Lab 24a Sampling an Impulse Response and Determining the Digital Compensation Filter

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EE145M   Station 8

Aim

To sample the impulse response of a single-pole low-pass filter, and to compute the corresponding digital filter parameters.
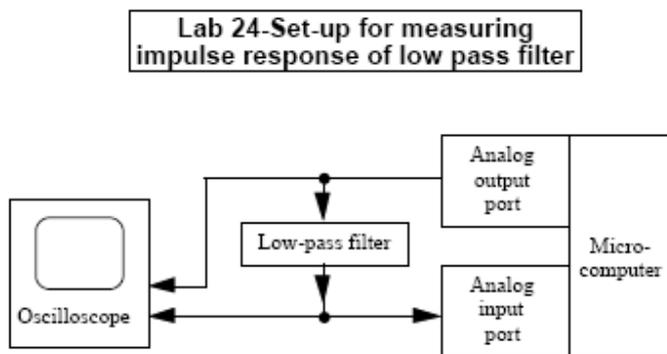
1. Setup

Figure 1 General lab setup for determining the impulse response of a low-pass analog filter. This set-up sends a computer-generated impulse through a low pass filter and records the output waveform.
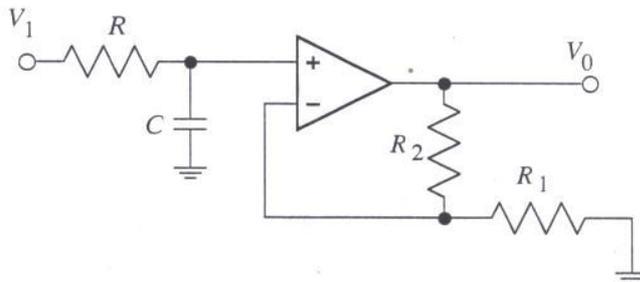

Figure 2 The setup for a single-pole low pass analog filter. The gain was picked as 100, so R=51k, C=0.1uF, R1=1k-ohm, and R2=100k-ohm. The RC exponential decay time is 5.1ms.
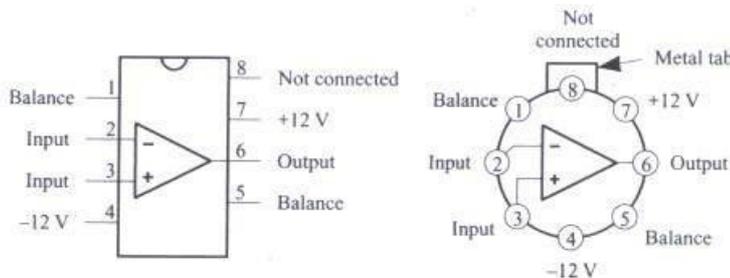

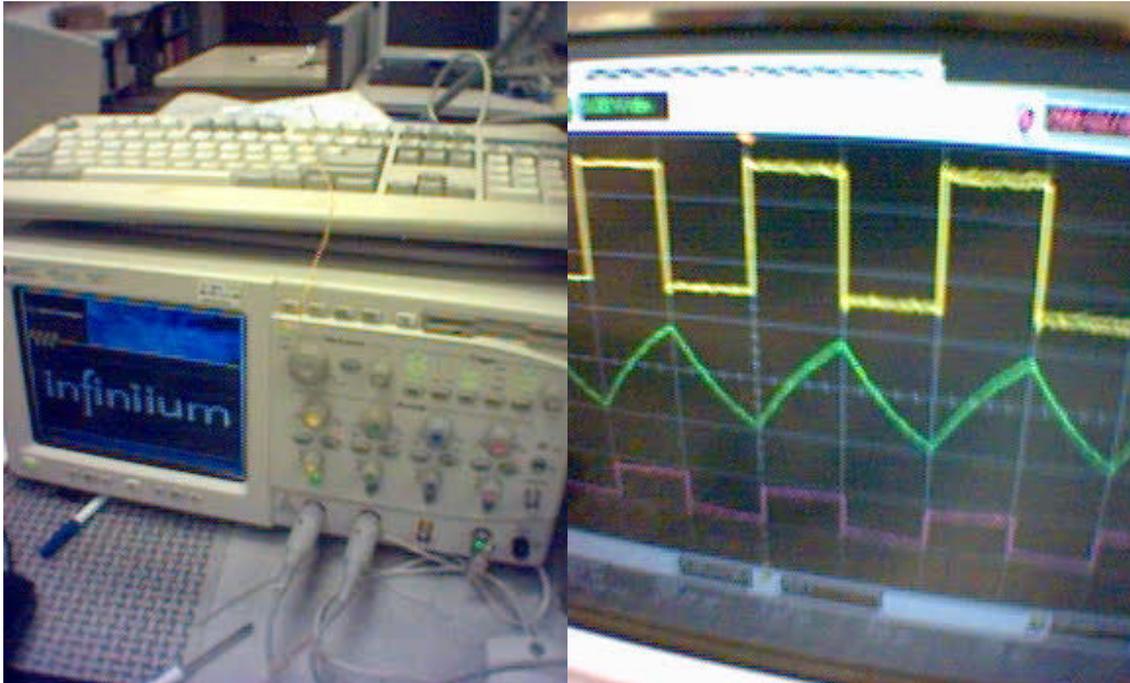Figure 3 Datasheet of the Op-Amp chip LM356 used for the low pass filter.

