Acoustical Analysis Unit

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Design Review Ver. 3 for ECE 445, Senior Design, Spring 2013

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March 11, 2013

Project No. 14

Contents

1 Introduction

1.1 Statement of Purpose

The goal of this project is to design an acoustical analysis unit that will be used for research. Professor Swenson and Dr. White presented this project idea to the senior design class. The final product should be able to do real time analysis of acoustical waves between 20Hz to around 10kHz and show the data in a simple and clean manner. Our device should be able to take input from a microphone probe, such as the Brel & Kjr microphones that the army acoustics research lab possesses. The resulting waveforms will be display or analyzed by the device, as the user commands. The user will be able to treat this like any other lab equipment, as it will have several menus and user inputs for navigation. All hardware and software will be properly documented so that the research team will be able to add on new functions as needed, making it far better than any consumer off the shelf equipment.

1.2 Objectives

1.2.1 Goals

- Create a device capable of being more portable to be used outdoors and that can be used with different types of microphones
- Use audio inputs from microphones to graph and collect acoustic intensity
- Have modularity in programming to allow additional data collection, functionality, calculations, etc. for end customer

1.2.2 Functions

- Inputs audio from microphones
- Distinguishes what noise/audio coming from each microphone
- Outputs acoustic intensity in the form of moving graphical bars
- Has various physical buttons for menus

1.2.3 Benefits & Features

The current device used for acoustical analysis with microphone probes is quite cumbersome and heavy to carry around in the field. This is not very suitable for the reason that the microphones and device need to be portable in order to be brought around outdoors. As for the user interface, it is quite difficult and not intuitive since there are too many buttons and options. Dr. Swenson and Dr. White would like the device to be more user-friendly and more efficient when trying to take in audio data from the microphone probes. Also, they would like the software to be modular such that it can be easier to add different functions.

So, our device will solve these by being able to:

- Be smaller and lighter for ease of use outdoors

- Have an intuitive program that is capable to be easily modified to increase the amount of functions
- Be compatible with the existing 2-probe microphone (BNC connections)

As of right now, the screen on the original device is quite dark. We chose to implement a large 3.5 inch LCD with a backlight. This would give the device a cleaner look without sacrificing any of the data/menus that need to be shown.

Our device will have an intuitive program and interface so that Dr. Swenson, Dr. White, or other individuals would like to refine the functions already implemented or input additional functions. The user interface should be spread out nicely and not have too many menus. There should be a clear and understandable way of getting the desired function.

