The University of Queensland
School of Information Technology and Electrical Engineering

ELEC3004/7312: Signals Systems & Controls

EXPERIMENT 2: SAMPLING AND RECONSTRUCTION ON THE NEXYS 2

Aims

In this laboratory session you will:

1. Gain familiarity with the workings of an audio codec on the Nexys 2;
2. Gain practical experience of the sampling and reconstruction of analogue signals, in particular you will characterise the anti-aliasing and reconstruction filters and observe the effects of aliasing;

Introduction

The Digilent Nexys 2 provides an FPGA-based **codec** and DSP system, where the functionality is implemented in VHDL, a **H**ardware **D**escription **L**anguage, rather than in conventional software. This implementation uses a **schematic** approach, so that the experimenter can see the structure of the DSP system graphically, then view and modify as necessary, the contents of the relevant modules in the system.

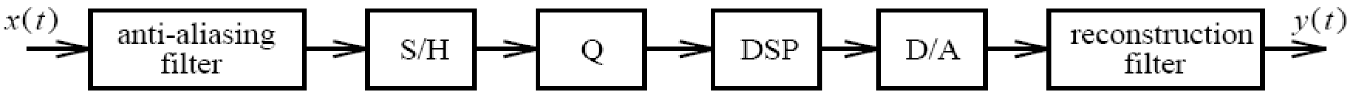


Figure 1: A block diagram of a practical DSP system.

Figure 1 shows a block diagram of a typical digital signal processing (DSP) system. On the Nexys 2 the ADC has a low pass filter with nominal cut-off frequency of around 500 kHz, however additional 10 kHz lowpass filters can be inserted at input and output. The Nexys 2 codec is implemented as separate analogue to digital (A2D) and digital to analogue (D2A) converters, these being represented by the sample and hold (S/H) plus quantiser (Q) and D/A blocks in Figure 1 respectively. Finally, the DSP block in Figure 1 is where the difference equation for implementing the desired digital filter is performed.

Equipment

1. PC with Xilinx ISE , Digilent Adept & Matlab;
2. PMOD AD1 and DA2 boards
3. 2 x PMOD CON4 boards
4. ADC and DAC Filter boards
5. Nexys 2 + JTAG interface cable/s;
6. Oscilloscope (preferably with FFT function);
7. 2 x cable: mono RCA male to mono BNC male, 0.5 - 1 metre long
8. Mono or stereo 3.5mm male to mono or stereo RCA male
9. Mono or stereo Y-adaptor, 3.5mm Male to 2 x 3.5mm Female
10. Signal Generator;
11. External speakers + audio jack cable + power pack;
12. 1 BNC T-adaptor M to 2F (F-M-F);
13. 1 x cable BNC Male to BNC Male, 0.5 - 1 metre long.

