ECEN 5053 - Embedding Sensors and Actuators.

Lab No: 1 Condenser microphone interface

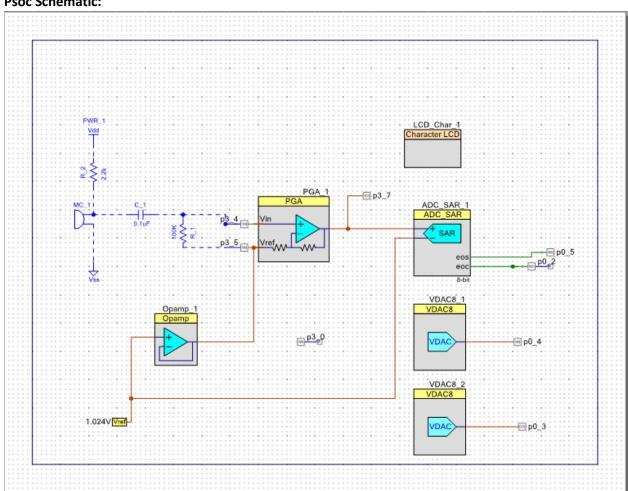
Report by: Monish H Nene

Goal: Familiarize yourself with using PSoC Creator. Learn how to implement a simple microphone interface. Implement simple digital filters.

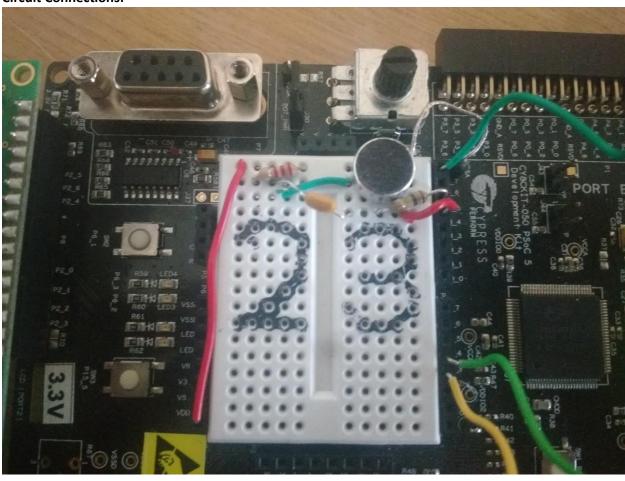
Equipment Used: Cypress PSoC CYCKIT-050B, Agilent U8002A power supply, digital oscilloscope, Buzzer ,Electret Condenser Microphone made by CUI Inc, model CMA-4544PF-W.

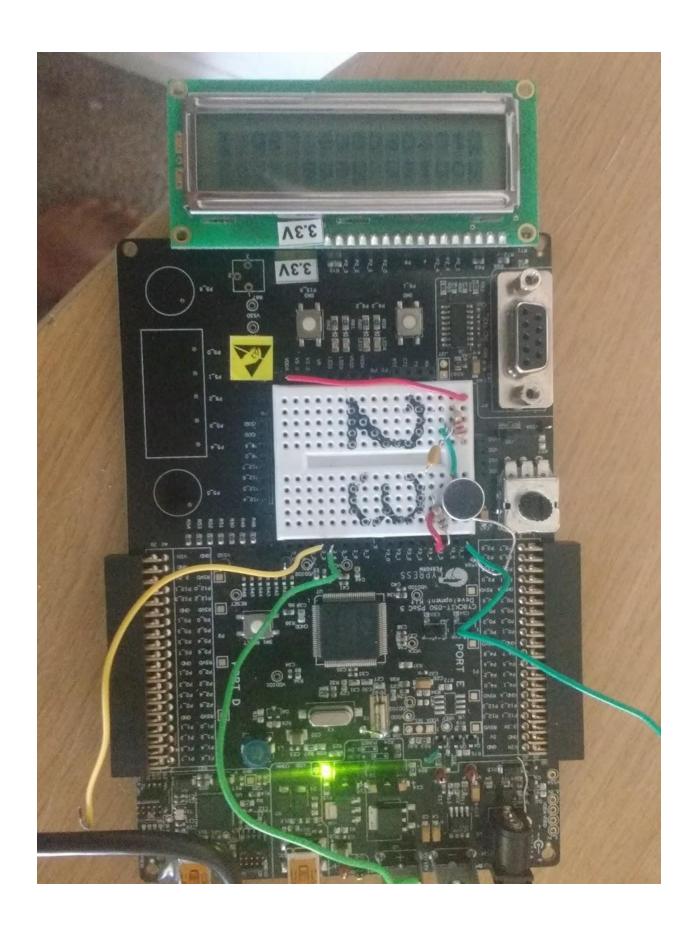
Methodology: Brief description of what experiments were performed to get the required analysis