2 Design

2.1 Block Diagrams

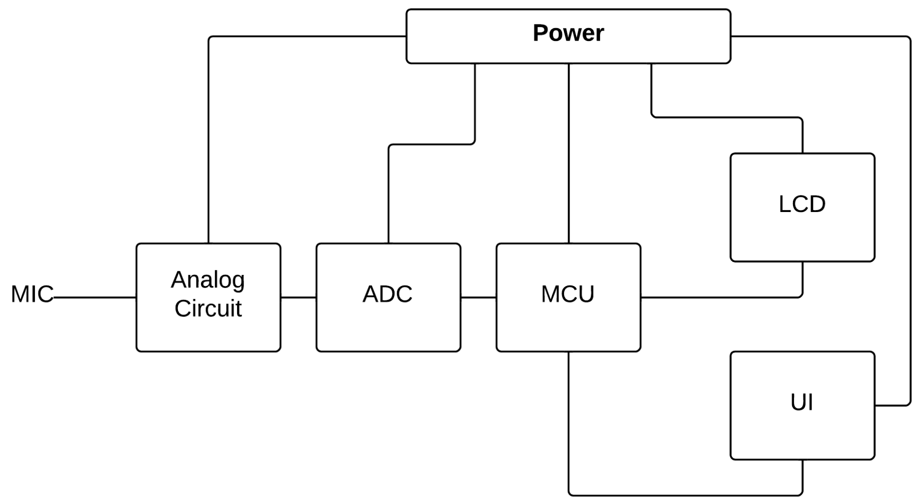


Figure 1: Top Level Block Diagram

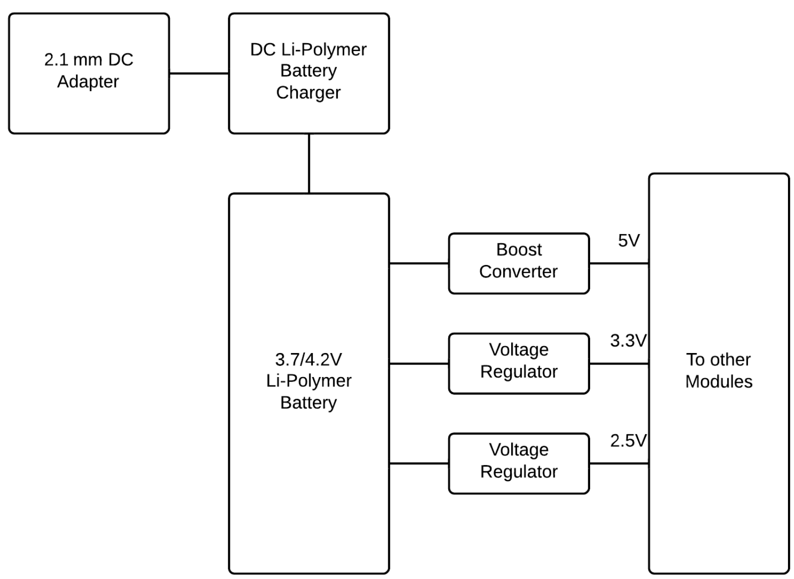


Figure 2: Power Supply Block Diagram

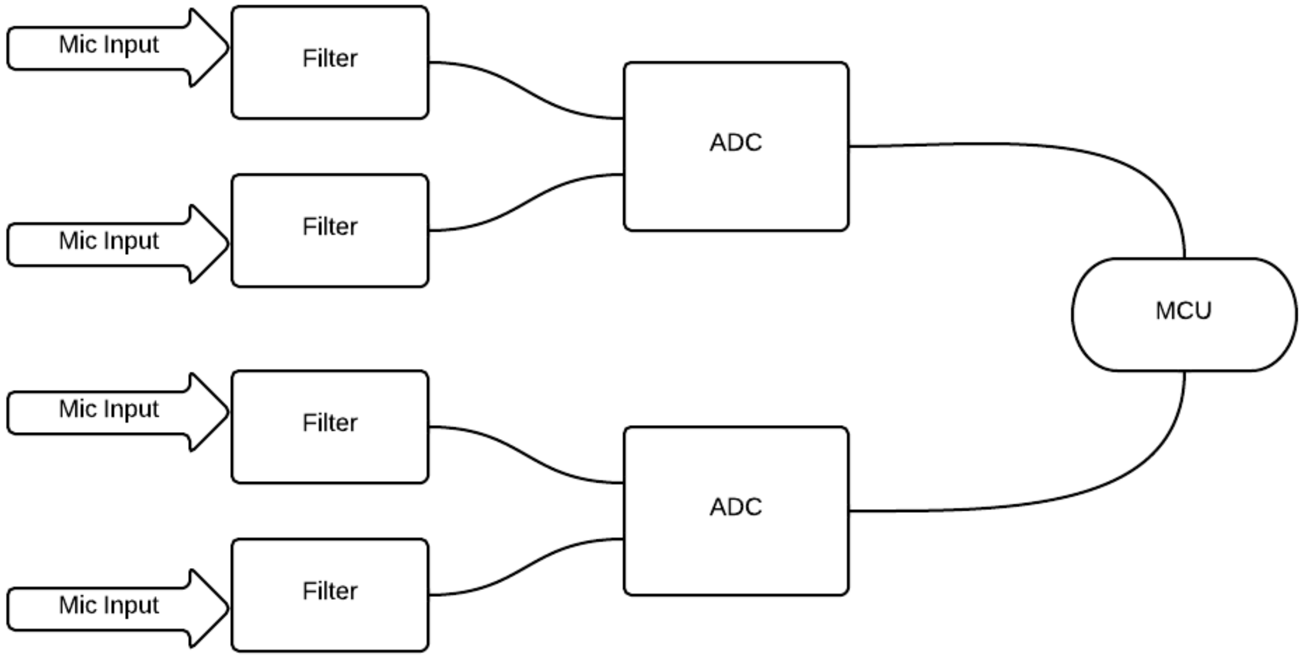


Figure 3: Analog Input Block Diagram

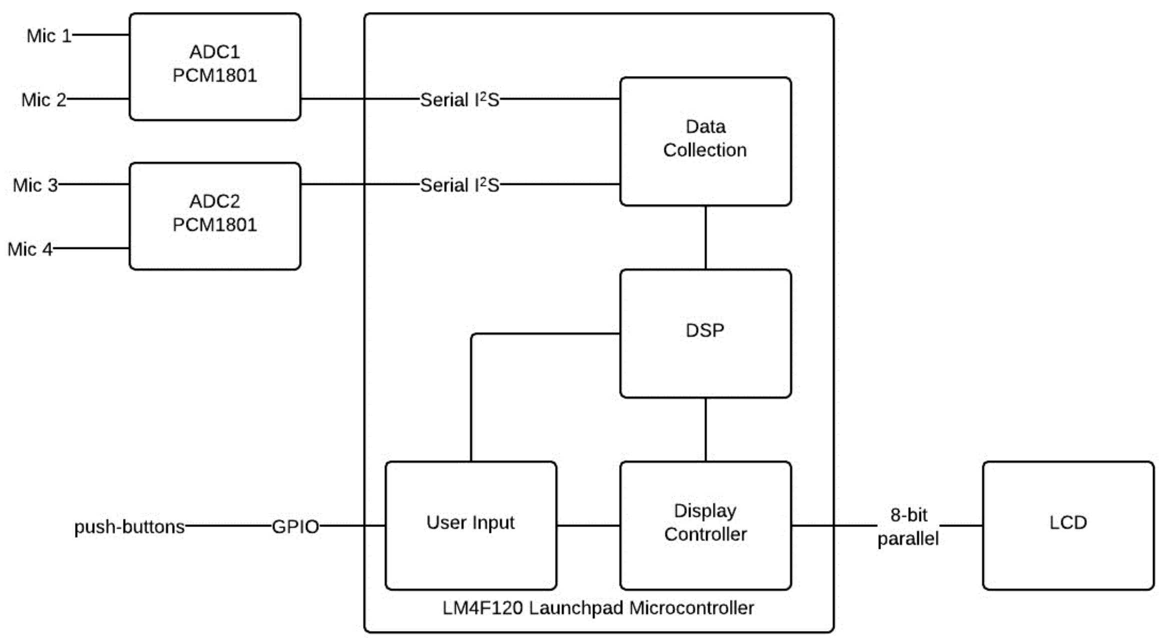


Figure 4: MCU Block Diagram

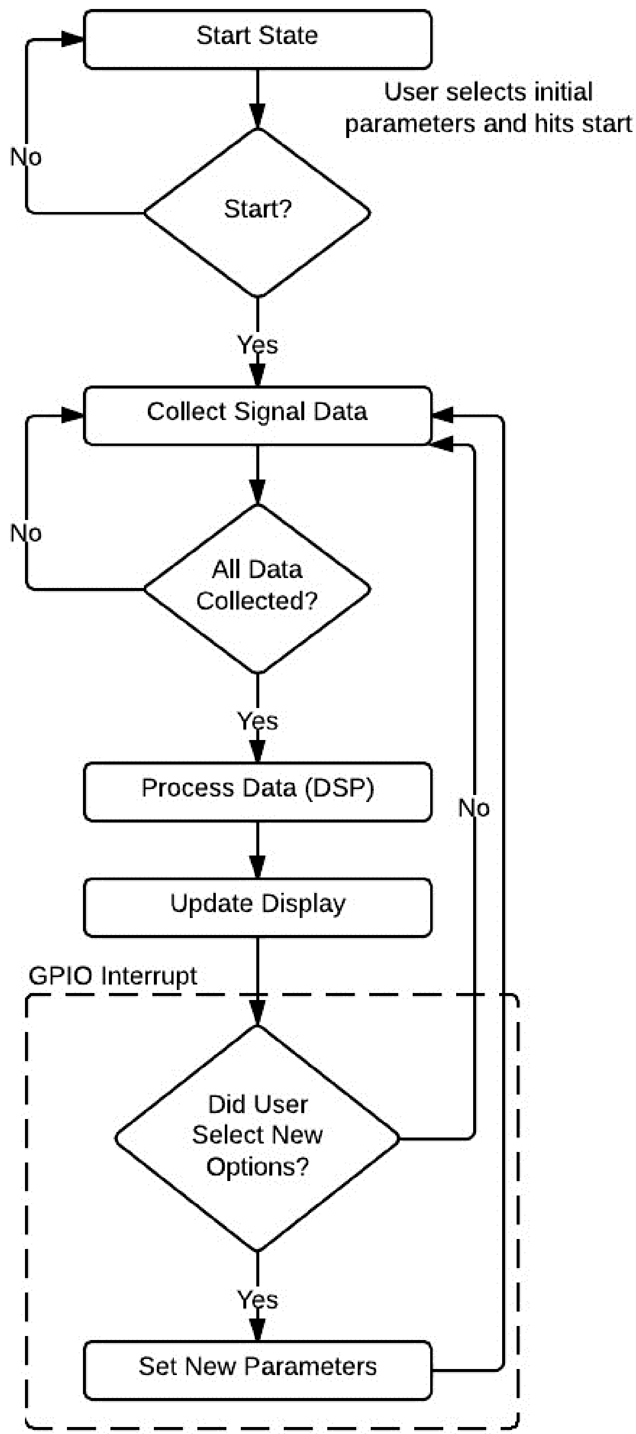


Figure 5: Software Flowchart

2.2 Block Diagram Details

2.2.1 Module Descriptions

Summary

The acoustic analyzer unit will consist of 3 main parts: power, MCU, and display. The overall system will take the audio from the microphones, process it, and show it to the user through the display. The user can make the system display the specific data that he/she wants and in the form he/she wants through manipulation of the user interface. If there are extra functions that the user needs that are not already preprogrammed, he/she can add them through the programmer.

Microcontroller

This module controls all the digital components of the system. It receives signal data from the ADC and inputs from the user via pushbuttons, processes the data, and then outputs it to the LCD. This module handles all the digital signal processing of the incoming audio signal data. It is also responsible for the GUI for the user and the users inputs. The sub-modules of this module are all parts of the software code.

1. **Data Collection:** The data collection sub-module handles acquiring the audio data from the two external ADC units. Once the sub-module has acquired all 4 sets of audio data (one from each microphone, two from each ADC), it passes the data to the DSP sub-module for processing.
2. **Digital Signal Processing (DSP):** The DSP sub-module handles all the processing of the audio data. The sub-module will initially only include the FFT of the audio data as part of the functionality. There will be functionality included to let the user add their own DSP functions to process the data. Once the data is processed, it will be passed to the display controller sub-module.
3. **Display Controller** This sub-module will update the LCD with the data obtained from the DSP sub-module. This module is also responsible for the GUI and manipulation of the GUI based on incoming inputs from the user input sub-module.

Power Module

This provides different voltages and power to the other modules: Analog circuit, ADC, Microcontroller, and LCD screen.

Analog-to-Digital Converter (ADC) Module

The Analog-to-Digital converter units are part of the input system. The ADCs convert the analog voltage signals received into a digital form for use by the microcontroller.

Analog

The audio block consists of the analog circuitry that will take the audio signal from the microphones to the MCU. The input will have to be able to take in a frequency range between 20Hz to around 10kHz. In order to do this a chunk of the circuit will be the anti-aliasing filter that will have an upper 3dB rolloff around 11kHz.

