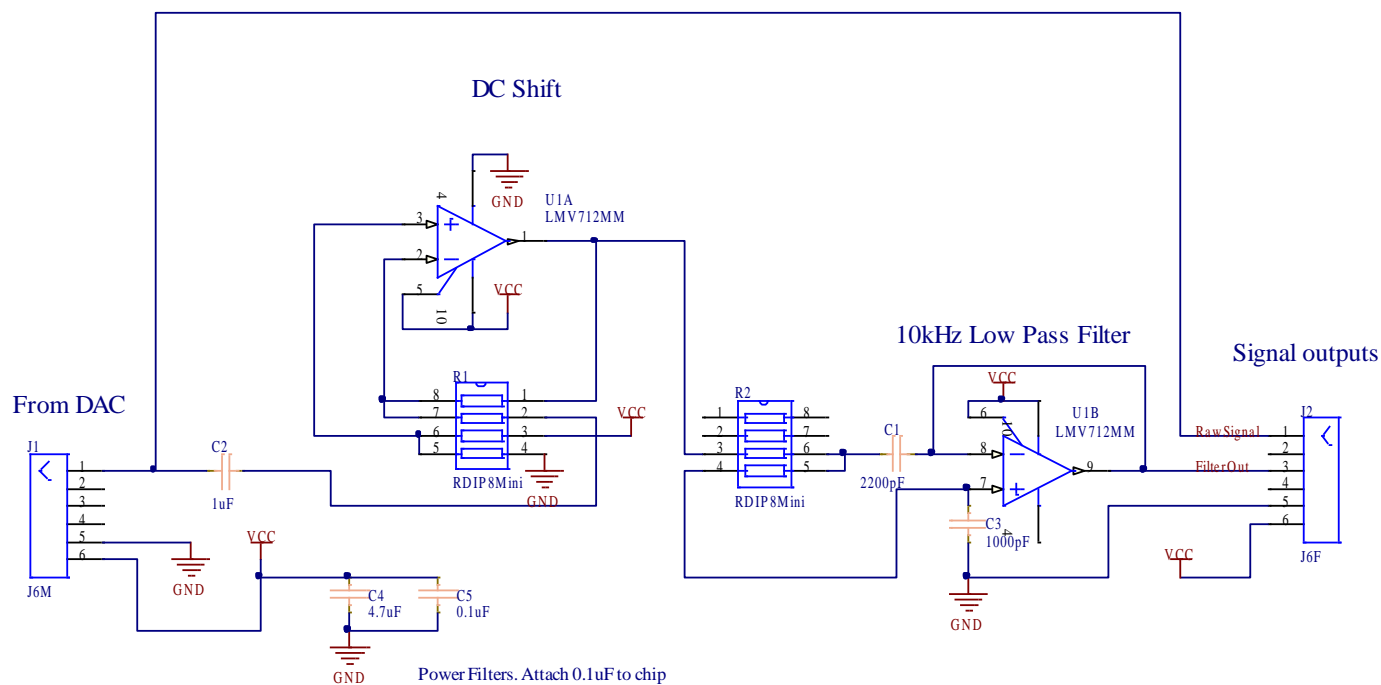
END OF LAB2

Appendix:

Schematic diagrams for filter boards:

Resistors are 10k.

Figure 18 ADC lowpass filter

Figure 19 DAC lowpass filter

Experiment II:
An Introduction to Spectra and Filtering

Hand-In Sheet

Name: _____

Student ID: _____

Date: _____

Group Name/Members: _____

_____

_____

| **Part Completed**	✔
	(Tutor)
Laboratory Part 1 (+1 point): Consider a 10 kHz analogue square wave that is sampled at 100 kSamples/second and then reconstructed back to analogue. Will the reconstructed signal be the same as the original signal? Using your knowledge of Fourier effects and sampling, please explain why or why not.	

Laboratory Part 2 (+1 point):
Based on this lab, what type of wave form function (e.g., sine, square, triangle, etc., etc.) might be best to transmit a 10kHz digital signal down a long transmission line.

Recall that the ideal transmission line model or telegraph equations (artwork from Wikipedia):
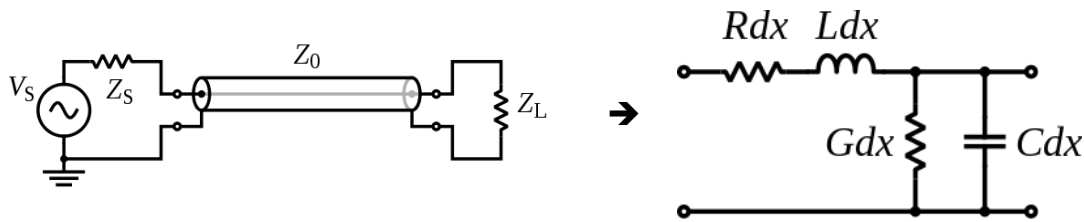
Giving: $\dfrac{d^2V(x)}{dx^2} + \omega^2 LC \cdot V(x) = 0$ and $\dfrac{d^2I(x)}{dx^2} + \omega^2 LC \cdot I(x) = 0$

Total Extra Credit Awarded: _____

Tutor Sign-Off: _____