Psoc Schematic:



Circuit Connections:





<u>Scope Shot 1:</u> ch 1 = signal before amplification (use ac coupling and high gain on the scope to get a large, detailed image), ch 2 = output of PGA (note PG settings used) dc coupled, 1. v/div to show the full range of the PGA output (which is 0 to \sim 4v)

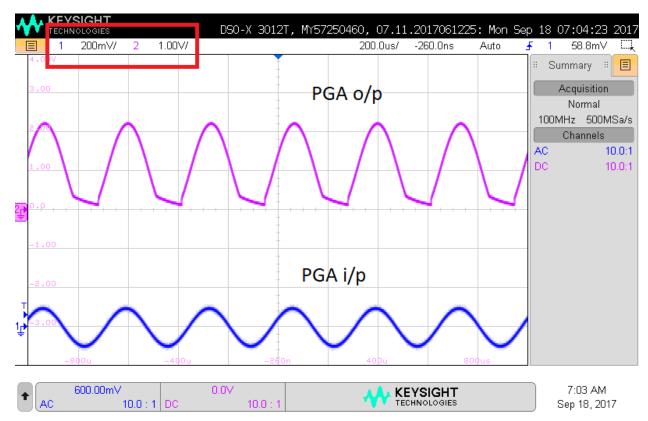
Question #1: What is the frequency of the tone produced by the buzzer?

Answer: 3KHz

Question #2: What is the minimum and maximum voltages of the output of the PGA stage?

Answer: Minimum = 0V

Maximum = Vref x $2 = 1.024 \times 2 = 2.048V$



If we increase the gain we get distorted output like above but the range doesn't increase. We have to set the gain such that we get no distortion in the output and it reached the min and max voltage.

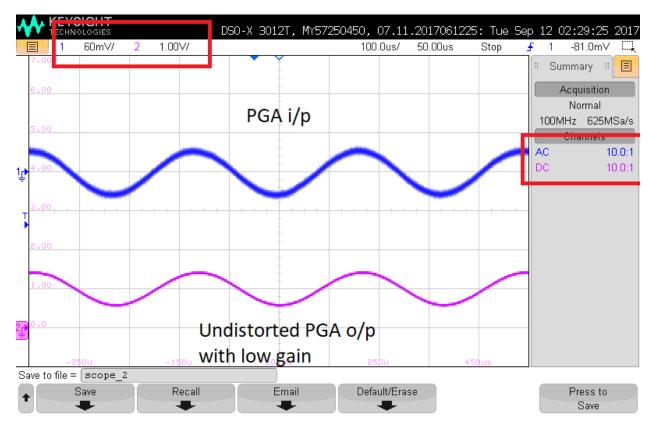


image 0: Scope shot of your measurement technique for Q3 & Q4.

*(UPDATED) Question #3: What is the loop execution frequency (how often are you reading the ADC)? What is the frequency that the ADC is sampling at? Use a GPIO pin and scope to determine this.

(add an image of the setup window).

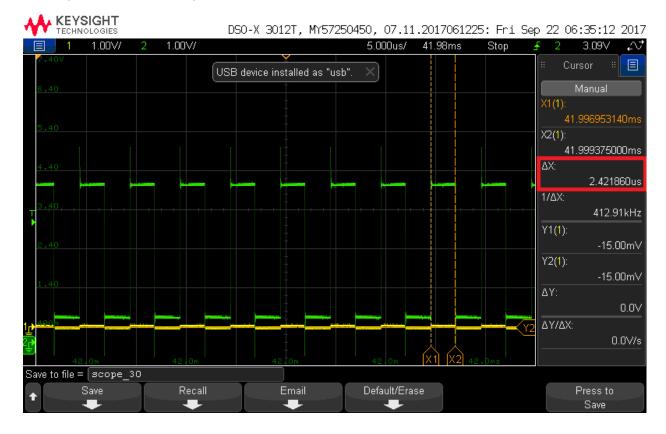
Answer:

The pin p3_3 is set or reset to get the loop duration.

