ME 461 Laboratory #3
Analog-to-Digital Conversion

Goals:

1. Learn how to configure and use the MSP430's 10-bit SAR ADC.
2. Measure the output voltage of your home-made DAC and compare it to the expected value.
3. Calibrate and use the MSP430's internal temperature sensor.
4. Design and implement an RC anti-aliasing filter.
5. Solder a microphone circuit to the breakout board and use it to sample audio signals.

Exercise 1:

In this exercise you will configure the MSP430's 10-bit analog-to-digital converter (ADC) to sample the output of the PWM + low pass filter you designed in lab 2.

This would be a good time to take another good look at, the lecture notes, the MSP 430 User's Guide ADC10 section, and the "msp430x22x2.h" header file. As you have heard many times already, control register names and predefined constants with (relatively) easy to remember names are declared in the "msp430x22x2.h" header file. For example with this source code:

```
ADC10CTL0 = SREF_0 + ADC10SHT_2 + ADC10ON + ADC10IE;
```

On line 172 of the "msp430x22x2.h" header file you will find:

```
#define SREF_0          (0*0x2000u)  /* VR+ = AVCC and VR- = AVSS */
```

Obviously, zero times anything is zero and we can assign this value of SREF_0 to ADC10CTL, as in

```
ADC10CTL0 = SREF_0;
```

The binary representation of ADC10CTL0 will then be %0000 0000 0000 0000. Note that the three most significant bits are '000' and, referring to the MSP 430 User's Guide, you will see that setting the three most significant bits of ADC10CTL0 to '000' tells the microprocessor to use V_{CC} as the positive voltage reference (V_{R+}) and V_{SS} (Ground) as the negative voltage reference (V_{R-}).

As you know from Lab #2, the full-scale range of your home-made DAC is V_{SS} = 0V to V_{CC} = 3.6V. This means, you can use the example above to help get you started with the configuration of the ADC. Note that because we are using a voltage reference set that is already available on the board (V_{CC} and GND), you do not need to worry about using any of the reference generator features.

On line 167 of the "msp430x22x2.h" header file you will find:

```
#define ADC10SHT_3          (3*0x800u)  /* 64 x ADC10CLKs */
```

Thus, the binary representation of ADC10SHT_3 = %0001 1000 0000 0000. We could now add SREF_0 and ADC10SHT_3 together, as in

ADC10CTL0 = SREF_0 + ADC10SHT_3;

Now the binary representation of ADC10CTL0 = %0001 1000 0000 0000. Note that three most significant bits remain '000' but the fourth and fifth most significant bits are now '11'. Again referring to the MSP 430 User's Guide you will see that setting the three most significant bits of ADC10CTL0 to '000' tells the microprocessor to use V_{CC} as the positive voltage reference (V_{R+}) and V_{SS} (Ground) as the negative voltage reference (V_{R-}). Setting the fourth and fifth most significant bits (bits 12 and 11 of a 16 bit value) of ADC10CTL0 to '11' tells the microprocessor to set the sample and hold time to be equal to 64 cycles of the ADC10 clock source.

The appropriate sample and hold time to use depends upon both the internal input resistance of the microprocessor and the external source resistance associated with any circuitry attached to the ADC input pin. Now refer to section 22.2.5 of the MSP430 user's guide to determine an appropriate minimum sample and hold time for your situation. The external (source) resistance R_S is the value of R in your RC low pass filter. In calculating the sample-and-hold time, assume your DAC output is roughly constant relative to the ADC sample rate so that you may neglect the settling time of the external RC lowpass filter. Record the minimum sample and hold time below.

$T_{S,min}$ = _____ seconds

Check the sample-and-hold time with your TA. It is recommended that you use the dedicated ADC oscillator, ADC10OSC, for the ADC10 conversion clock source (assume its speed is exactly 5MHz). Your minimum sample-and-hold time will dictate whether you need to divide the ADC10CLK source with the `ADC10DIVx` bits. Record the ADC clock rate after any ADC10DIVx divisions below and the resulting number of ADC10 clock pulses (an integer) that correspond to the minimum sample time you recorded above.

ADC10CLK rate after divide: _____ MHz $T_{S,min}$ = _____ ADC10 CLKS

Check these with your TA.

Continue to determine the remaining configuration parameters according to the following guidelines. Note that you are strongly encouraged to start making regular use of the predefined constants found in the "msp430x22x2.h" header file and to become comfortable combining them as shown in the examples above (and in the code examples provided in the lecture notes). Doing so will greatly simplify the effort required to properly configure control registers and improve the readability of and portability of your code.