Figure 4 (left)The oscilloscope used for lab 24.
Figure 5 (right) A square wave input signal on top, and the output through the analog low pass filter and the FPGA compensation filter at the bottom. The output signal resembles the input square wave.

2. Data Summary
2.1 From the plots of the impulse response and determine the RC time constant of the filter.
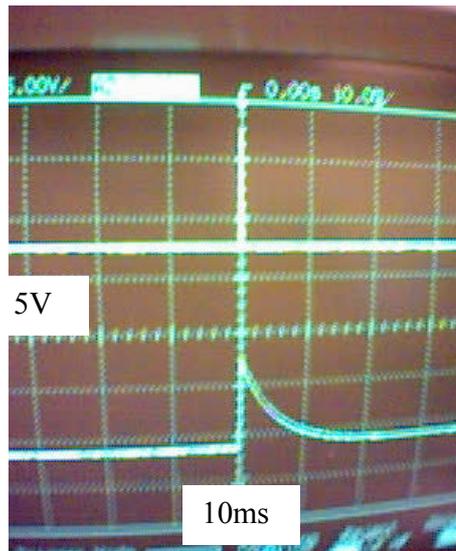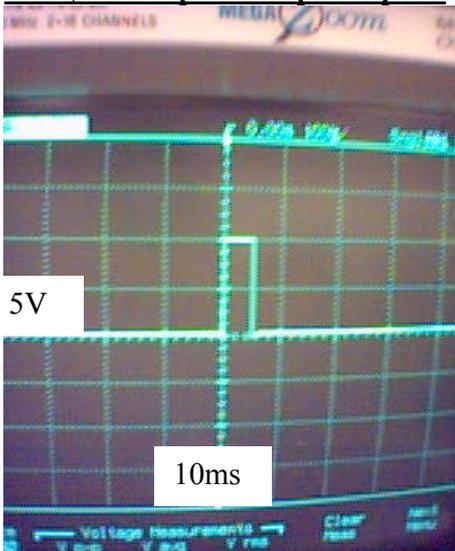
Fisrt, The impulse response plots

Figure 6(left) Enlarged Plot of an Impulse Signal
Figure 7 (right) Impulse Response of the Low Pass Filter

By feeding an impulse signal to the analog filter, the impulse response is obtained. The voltage levels versus time data are stored into the computer via the FPGA board. The data is plotted in Figure 8.
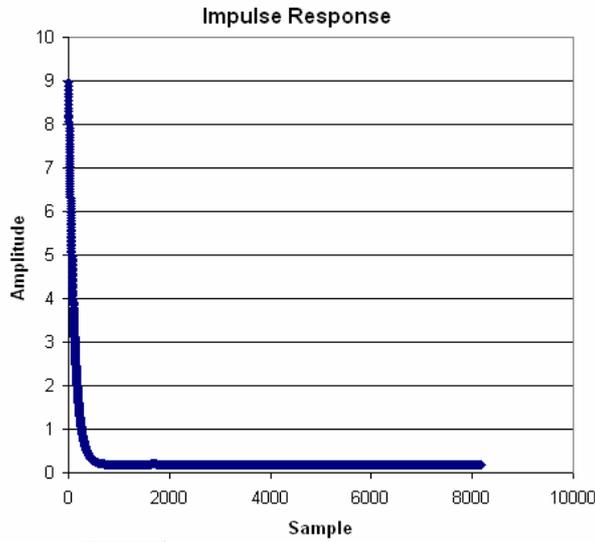