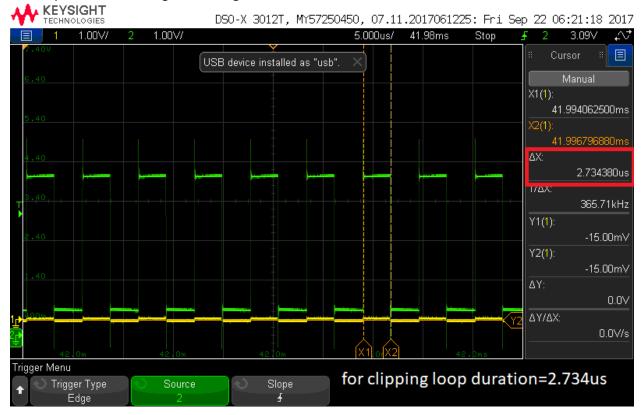
```
for (;;)
{
    if (p3_3_Read()==lu) {
        p3_3_Write(0u);
    }
    else
    {
        p3_3_Write(lu);
    }
    unfiltered = ADC_SAR_1_GetResult8(); // input from ADC 2's Compliment form
        VDAC8_1_SetValue(unfiltered);
}
```

The duration of the loop changes with the filter that is currently being used.

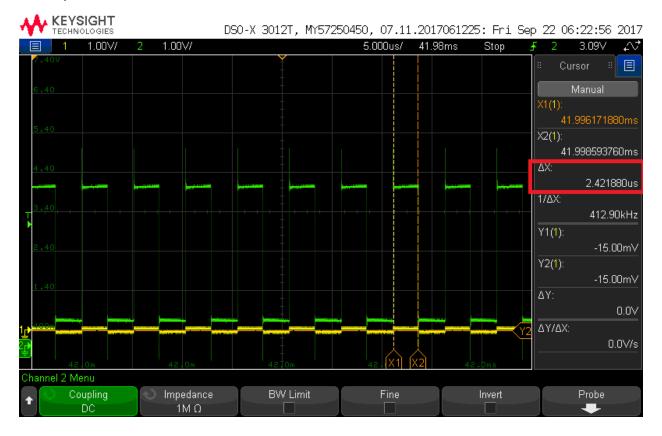
The loop duration for vdac op without filter= 2.42us



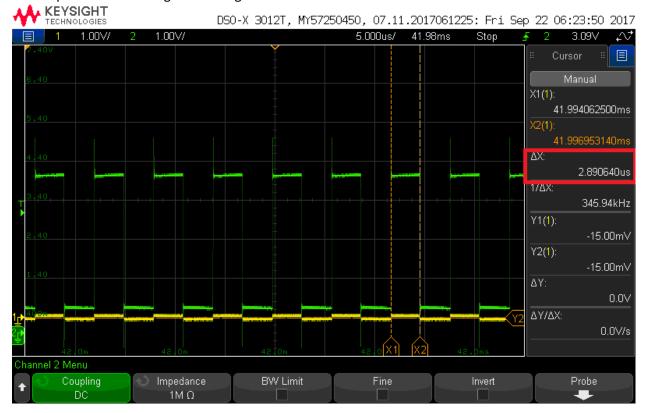
The loop duration for weighted average filter=2.734us



The loop duration for inverse filter=2.421us



The loop duration for weighted average filter=2.89us



Question #4: How long does it take the ADC to do a conversion? Note which ADC you used and what it's settings were.

Answer:

In Free running mode the conversion rate can be adjusted to overcome delay in other parts of the circuit. But the actual conversion rate is 100000 samples per second that comes around **10us**. It is the time for sampling + conversion. To find the conversion time we activate the eos(end of sampling) output of ADC along with eoc(end of conversion) output. Then we measure the time difference between eos and eoc pulses which is 7.9us. the So the **conversion rate of ADC_SAR is 7.9us**

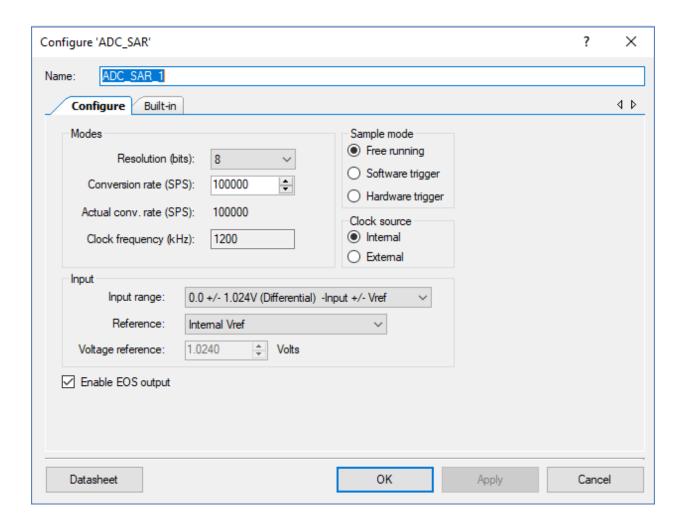
Mode

Manual

Y1: -15.00mV

Y2: -15.00mV

X1: 41.987100000ms



The duration between EOC pulses is also 10us in free running mode which confirms that the time required for ADC_SAR to do conversion is 10us.