The audio block consists of the analog circuitry that will take the audio signal from the microphones to the MCU. The circuitry will include any preamplification, filtering, and buffering that's needed and will provide a usable signal for the MCU's ADC as well as protecting the Microphone from any electrical damage and vice versa.

User Interface

The user input sub-module takes the input signals from the external push-buttons and manipulates the DSP settings and GUI settings accordingly

Display

This module displays the processed audio data on a LCD screen for the user.

2.2.2 Power

Inputs: The input is the battery source, which can be recharged via wall outlet.

Outputs: The 3.7/4.2V battery voltage will be boosted to 5V (for part of the analog circuitry, the ADC, and the LCD). It will be stepped down to 3.3V for the Microcontroller and to 2.5V for part of the analog circuitry.

Description: The main source of power is from a 3.7/4.2V Li-polymer battery. This battery can be recharged via wall outlet. We need both lower and higher voltages that the battery provides. Below is a table explaining the voltages needed:

Table 1: Voltage Requirements

Module	Required Voltage
Analog Circuitry	2.5V & 5V
ADC	5V
Clock Chip	3.3V
Microcontroller	3.3V
LCD Screen	5V

The picture below is the device, from adafruit.com, used to recharge the battery. The wall adapter is plugged in via a 2.1mm plug and will charge the Li-polymer battery. It shuts off after 14 hours to not overcharge the battery.

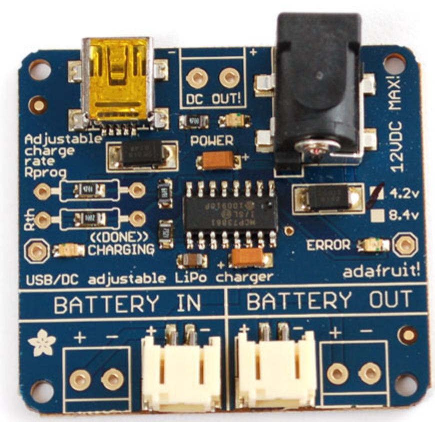


Figure 6: USB/DC Li-Polymer Battery Charger (Adafruit) [3]

From the Li-polymer battery, it needs to be boosted to 5V using the TPS61070. This 5V will be for the ADC as well as the LCD screen. Then, it needs to be stepped down to 3.3V and 2.5 using the LM317 (adjustable voltage regulator). The block diagram of the power module can be seen above in the Block Diagrams section.

Simulations and Calculations

1. LM317 Adjustable Voltage Regulator

   According to the data sheet from Texas Instruments for the LM317, the output voltage can be adjustable from 1.25V to 37V. For our purposes, I want to get a voltage of 3.3V (for the microcontroller) and 2.5V (to be used for the op amp as a reference).

   The capacitor Ci at the input of the LM317 is not recommended, however, a 0.1 F capacitor should be sufficient for bypassing. The capacitor C0 is not necessary either, but improves transient response. The value for this capacitor is 1.0 F. The capacitor CADJ is used for the prevention of a ripple rejection. There is an added 1N4002 diode for protection. Note that these values hold for both output voltages of 3.3V and 2.5V.