Figure 8 Raw Data for the Impulse Response of an Analog Filter

The data in Figure 8 is plotted in arbitrary units. In order to compare with the ideal impulse response, the vertical axis is converted to the unit of "voltage" and the horizontal axis is converted to the unit of "second".
Upon visual inspection, the vertical grid unit of the oscilloscope plot is 5V, and the horizontal grid unit is 10ms. Since the program has a default setting of 2000 ticks per sample, which gives a 2000*25ns = 50us per sample.
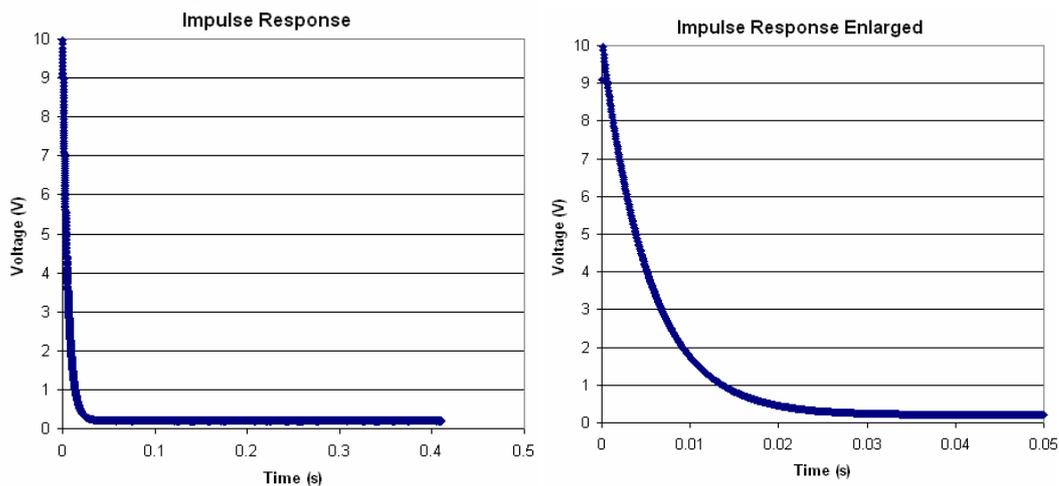

Figure 9 (left) Measured data in Time Domain for the Impulse Response of an Analog Filter.
Figure 10 (right) Measured data dnlarged in Time Domain for the Impulse Response of an Analog Filter.

Second, determine the RC constant

$$V(t) = \frac{\Delta V \Delta T}{RC} e^{-t/RC}$$ [Equation 1]

From Equation 1, V(t) = V(0)*e^(-t/RC)
When t=RC, V(t) = V(0)*e^-1 = 0.368*V(0)

From Figure 9, when the voltage V(t) equals to 0.368*V(0), time (t) equals to 5.57ms. Therefore, RC = 6.65ms

2.2 From the impulse sent to the low pass filter and the expected filter response (see Background section 1), compute the expected amplitude and shape of the low pass filter impulse response.
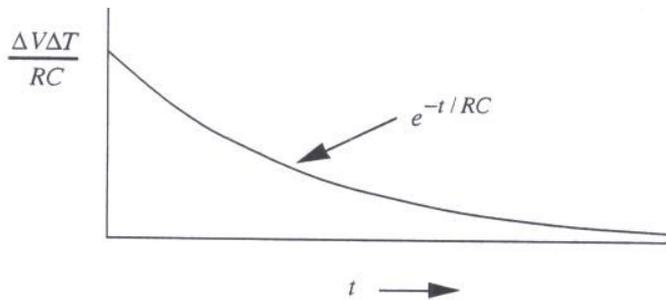
Figure 11 Ideal Expected filter response

In order to characterize the impulse response, the relative amplitude of the impulse response is arbitrary. The shape of the impulse response is more important because the amplitude can simply scale arbituarily. For the expected impulse response, the V(0) is chosen to be 9.88V based on the visual inspection of the oscilloscope plot.