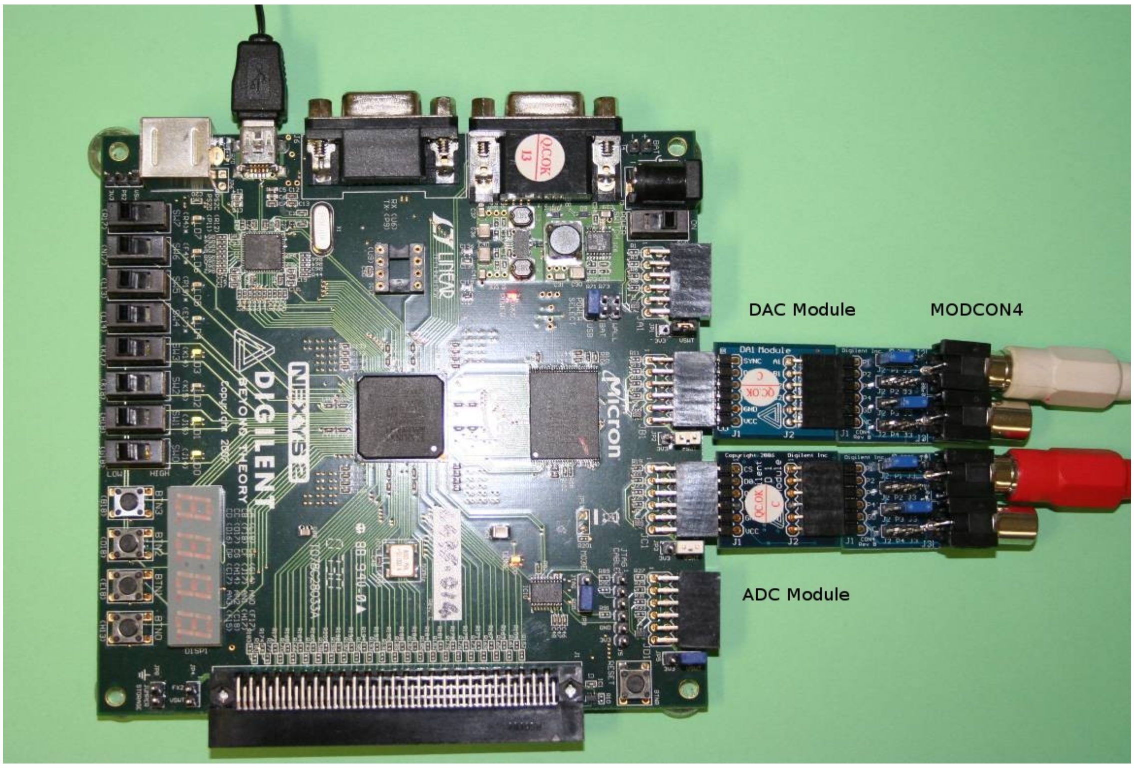


Figure 2. Nexys2 board with cable connections

Preparation

Note: preparation will be checked at the start of each laboratory class.

Answer the following questions:

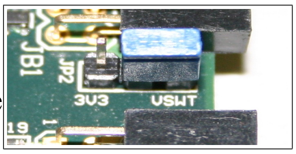
1. Bearing in mind that the anti-aliasing filter on the PMOD-AD1 is set to approx. 500kHz, explain what happens to the analogue signal, x(t), after it has passed through this filter and is presented to the input of the A2D, when:
   a. x(t) is a sinusoidal signal of frequency 10 kHz;
   b. x(t) is a square wave (50% duty cycle) of fundamental frequency 4 kHz.
2. Write an equation describing the Fourier series representation of a square wave and of a triangle wave.
3. If the DSP block in Figure 1 is assumed to have a sampling frequency of 25 kHz and performs the following difference equation: y[n] = x[n], (that is, the output of the A2D is copied directly to the input of the D2A) what is the frequency and approximate amplitude of the signal, y(t), observed at the **output** of the reconstruction filter, when:
   a. x(t) is a sinusoidal signal of frequency 10 kHz;
   b. x(t) is a sinusoidal signal of frequency 12.5 kHz;
   c. x(t) is a sinusoidal signal of frequency 15 kHz;
   d. x(t) is a sinusoidal signal of frequency 25 kHz;
   e. x(t) is a square wave of fundamental frequency 4 kHz;
4. Assume a 2^{nd} order Sallen-Key lowpass filter with corner frequency (Fc) of 10 kHz is used for the anti-aliasing and reconstruction filters.
   a. What is the expected attenuation at Fc?
   b. What is the cutoff slope in dB/octave and dB/decade?
   c. What is the expected attenuation of a 4 kHz sinewave after passing through two 2^{nd} order 10 kHz lowpass filters?
   d. What is the frequency of the 5^{th} harmonic of a 4 kHz square wave? What is the expected attenuation of this 5^{th} harmonic after passing through one 2^{nd} order 10 kHz lowpass filter?

Procedure

Part 1: Sampling and Reconstruction

With reference to EXPERIMENT 1: INTRODUCTION TO THE Nexys2, carry out the following:

1. Start Xilinx ISE and Digilent Adept
2. Connect PMOD DA2 to JB1, and connect PMOD AD1 to JC1, both on the top row, then connect a PMOD CON4 to each, so that your connection looks like figure 2 above.