   To get the resistor values for the desired output voltages, we use this equation:

   $$V_0 = V_{ref}(1 + \tfrac{R_2}{R_1}) + I_{adj}R_2$$

   From the data sheet, normally Vref =1.25V and R1=240Ω . "Iadj" is typically 50 A, and can be negligible, but we used it in our calculations.

   $$3.3 = 1.25(1 + \tfrac{R_2}{240}) + 50\mu(R_2)$$
   $$\therefore R_2 = 390\Omega$$

   For an output voltage of 2.5V, the calculation is as follows:

   $$2.5 = 1.25(1 + \tfrac{R_2}{240}) + 50\mu(R_2)$$
   $$\therefore R_2 = 238\Omega$$

   Because we are using standard resistor values (±5%), we would put a 220Ω and 18Ω resistors in series to get the 238Ω needed. Below is a snippet from the TI data sheet.

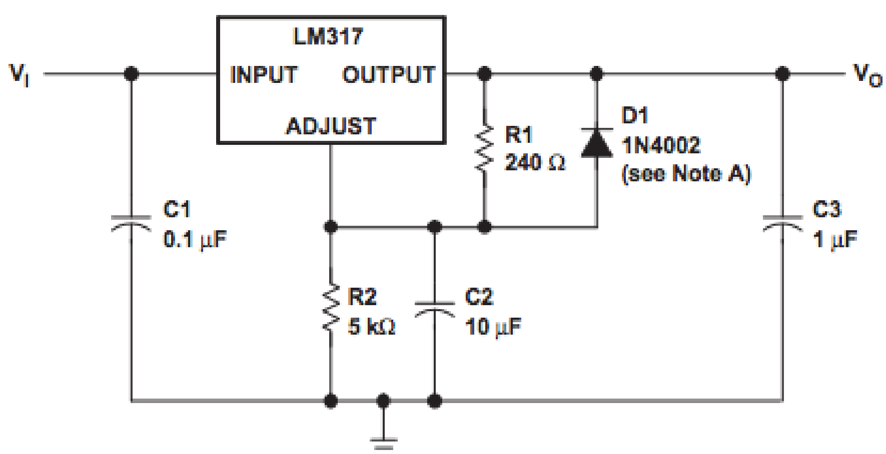


Figure 7: LM317 Typical Configuration [1]

2. TPS61070 Synchronous Boost Converter

According to the data sheet from Texas Instruments, this chip can have an adjustable output voltage up to 5.5V. Since we are using a single-cell Li-polymer battery (3.7/4.2V), we would need to this boost converter to get the desired 5V output. /par In the data sheet, there were detailed notes and calculations required to get the desired output voltage. The calculations for the resistors are below:

$$R_1 = R_2(\frac{V_0}{V_{fb}} - 1) = 180k\Omega(\frac{V_0}{500mV} - 1)$$
$$R_1 = 180k\Omega(\frac{5}{500mV} - 1)$$
$$\therefore R_1 = 1620k\Omega$$

Because we are using standard resistor values (5%), we would put a 1.6MΩ and 20kΩ resistors in series to get the needed resistance.

Next, we needed to find the inductor value. Calculations for the current through the inductor, as well as the inductor value is below:

$$I_L = I_0(\frac{V_{out}}{V_{hat}*0.8})$$

The desired Io is 200mA for the LCD screen (the module that requires a 5V input).

$$I_L = 200mA(\frac{V_{out}}{V_{hat}*0.8})$$

In the below equation, Δ IL is the ripple current in the inductor and f is the switching frequency.

$$L = \frac{V_{bat}(V_{out}-V_{bat})}{\Delta_{IL}*f*V_{out}}$$
$$L = \frac{3.7(5-3.7)}{0.2*350k*5}$$
$$\therefore L = 13.74\mu H$$

Because we are using standard inductor values, we used a 10 uH inductor.

The final value needed is the value for the capacitor at the output. Equation is below:

$$C = \frac{I_0(V_{out}-V_{bat})}{\Delta V*f*V_{out}}$$

Where ”” V is the maximum allowed output voltage ripple, here to be taken as 10 mV.

$$C = \frac{200mA(5-3.7)}{10mV*350k*5}$$
$$\therefore C = 14.86\mu F$$

Because we are using standard capacitor values, we used a 10 uF capacitor.

Below is a snippet from the TI data sheet.

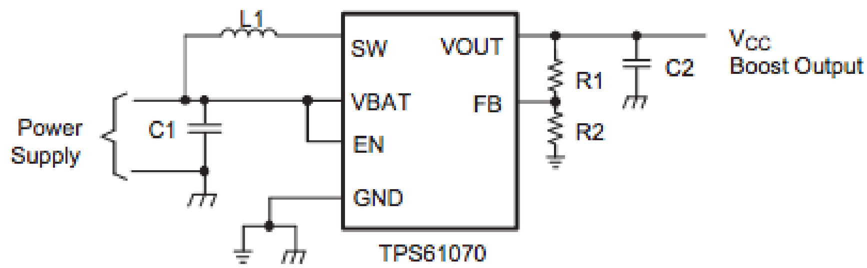


Figure 8: TPS61070 Typical Configuration [2]

2.2.3 Analog

Input The input of the analog block would be the microphone inputs. Our group is not responsible for the construction of the microphones; however, our device must be compatible with their existing systems. The only hard requirement they gave us for the input is that we should make the input connectors are BNC.

Output The block will output the analog filtered result of the signal to the ADC for conversion.

Description

The Analog Module has to accept frequencies between 20Hz to 10kHz. In order to do so, we are going to use a low pass filter with a decoupling cap. When designing the filter, we need to make sure that the dB output is stable within our operating frequencies as well as having a linear phase change.

In the end, we decided to go with a second order filter to have a steeper drop-off. Since we are working with a small frequency range, we feel that we need at least a second order filter to match our needs. Figure 9 shows the filter design.