Substituting into the equation,
V(t) = V(0)*e^(-t/RC)
V(t) = 9.98V*e^(-t/5.57ms) Equation 2

Figure 12 (left) Expected filter response from Equation 2.
Figure 13 (right) Expected filter response from Equation 2 and measured filter response.
         The blue line is the expected data and the red line is the measured data

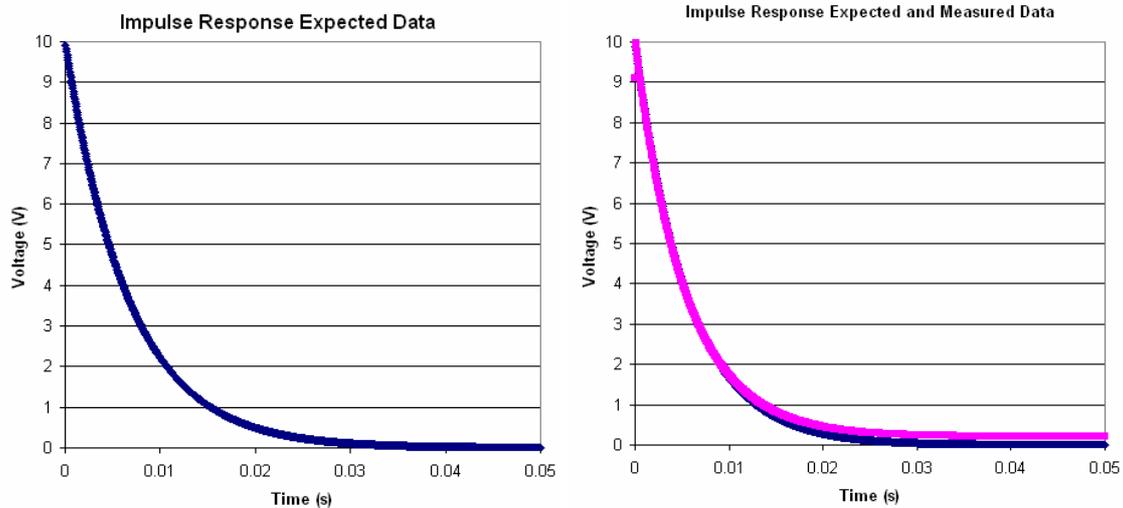Comparing to the expected data (blue), the measured data (red) is almost exactly the same except when the data is below 1V. The measured data has higher voltage value than that of the expected value in that case. In other words, the actual output voltage of the low-pass filter has a higher value than the expected output voltage.

In general, since amplitude of the measured data is of arbitrary unit, the amplitudes of the two sets of data are made the same. The shape of the two set of data are very similar expect at the tail of the signal. This might be because the analog filter is not ideal, and does not cut off the low voltages caused by noise. Therefore, there is a low voltage remained at the output of the low pass filter when the expected voltage value should be lower.

3. Discussion
3.1 Compare the observed low pass filter output from procedure section 2 with the expected RC filter response. Discuss agreement and reasons for any observed differences.
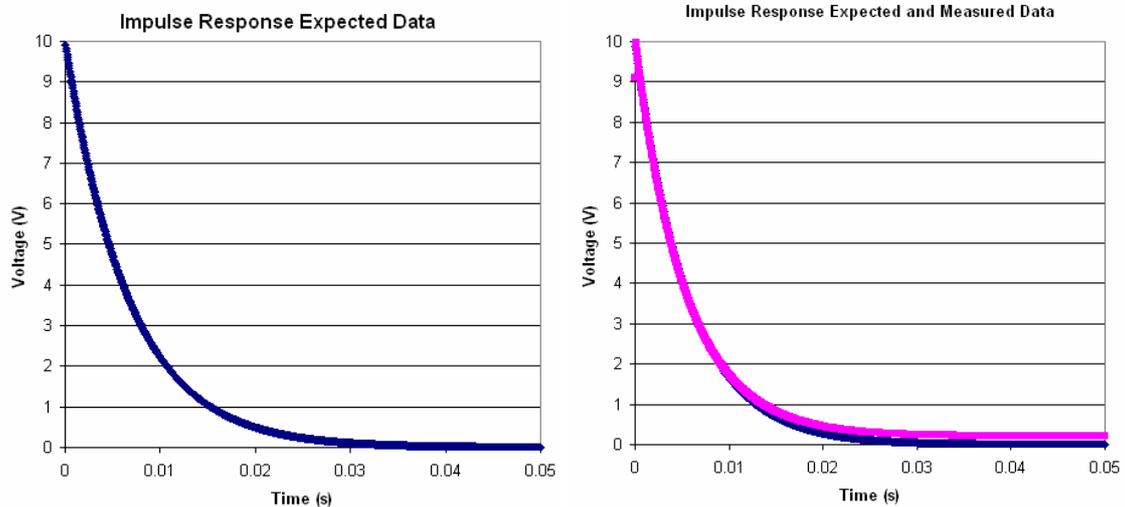