3. Ensure that JP2 and JP3 (near JB1 and JC1) have jumper blocks connecting the centre pin and VSWT.
4. Using an appropriate cable and T-piece combination, connect the output of the signal generator via the T-piece, to PMOD AD1 and to channel 1 of the oscilloscope.
5. Connect the output of PMOD DA2 to channel 2 of the oscilloscope.
6. Set CH1 and CH2 of oscilloscope to **DC Coupling** from the CH1/CH2 Menu buttons
7. On the function generator, pull the **Offset** button out and set the **Amplitude** control counter-clockwise and select **SINE** wave.
8. Open the FPGA Project: **Prac2_Part1_2012.xise**
9. Open **Prac2_Part1_2012.sch** and **codec.vhd** file by double clicking on these file names.
10. **Ensure that slide switch 0 (SW0) is set to 0, i.e. down, or nearest the edge of the board.**

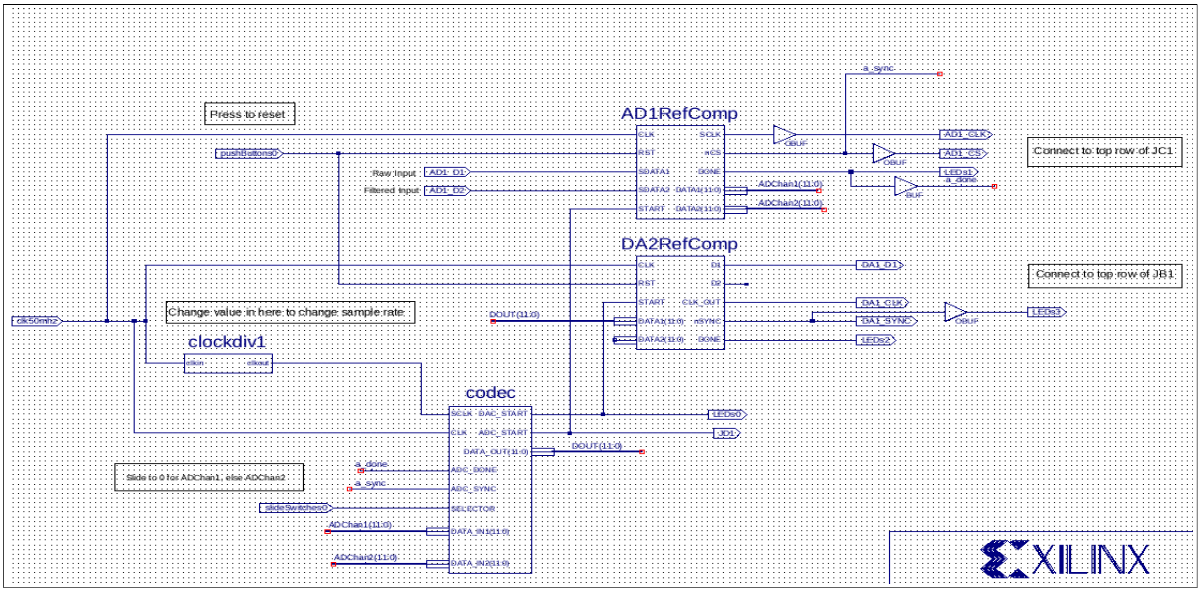


Figure 3. Sample and Reconstruct schematic

11. In the Design View select **Generate Programming File | ReRun All** and wait until the bitfile is produced.
12. Download it to the Nexys2 board using Adept.
13. You will need to experiment with offset voltage and amplitude (**Page 3**) on the signal generator. Suitable starting values would be approx 2.5 volts DC for **Offset**, then **Amplitude** of approx. 4.5 Vp-p.

Part 2: Tasks and questions

Please ensure you adjust your input so that the output signal is sinusoidal, not clipped or distorted.
For the phase response, just record the time difference (in uS) between input and output signal.