Question #4A: How do the frequencies measure in Q3 and Q4 relate to your signal frequency? Does this satisfy the Nyquist Criteria? Is it much greater? Do you want it to be much greater or would you want it to just satisfy the Nyquist Criteria?

Answer: The frequency of the sample is 3kHz. We are sampling at 10kHz. So we satisfy the Nyquist Criteria that is the sampling frequency must be at least twice the frequency of signal to be sampled.

To get a good waveform with no distortions the sampling frequency must be at least 10 times the frequency of the input signal which comes around 30khz in our case. If we just satisfy the Nyquist Criteria then the output has a lot of distortions. We want our sampling frequency to be much greater i.e. 10 times and not 2 times which will just satisfy the Nyquist Criteria.

POA 1

Character LCD

Char 1

Character LCD

Character LCD

Char 1

Character LCD

POA 1

ADC SAR 1

ADC SAR 1

ADC SAR 1

ADC SAR 1

VDAC6 1

VDAC6 2

VDAC6 3

VDAC6 2

VDAC6 2

VDAC6 3

VDAC6 2

VDAC6 3

VDAC6 3

VDAC6 3

VDAC6 4

VDAC6 2

VDAC6 3

VDAC6 4

VDAC6 2

VDAC6 3

VDAC6 4

VDAC6 4

VDAC6 4

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VDAC6 2

VDAC6 2

VDAC6 3

VDAC6 3

VDAC6 3

VDAC6 4

VDAC6 5

VDAC6 4

VDAC6 5

VDAC6 4

VDAC6 4

VDAC6 4

VDAC6 4

VDAC6 5

VDAC6 5

VDAC6 5

VDAC6 7

VDAC

Image 1: show your schematic. It should be clear and well organized.

Scope Shot 2: ch 1 = PGA output, ch 2 = DAC output. Use same vertical scales for both and dc coupling.

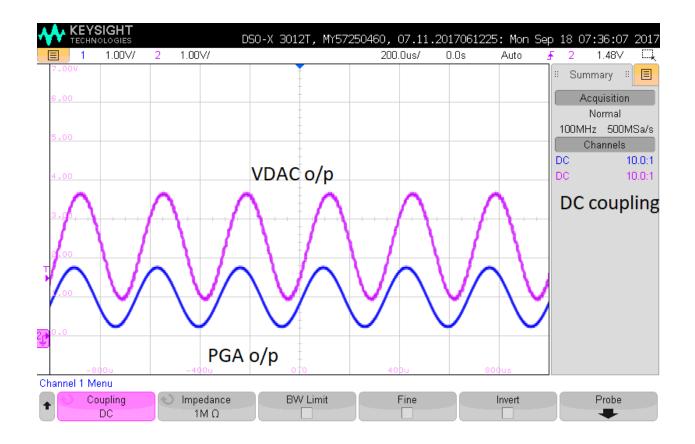
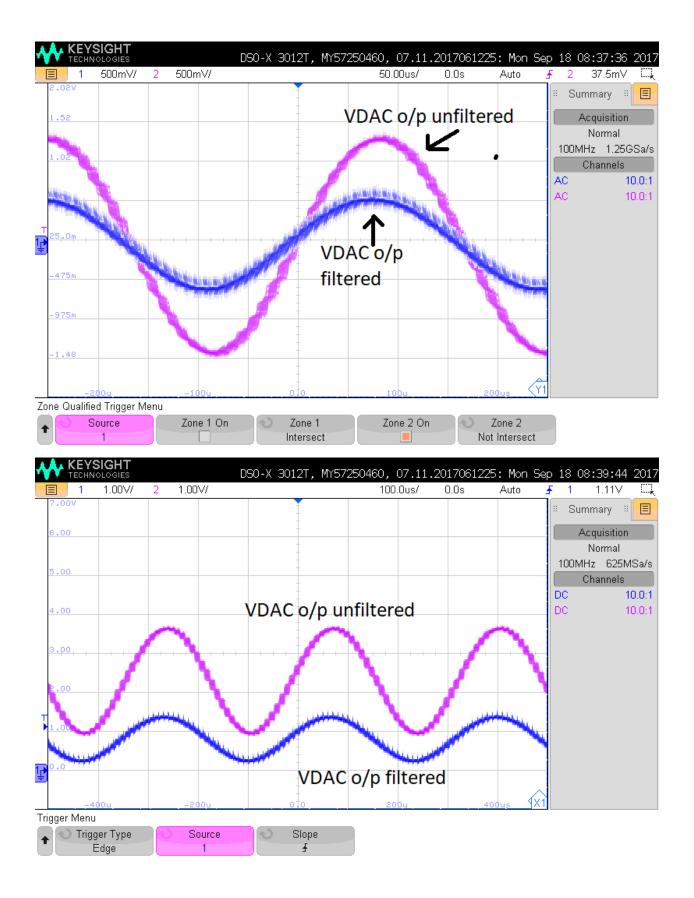


Image 2: Show the equation for your filter.