You should connect the PWM DAC output to one of the analog inputs on the port 2 pins. To set that pin to analog input mode, you need to set the corresponding bit in the `ADC10AEx` register. Also, you must tell the ADC which input channel to convert. You may use any of the conversion triggers (referred to in the user's guide as the *sample-and-hold source*) you like. The `ADC10SC` bit is a software trigger that

is easy to use and thus recommended at this point, but the other (timer-based) triggers are better suited to fast sample rates (>10kHz) and may be necessary in future lab exercises. Whatever trigger source you use, you should **trigger conversions every millisecond**. Use the knowledge (and code) you gained from the first half of Lab 2 to generate a 1ms timing interval. For now, use the straight-binary data format.

Finally, make sure you enable the ADC10 interrupt and turn the ADC10 on. For general ADC conversion flow, refer to figure 22-5 in the user's guide. Your program should control ENC and ADC10SC in accordance with this diagram. Also, not because you have to but because it is the appropriate place to perform this operation, be sure that you only read the conversion result (`ADC10MEM`) in the interrupt service routine. In your program, print out the raw conversion results and the corresponding integer millivolts to the serial port. Also, print the expected DAC voltage as you did in Lab 2. In performing the conversion from raw ADC results to millivolts, you may not use floating-point numbers, so keep in mind the caveats of integer math (such as 2/3 = 0). **Demonstrate to your TA.**

Exercise 2:

Now, change the analog input channel so that the ADC converts the internal temperature sensor's voltage. You will also have to modify the ADC clock input divider and sample-hold time to achieve the large minimum sample period listed in the user's guide section 22.2.8. Print the voltage in millivolts to the terminal window. **Demonstrate to your TA.**

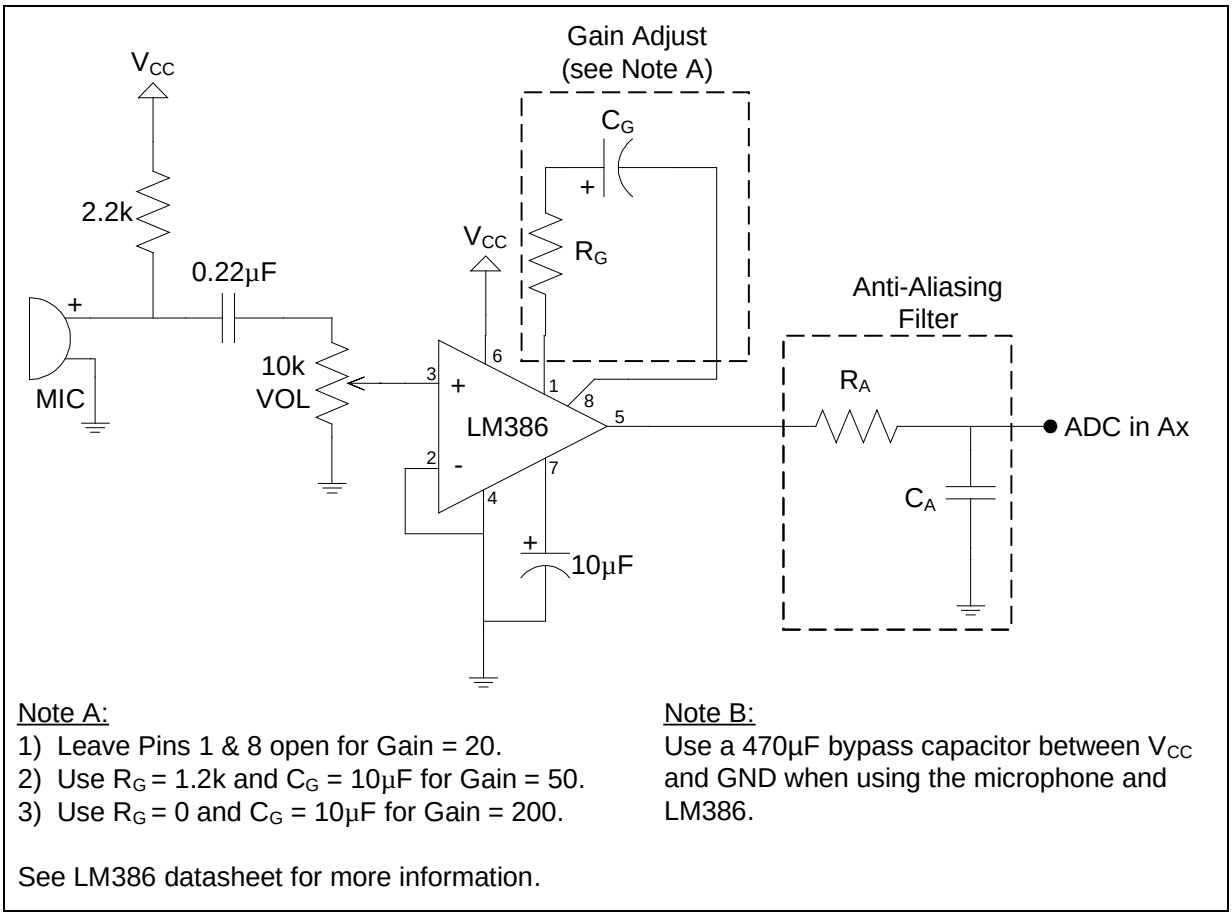
You will now calibrate the temperature sensor using the thermometer found in lab. Refer to section 22.2.8 of the user's guide. As you can see from Figure 22-13, the sensor voltage is a linear function of temperature. You may use the published value of the slope (3.55mV/^{o}C), but you will have to determine the offset voltage using the ambient temperature in lab and the corresponding sensor voltage. Record the value of the temperature sensor offset voltage below.