Figure 14 (left) Expected low pass filter output.
Figure 15 (right) Expected and measured low pass filter output.

As indicated in the two figures above, the signals are almost the same. In other words, the measured low pass filter output resembles the equation

$$V(t) = 9.98V*e^{\wedge}(-t/5.57ms)$$

The explanation of the differences are stated in Section 2.2.

3.2 From your knowledge (in Laboratory Exercise 23) of the digital low-pass filter that corresponds to the analog low-pass filter of Laboratory Figure 24a.1, show that the impulse response of the combined low-pass filter and the compensation filter is an impulse.

$$\tilde{a}(t) = \Big[a(t) \bullet b(t) \Big] \bullet c(t)$$

The goal of this lab is to construct a'(t) by the computer so the result is similar to a(t)*b(t)*c(t), where a(t) is the input signal, b(t) is the impulse response of the compensation filter, c(t) is the impulse response of the low pass filter, and * is the convolution operator.

By Fourier convolution theorem,

$$F(a) = F(a) \times F(b) \times F(c)$$

So

$$F(b) = 1/F(c)$$

In other words, F(b)*F(c) =1. Therefore, if the input is an impulse, and the combined filter consists of the low-pass filter and the compensation filter, then the output of the combined filter would be (combined filter) convolve (impulse) = impulse. This is why the impulse response of an impulse is an impulse, because the combined filter (the low pass filter and the compensation filter) "passes" the input to the output without modification.

4. Question
4.1 What were the measured amplitude and decay time of your impulse response? What were the expected amplitude and decay time of your impulse response? How well did they agree?

The measured amplitude is 9.98V, and the decay time is 5.57ms. The decay time is the RC constant calculated in Section 2.1. The expected amplitude of the filter is 10V.

The measure amplitude is 9.98V and the ideal amplitude is 10V. The error is 0.2%. The measured decay time is 5.57ms, and the ideal decay time should be 5.1ms. The error is 9.21%.

4.2 What problems would arise in this laboratory exercise if the Butterworth low-pass filter had order n=16?

When the order (n) is 16 instead of 1, the output signal of low pass filter will decay faster. This results in a sharper cutoff frequency. The equation for the 16th order filter will be more complicated.