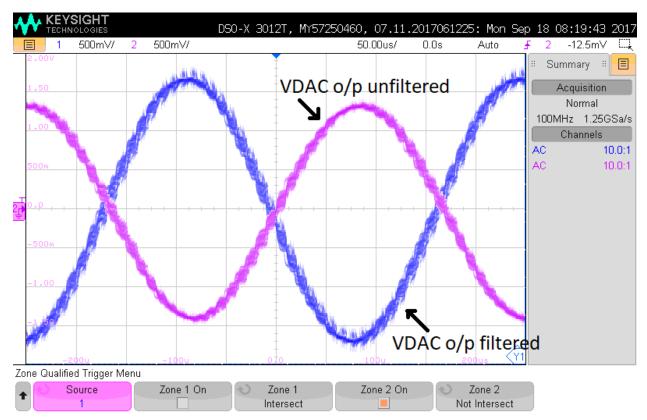
Filtered =(6*prevvalue + 5*prevvalue2 +k)/3

```
int prevvalue=1,prevvalue2=1,k=0;
      /* Place your initialization/startup code here (e.g. MyInst Start()) */
     LCD();
     PGA_1_Start();
     Opamp_1_Start();
     ADC SAR 1 Start();
     ADC_SAR_1_StartConvert();
     VDAC8 1 Start();
VDAC8 1 SetRange(255);
VDAC8 2 Start();
VDAC8 2 SetRange(255);
     for (;;)
           if(p3_0_Read() == lu)
              p3_0_Write(Ou);
           else
              p3_0_Write(lu);
          adcop = ADC_SAR_1_GetResult8();
          unfiltered = adcop + 128; // input from ADC 3's Compliment form
          VDACS 1 SetValue(unfiltered);
          k=unfiltered:
          filtered = (6*prevvalue+5*prevvalue2+k)/3; //For weighted average half
//filtered = 234-unfiltered; //For inverse
            /*if(unfiltered<88)
              (filtered=88;)
            else if (unfiltered>168)
              (filtered=168:)
            else
              (filtered=unfiltered: */ //For mean clipping
          VDAC8_2_SetValue(filtered);
          //prevvalue2=prevvalue;
          //prevvalue=unfiltered;
-/* II END OF FILE */
```

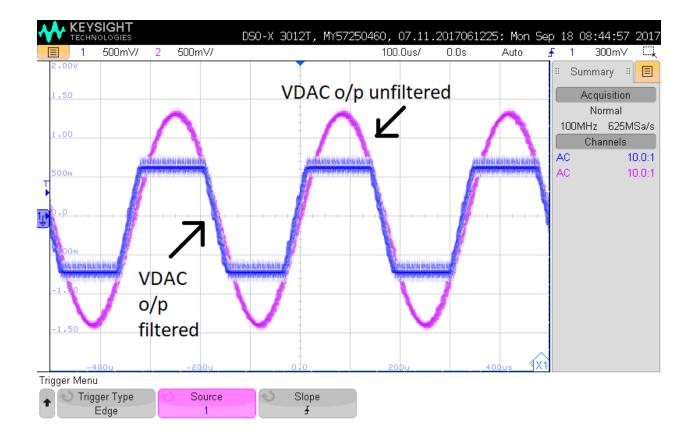
Scope Shot 3: ch 1 = unfiltered signal from a first DAC, ch 2 = filtered signal from a second DAC. (Make sure that both channels are at the same scale for comparison purposes.).



Scope Shot 4: ch 1 = unfiltered signal from first DAC, ch 2 = inverted signal from a second DAC. Use the same scale for both and set the offset voltage to the same grid line on the screen for easy comparison.



Scope Shot 4: ch 1 = unfiltered signal from the first DAC, ch 2 = clipped signal from the second DAC. Use the same scale for both and set the offset voltage to the same grid line on the screen for easy comparison.



Extra Credit: (+0.5 per upto 2 points, at the discretion of the instructing team):

- 1. Any interesting or unusual findings, supported with a description of what is happening.
- 2. Additional functionality worthy of extra credit, must be documented fully in your report.