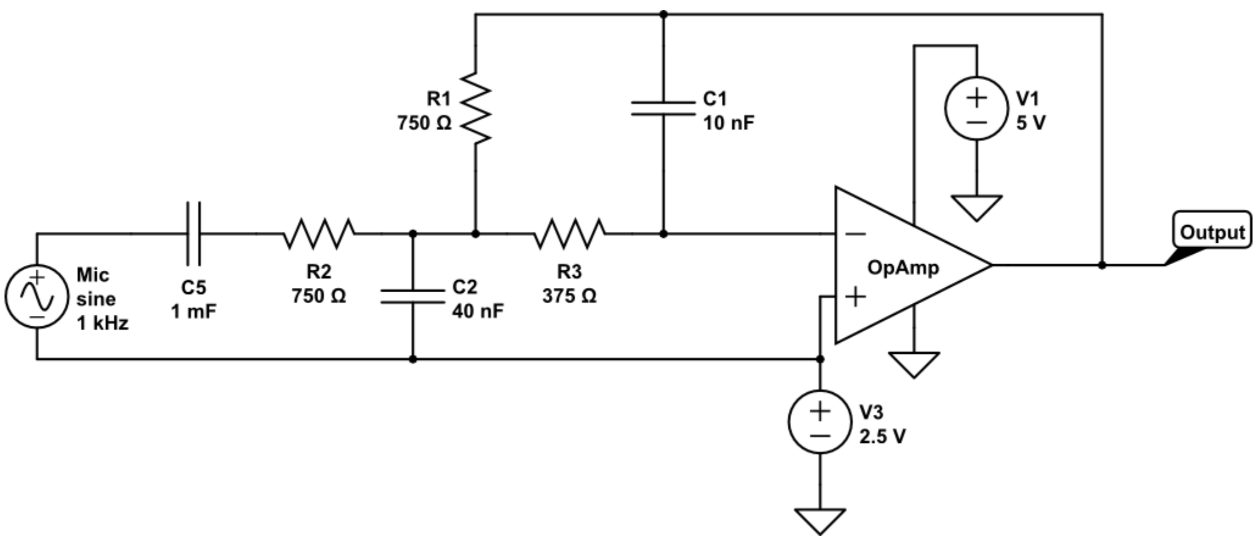


Figure 9: Input Circuit Sim Schematic

Even though the highest frequency we will be working with is 10kHz, we did not choose this as our upper cutoff. If 10kHz were selected as the upper cutoff, then at 10kHz the dB would have dropped around -3dB, theoretically. Knowing this problem can occur; we decided to choose a frequency that is higher - in our case it is 15kHz.

The following is the equation we have to solve to get the proper component values.

$$F_0 = \frac{1}{2*\sqrt{2*\pi*R*C}}$$

Even though we are solving for the product of R and C it was in our best interest to pre-select the values for the capacitor. It would be significantly harder to mix and match capacitor values to come up with the correct value. Resistors on the other hand have more flexibility and easier to work with. We chose to use a

.01uF capacitor which is a standard and abundant value.

$$15kHz = \frac{1}{2*\sqrt{2*\pi*R*.01\mu F}}$$

The selection of our OpAmp was determined by several factors such as the amount of noise, bandwidth, supply voltages, settling time, and also based on the applications notes of the datasheets. The OpAmp that was chosen is the OPA227PA that has the following specifications

- low noise @ 3nV/Hz
- wide bandwidth @ 8MHz
- settling time @ 5us
- open loop gain @ 160dB
- input bias current @ 10nA
- offset voltage max @ 75uV
- supply voltage @ ±2.5V to ±18V

With the following information above, we were able to run a simulation to get the results shown in figure 10.

We can see that the frequency response if pretty flat within our operating frequency range. Our phase plot is also pretty stable until we get to the much higher frequencies.

Pressure Microphone

The pressure microphone that will be used with the project has a sensitivity of 3.1 mV/p. Keeping this rating in mind, we can calculate the highest expected output for the microphone which in turn can tell us if any special protection circuit is required.

Referencing a dB chart with a wide range of values, the highest noise recorded was a firework explosion from 3 feet away. This caused a dB rating of 150dB, which is louder than standing beside a jet engine. When this dB rating is converted into Pascals, we get 632.456p. Using the sensitivity, we can get the expected voltage output, which is calculated below:

632.456p * 3.16mV/p = 1998.56mV

With an expected max output voltage of 2V, we know that this is well within our projects safe operating parameters.