When the first order low pass filter is used, there are only 2 coefficients needed from the Fourier transformation to characterize the compensation filter. If the 16th order filter is used, then for lab 24b, the compensation filter will need more coefficients. The computer needs to calculate more FFT and do more multiplication. This increases the process time of the digital filter.
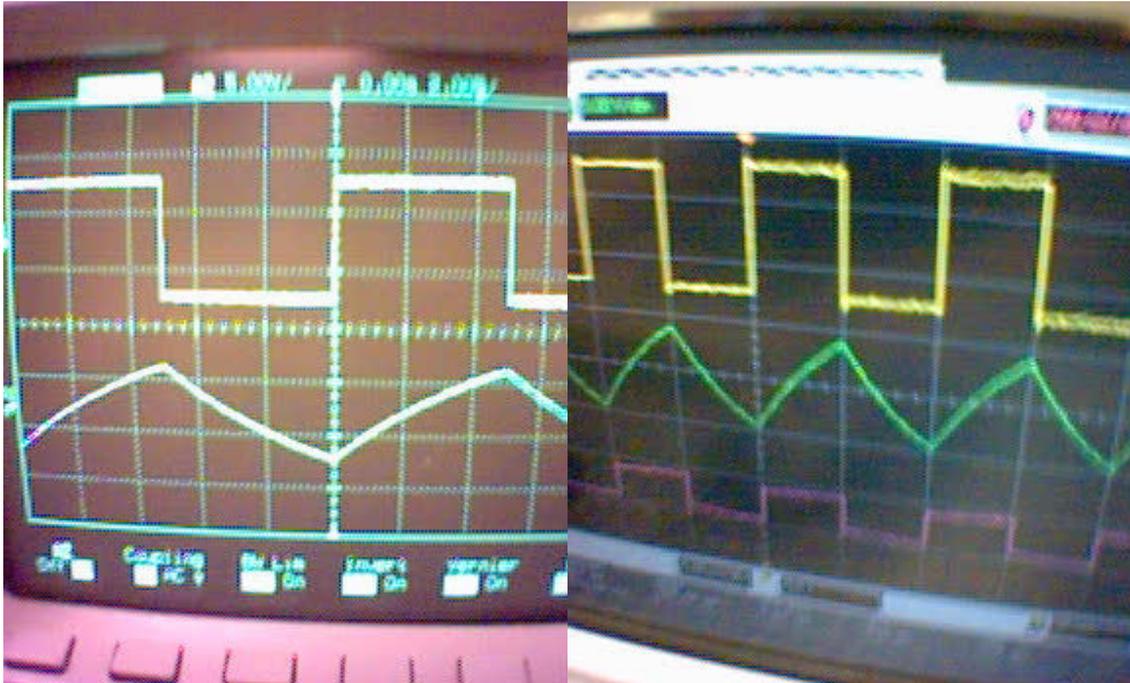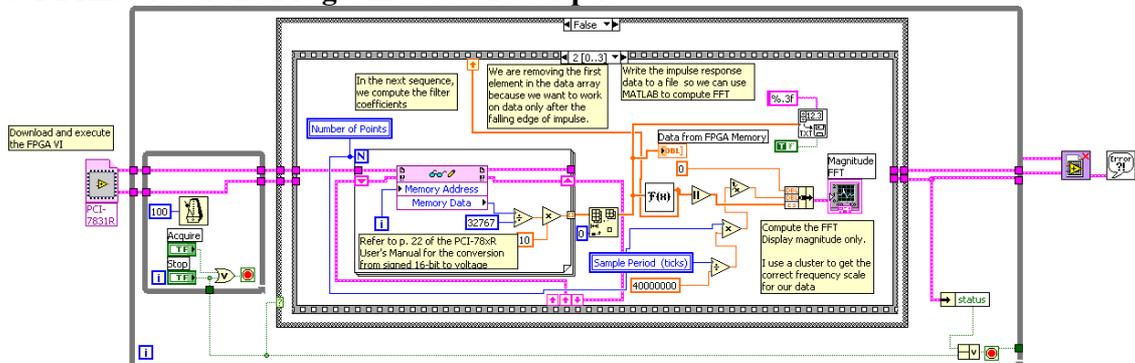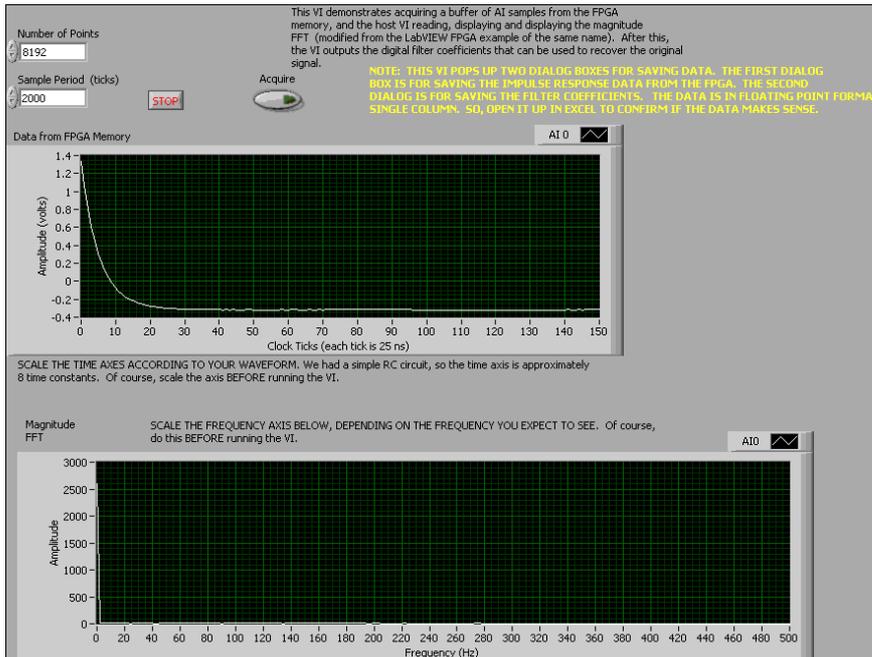
5. Program and Laboratory Data Sheets