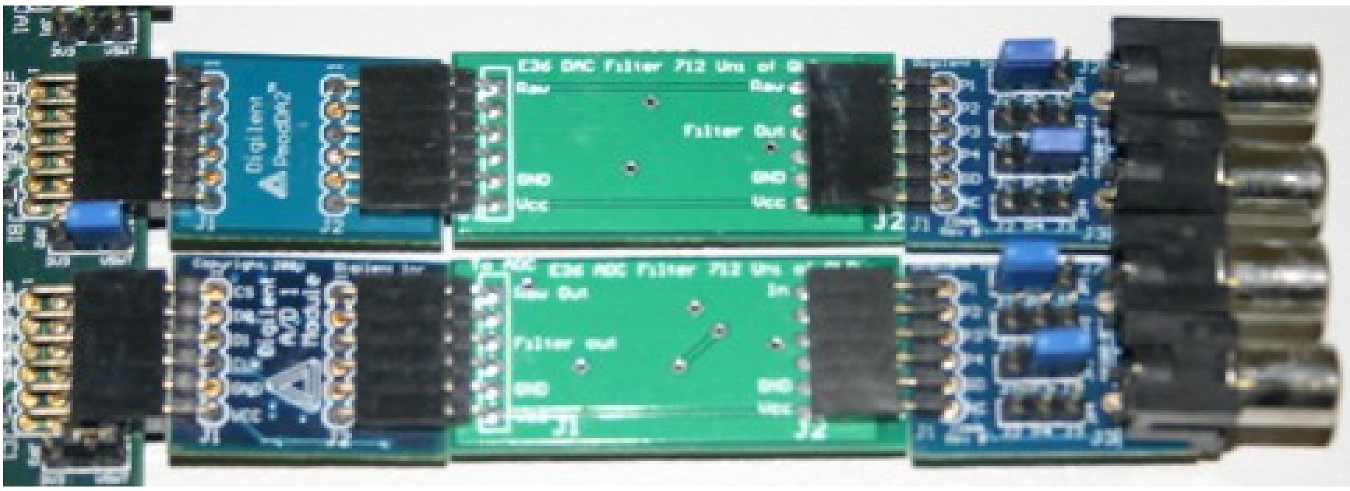
This section does not use the filter boards.

1. Use the signal generator to plot the magnitude and phase response, H(ω), of the system (i.e., from x(t), the input, through to y(t), the output) between DC and 12 kHz. To do this it is suggested that you make measurements at (at least) the following frequencies: 0.125, 0.25, 0.5, 1, 2, 4, 6, 8, 10, and 12 kHz;
   a. What components of the block diagram in Figure 1 are contributing to your measured H(ω)?
   b. Is your measured phase response linear? Should it be?
   c. Do your measurements confirm your answer to preparation question 1 (a)? If not, why?
2. Now slowly increase the frequency of the input sine wave, x(t), from 10 to 15 kHz.
   a. What happens to the frequency of the output sine wave, y(t)?
   b. Explain what is happening in terms of the Nyquist rate and the sampling theorem;
   c. Do your measurements confirm your answer to preparation question 2 (a), (b) and (c)?
   d. What happens to the frequency of the output sine wave, y(t), when the input sine wave, x(t), approaches 25 kHz? That is, when x(t) is equal to the sampling frequency.
3. Set up the function generator to produce a **square** wave. Observe the audio output on the CRO both in the time and frequency domain (*using the FFT function from the MATH menu*).
   a. What distortions do you observe in the audio output as you vary the fundamental frequency of the square wave from 100 Hz to 12 kHz? i.e., as compared to spectrum of the audio input.
   b. Do your measurements confirm your answer to preparation question 2 (d)?
   c. How can these distortions be explained? i.e., which components of Figure 1 are causing them? And what effect is aliasing having? Is your input signal amplitude too high?
   d. Observe the time difference between the square wave input and the reconstructed output. How does the time difference change as you vary the frequency from 0 to 4 kHz?
   e. What is the cause?

Part 3 :Tasks and questions

This section USES the filter boards.

Fit the DAC filter between the DA2 board and its CON4 board. Fit the ADC filter between the AD1 board and its CON4 board.



If you look at the schematics in the appendix, you will see that there are connections for Raw Out and Filtered Out. This allows you to experiment with various combinations of input and output signals, being:

a) Raw in, Raw out
b) Raw in, Filtered out
c) Filtered in, Raw out
d) Filtered in, Filtered out.

Slide switch 0, **SW0**, allows you to select raw In (0), or Filtered In (1) for the ADC. The DAC filter board has Raw Out on the top connector and Filtered Out on the bottom connector. Combination a) is the same as not having the filter boards at all.