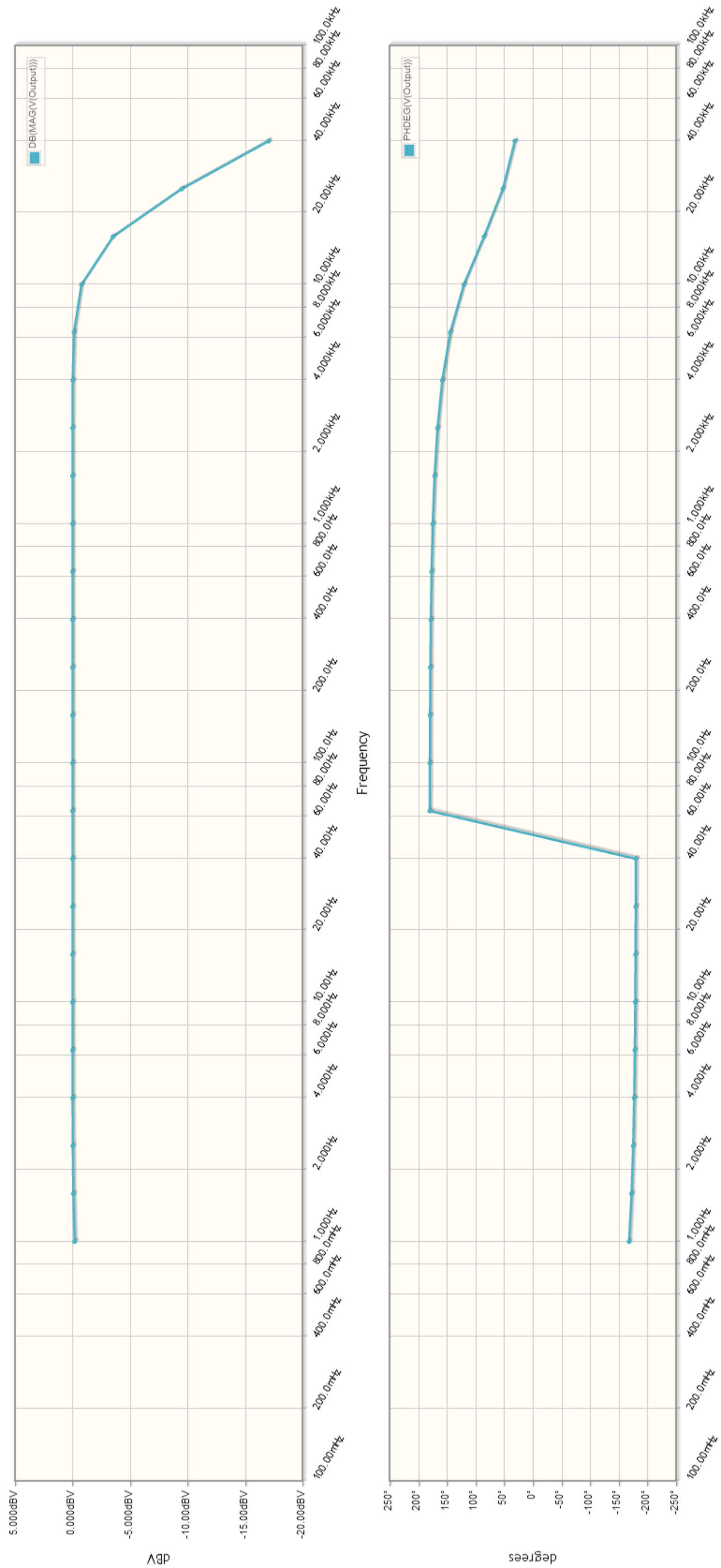


Figure 10: Analog Simulation Results

2.2.4 ADC

Inputs: Each individual ADC module will have two analog audio signals as inputs. They also require a master clock and reset signal that is common for synchronization.(insert details on signals voltage, single ended, etc.)

Outputs: The ADC outputs data through a serial connection to the microcontroller.

Description: This entity consists of the PCM1801 chip. The chip operates in I2S slave mode. The FMT pin on the chip needs to be set high for I2S format data. The timing diagram for the serial communication is shown in figure 11.

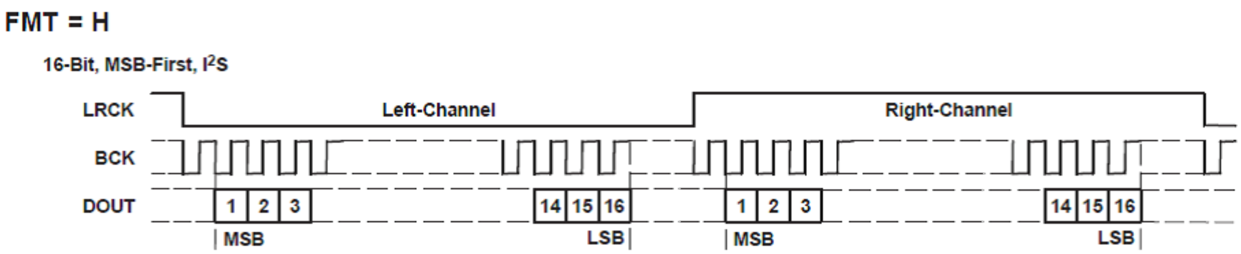


Figure 11: Slave Mode Serial Data Output Timing [6]

The approximite max voltage that we can expect to go into the ADC is around 2 volts. The ADC has 16 bits of resolution which means that it can distinguish up to 2̂16 different values for a total of 65536 datapoints. With a maximum input voltage about 5 volts we can expect 0.00007629394V per step which is more than enough accuracy for our purposes.

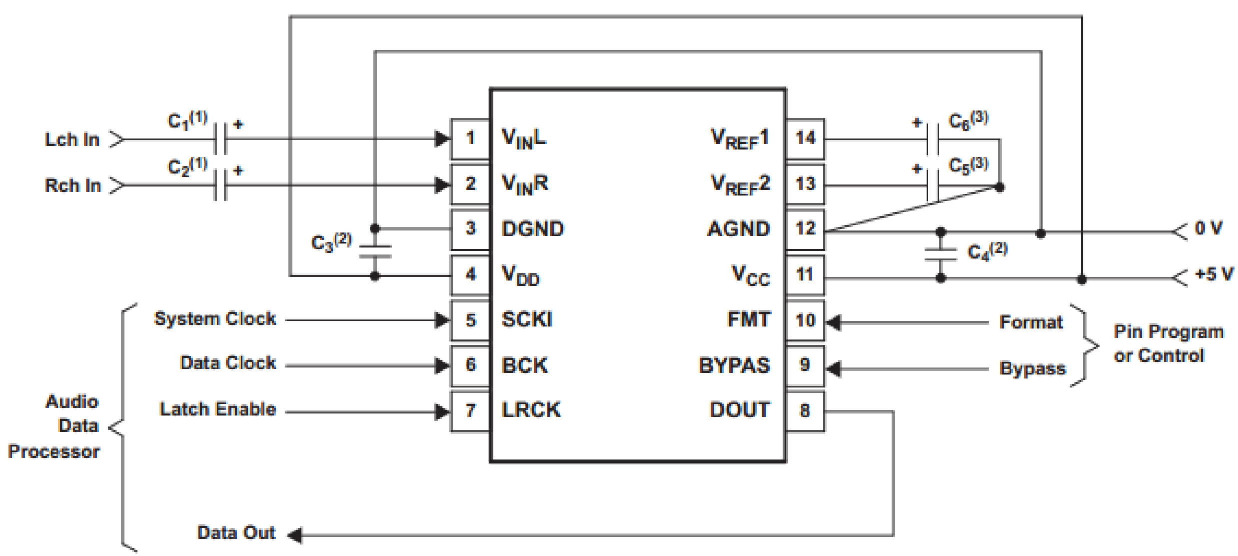
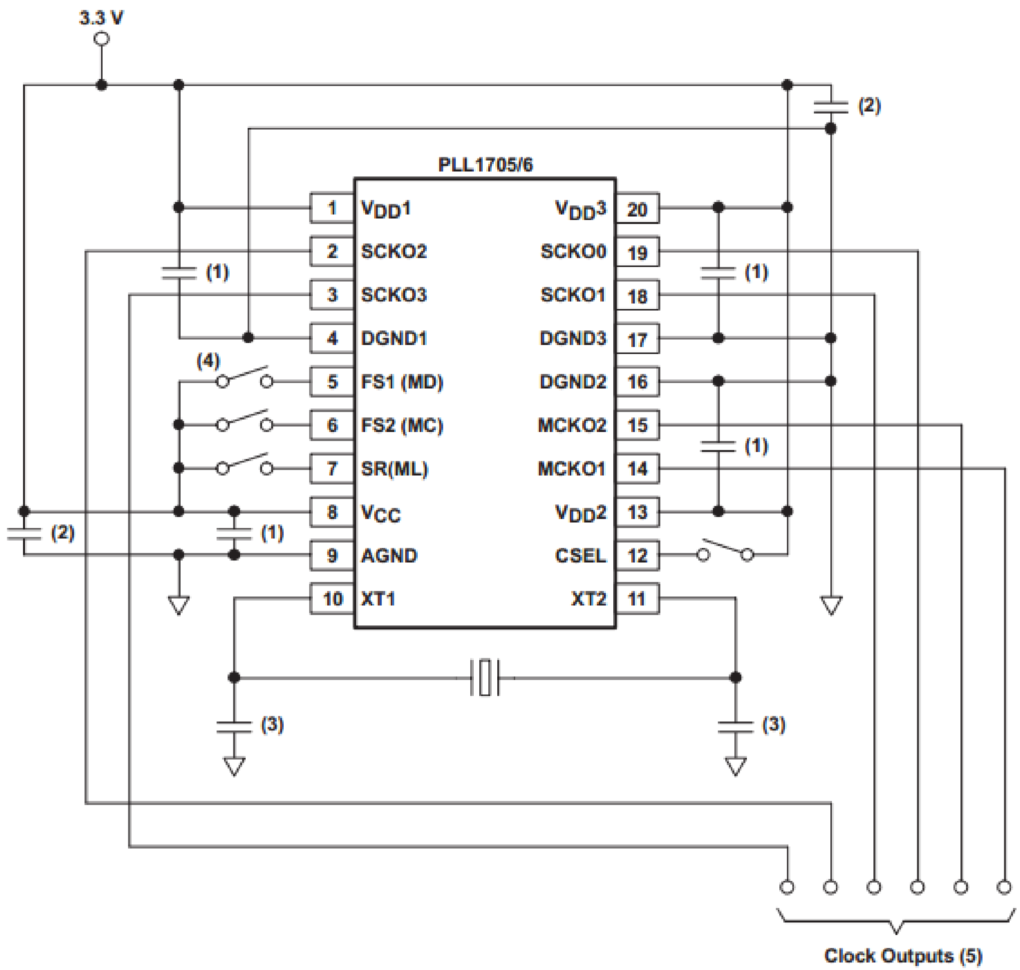