Figure 16 (left) the input square wave on top and the output through a low pass filter at the bottom.

Figure 17 (right) the input square wave on top and the output through a low pass filter in the middle and the output through the analog low pass filter and the FPGA compensation filter at the bottom.
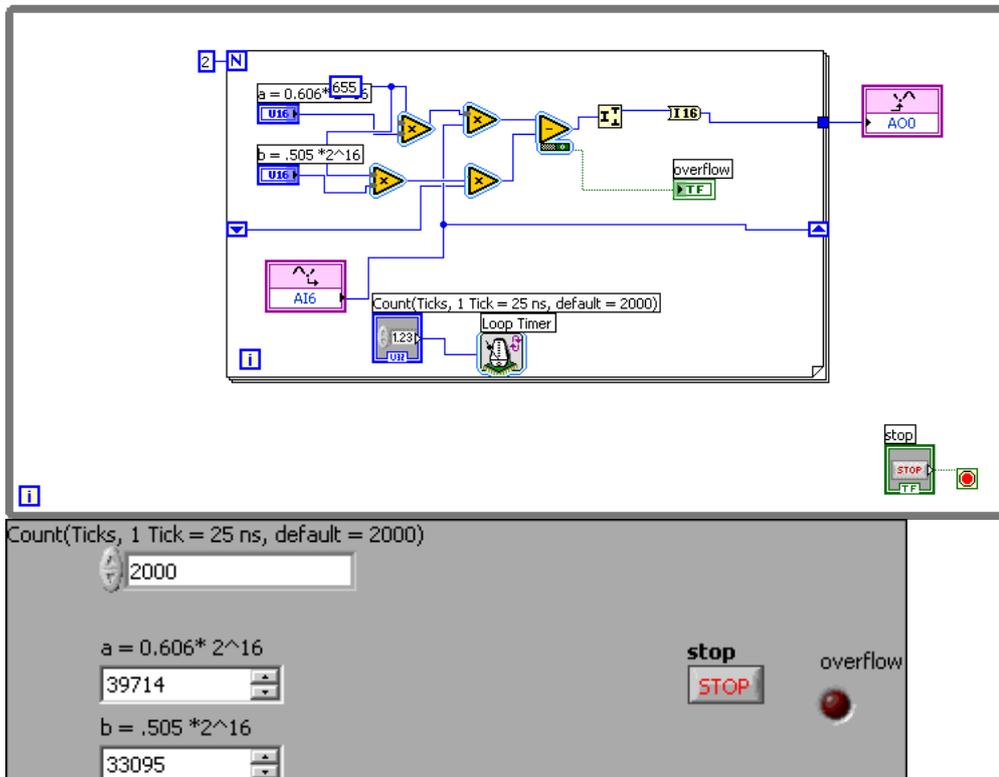
5.1 Printouts of the Program code and output

This VI demonstrates acquiring a buffer of AI samples from the FPGA memory, and the host VI reading, displaying and displaying the magnitude FFT (modified from the LabVIEW FPGA example of the same name). After this, the VI outputs the digital filter coefficients that can be used to recover the original signal.

NOTE: THIS VI POPS UP TWO DIALOG BOXES FOR SAVING DATA. THE FIRST DIALOG BOX IS FOR SAVING THE IMPULSE RESPONSE DATA FROM THE FPGA. THE SECOND DIALOG IS FOR SAVING THE FILTER COEFFICIENTS. THE DATA IS IN FLOATING POINT FORMAT, SINGLE COLUMN. SO, OPEN IT UP IN EXCEL TO CONFIRM IF THE DATA MAKES SENSE.