Part 3 continued:

If you move SW0, you can select between a) and c) for the top connector (Raw Out), and between b) and d) for the bottom connector (Filtered Out).

Remember to keep the input pk-pk value under 5V.

Connect the function generator to CH1 of the oscilloscope and the ADC.
Set the function generator to Sinewave, 5 V p-p and DC offset of 2.5 volts.

For the following data, plot this in a spreadsheet and show to a tutor.

1. Connect the Raw Out (top connector) to CH2.

Raw in – Raw out

2. Move SW0 to 0 (Raw in) then sweep the frequency from about 100Hz to about 25 kHz and observe the output on CH2. Record the input and output amplitudes, plus Frequency Out at 1, 4, 10, 15 and 20 kHz.

Filtered in – Raw out

3. Now move SW0 to 1 (Filtered In) and repeat.

4. Now connect Filtered out (bottom connector) to CH2.

Raw in – Filtered out

5. Move SW0 to 0 (Raw in) then sweep the frequency from about 100Hz to about 25 kHz and observe the output on CH2. Record the input and output amplitudes, plus Frequency Out at 1, 4, 10, 15 and 20 kHz.

Filtered in – Filtered out

6. Now move SW0 to 1 (Filtered In) and repeat.

Set the function generator to squarewave and repeat 1 to 6.

Record the results in the tables below or in a spreadsheet.

Either printout your plots, or save the files to your network drive for later retrieval and submission for marking.

Part 4:

Q1. Did you notice any difference between the output frequency and the input frequency? If so, what is the cause?

Sinewave Filtered in – Raw out (SW0 = 1)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Sinewave Raw in – Filtered out (SW0 = 0)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Sinewave Filtered in – Filtered out (SW0 = 1)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Squarewave Raw in – Raw out (SW0 = 0)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Squarewave Filtered in – Raw out (SW0 = 1)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Squarewave Raw in – Filtered out (SW0 = 0)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Squarewave Filtered in – Filtered out (SW0 = 1)

Frequency in (kHz)	Vin pk-pk	Vout pk-pk	Frequency out
1			
4			
10			
15			
20			

Appendix:

Schematic diagrams for filter boards:

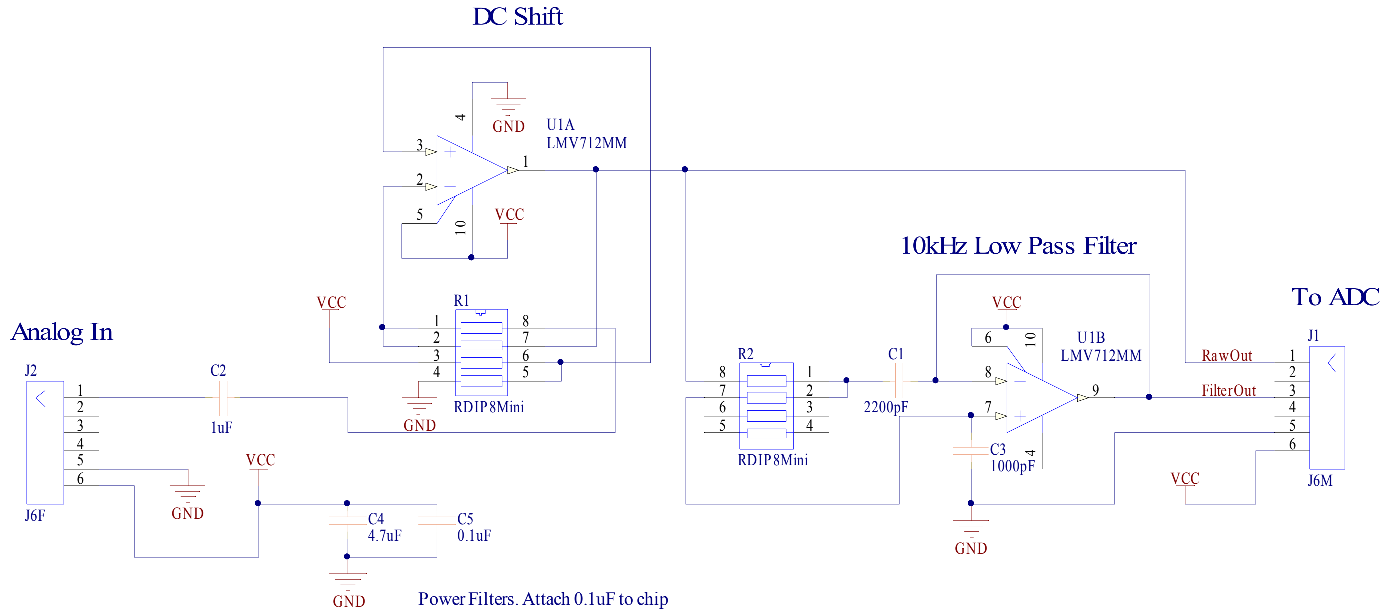


Figure 4: ADC lowpass filter

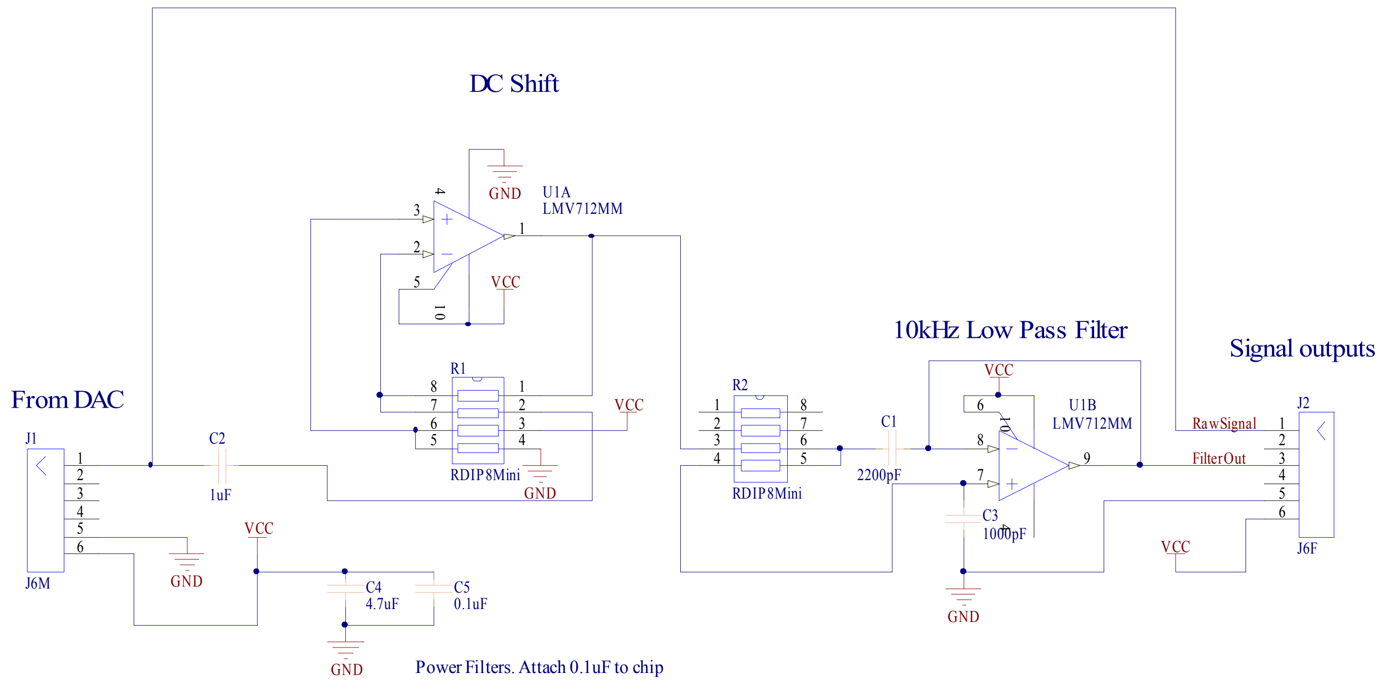


Figure 5: DAC lowpass filter