Figure 12: PCM1801 Typical Configuration [6]

Looking at the datasheet we need an input clock about 24.5760Mhz to sample at 48kHz. This is a very specific clock that we wont be able to produce with traditional means (oscillators and crystals) so we are also going to include a clock chip. The clock chip that was selected for the job is PLL1705 which is a IC that is designed to produce odd clock frequencies. The only input that this chip needs is a standard 27Mhz clock. Figure 13 shows an example of a typical setup.



(1) 0.1-µF ceramic capacitor typical, depending on quality of power supply and pattern layout
(2) 10-µF aluminum electrolytic capacitor typical, depending on quality of power supply and pattern layout
(3) 27-MHz quartz crystal and 10–33 pF × 2 ceramic capacitors, which generate the appropriate amplitude of oscillation on XT1/XT2
(4) This connection is for PLL1705 (parallel mode); when PLL1706 (serial mode) is to be used, control pins must be connected to serial interfaced controller.
(5) For good jitter performance, minimize the load capacitance on the clock output.

Figure 13: PLL1705 Typical Configuration [6]

2.2.5 Microcontroller

1. ***Data Collection***

   Inputs: The Data Collection sub-module has two inputs from each external ADC units. These are serial connections connected through the microcontrollers SSI interface. Since the ADC transmits data as per the I2S specification, the input requires two SSI interfaces per ADC. One of the SSI interfaces FSS output is connected to the I2S word clock to determine channel data as seen in figure 14.

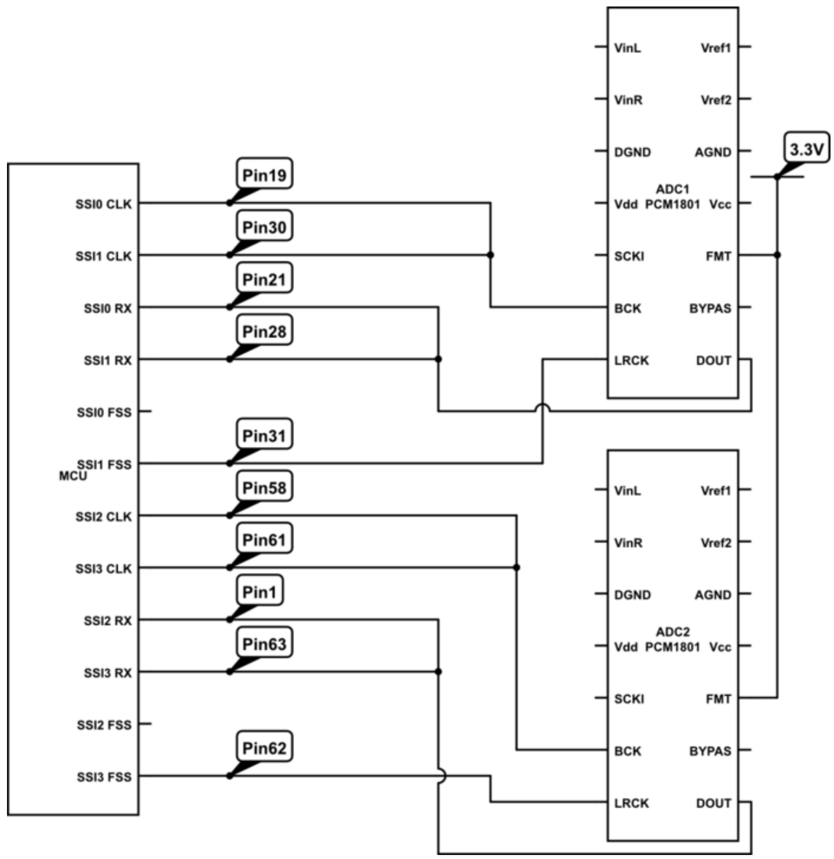


Figure 14: MCU to ADC connections

   Outputs: Data will be outputted in the form of a data structure since the connection to the DSP sub-module is internal within the software. Each microphones data will be stored in memory in a data structure which will act as the connection between sub-modules.

   Description: The Data Collection code will continuously acquire data from the external ADCs. Each microphones data will be stored in one channel of I2S data from the ADC. Data from the ADCs will be collected sequentially. Once data from all four microphones is collected, it will be stored in memory and then control will be passed to the DSP sub-module.

2. ***Digital Signal Processing***

   Inputs: Input data comes from the Data Collection sub-module and will be stored in memory where the DSP sub-module can access it. There will also be input from the user input sub-module in the form of changes to certain variables in memory that represent the parameters of any DSP functionality.

   Outputs: Output data will be stored in memory for the Display Controller sub-module to use. Each new DSP function implemented by the user will need its own data structure in memory to pass on to the Display Controller.