$$V_{TEMP}\,(mV) = 3.55 \times Temp(^{o}C) + \_\_\_\_\_\ mV$$

Once you have found the offset voltage, convert the temperature sensor voltage to degrees Celsius and print the value to the serial port. **Demonstrate to your TA.**

Exercise 3:

In this exercise, you are going to solder a microphone circuit to the breakout board and use it to sample audio signals. The circuit schematic is shown below.



Microphone circuit schematic

Notice that the schematic includes an RC anti-aliasing filter on the output of the amplifier. This is a necessary component of a sampled-data system whenever the input signal is not band-limited to half the sample rate. Before you begin soldering the components, you are going to design the anti-aliasing filter.

Our goal for the microphone circuit is to sample common audio signals. The audio signals you hear from a CD or mp3 player have been sampled at a rate of 44.1 kHz. This ensures that every signal you can hear with your ears is accurately represented. In most cases, though, a sample rate this high is more than enough. It turns out that most speech and common sounds are in the low-frequency range. We are going to use a sample rate of **10 kHz**, which means we can capture audio signals in the 0 - 5 kHz range. However, there will still be some sounds and noise outside this range. The goal of the anti-aliasing filter is to attenuate frequencies above the 5 kHz Nyquist frequency.

Your job is to select an RC that satisfies the requirement that at 5 kHz the attenuation should be roughly 50%, that is, only half the amplitude of a 5-kHz signal should be passed through. *Hint:* recall that the transfer function of an RC low pass filter is $G(j\omega) = 1/(1+RCj\omega)$. Record the cutoff frequency and the RC that achieves this below. Feel free to use MATLAB (with the `tf` and `bode` functions) to check your results.

f_C: _____ Hz RC: _____

Check these numbers with your TA. Now, from the list of components written on the whiteboard, choose a resistor and a capacitor that gives an RC close to what you calculated. Record the values of these components below. **Check with your TA.**

R_A: _____ C_A: _____

Now, use the schematic above and the demo board in lab to guide you as you solder the components to your board. For now, leave pins 1 and 8 on the LM386 open for a gain setting of 20. You may need to increase the gain later. When you are done soldering the components to the board, scope the output of the amplifier as you make some noise into the microphone. You should see the "typical" audio signal waveform biased around ½ V_{CC}. If the output of the amplifier uses only a small portion of the voltage range (0V-V_{CC}) for reasonably audible noises, increase the gain according to the note in the schematic. **Show the audio signal to your TA.** Also, scope the output of the anti-aliasing filter and see if you notice any differences.

Now, sample the microphone circuit's output at a rate of **10 kHz**. You may use the software trigger bit (`AD10SC`) as long as you don't run into lags created by calling an ISR too often. You will have to perform another minimum sample-hold time calculation using the R_A you found above. Change the data format to 2's complement. This will cause the ADC to read 0 when there is no input to the microphone, since it is biased at ½ V_{CC}. Write code to light an LED when the magnitude (absolute value) of the ADC reading is greater than something like half of the maximum. You do not need to convert the ADC reading to a voltage. *Hint:* in 2's complement format, the (linear) mapping from voltage to digital reading for our reference set is 0V → -512, $V_{CC}/2$ → 0, and V_{CC} → 511. **Demonstrate to your TA.**