SCALE THE TIME AXES ACCORDING TO YOUR WAVEFORM. We had a simple RC circuit, so the time axis is approximately 8 time constants. Of course, scale the axis BEFORE running the VI.

SCALE THE FREQUENCY AXIS BELOW, DEPENDING ON THE FREQUENCY YOU EXPECT TO SEE. Of course, do this BEFORE running the VI.
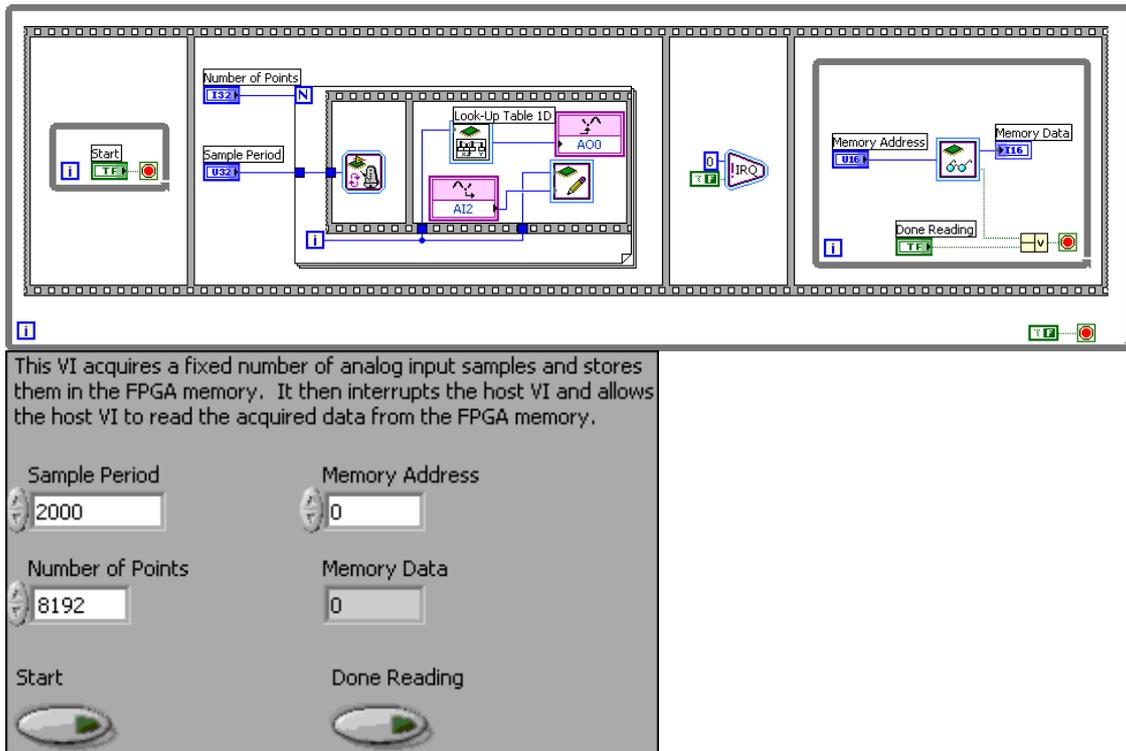
MANY THANKS TO MIKE AND THE ENTIRE FPGA TEAM AT NI! Without their help this would have been impossible!

This VI implements the difference equation: y(n) = 0.606*x(n) + -.505*x(n-1) = 0.606*x(n) - .505*x(n-1) Since the FPGA does not have an FPU, we scale the filter coefficients to a 16-bit unsigned integer, perform the filtering and then extract the approrpiate bits from the result.

This VI acquires a fixed number of analog input samples and stores them in the FPGA memory. It then interrupts the host VI and allows the host VI to read the acquired data from the FPGA memory.

Sample Period
2000

Memory Address
0

Number of Points
8192

Memory Data
0

Start

Done Reading

5.2 Handwritten Data Sheet

Filter Coefficient Raw Data
0.125
-0.137
0.015
-0.002
0.000
0.000
… (all zeros afterwards)

Filter Coefficients
0.125
-0.137

0.125 * 2^17, compensating filter is 16 bits, multiplies by 2 to double the amplitude.
-0.137* 2^17

Multiplies by 2 is not necessary

16384
17984