   Description: When control is passed to the DSP sub-module, the sub-module will extract the data and process it in each signal processing function set by the user in order. Initially, the DSP sub-module will only include processing the data via a FFT. After the audio data is processed and saved in respective data structures in memory, control will be passed to the Display Controller sub-module.

3. ***User Input***

   Inputs: Signals from pushbuttons connected to GPIO pins act as inputs for this sub-module. Some data from the Display Controller sub-module will also serve as inputs to this sub-module so user input from the pushbuttons may be processed in context to what is displayed to the user on the LCD screen.

   Outputs: Output data will be stored in memory for access by the DSP and Display Controller sub-modules.

   Description: The User Input sub-module will only gain control through GPIO based interrupts. When the user depresses a push-button, the signal on the GPIO pin will trigger an interrupt, the sub-module will change any parameters that have been changed and then update the display accordingly. The push-buttons will be connected to the microcontroller GPIO as shown in figure 15.

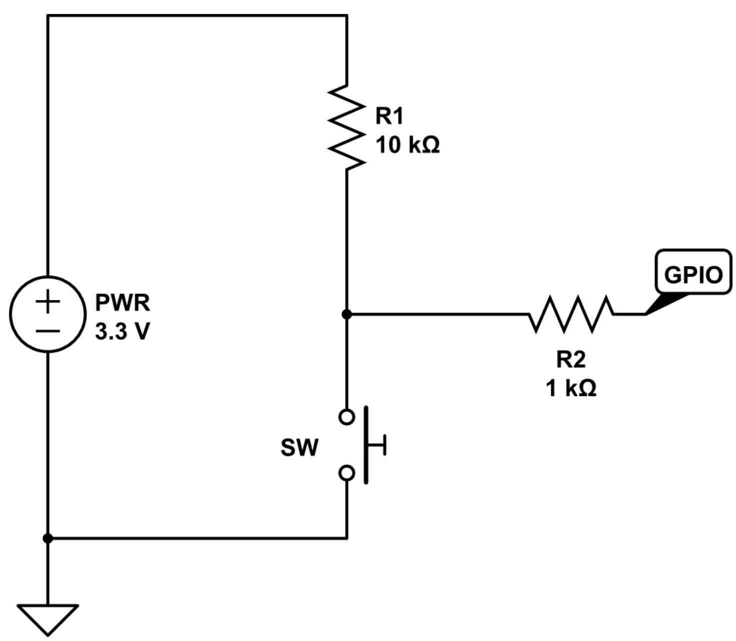


Figure 15: GPIO pushbutton circuit

4. ***Display Controller***

Inputs: The Display Controller will have inputs from the DSP sub-module in the form of processed audio data saved in memory. The User Input sub-module will directly control this sub-module during interrupts triggered by the User Input sub-module (such as changing menus etc.).

Outputs: The Display Controller sub-module will interface with an external LCD screen through an 8-bit parallel data connection. This interface will be through GPIO pins on the microcontroller.

Description: This sub-module is responsible for all the GUI processing and interfacing with the LCD screen. Texas Instruments Stellaris group provides a graphical library that involves three layers (driver, primitives, widgets). Normally the driver layer is specific to LCD screen, however the LCD screen chosen is officially supported by Stellaris with use by their LaunchPad microcontroller and therefore the driver code is written and provided. Therefore, the Display Controller module will only interface through graphical primitives and widgets. Graphical primitives allow for individual items to be displayed such as lines, basic shapes, and text. The widget layer allows for more complex items to be displayed such as graphs, buttons, and sliders.

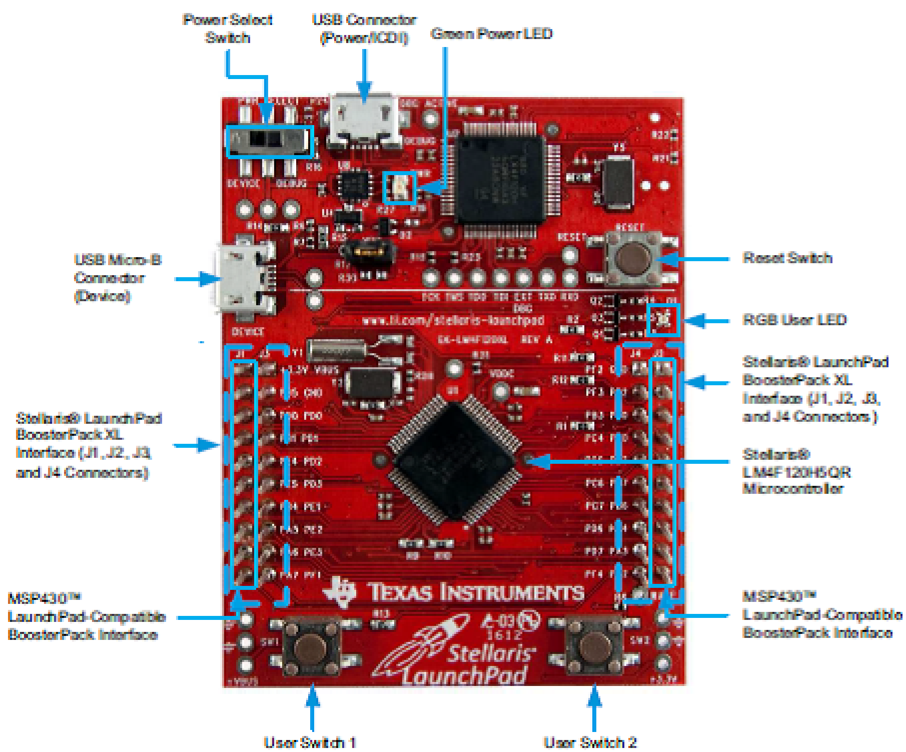


Figure 16: Stellaris LM4F120 LaunchPad Evaluation Board [4]

2.2.6 LCD

Inputs: The LCD module has an 8-bit parallel data input from the microcontroller to the LCD screen.

Outputs: The outputs of this module will be the items displayed on screen to the user.

Description: The LCD screen is a Kentec EB-LM4F120-L35 made specifically as a boosterpack for the Stellaris LaunchPad microcontroller used. The LCD is a 3.5 inch QVGA TFT LCD screen. It has support for 8-bit parallel interface or 3-wire and 4-wire SPI interface. The default driver code uses the 8-bit parallel interface so that is what will be used.

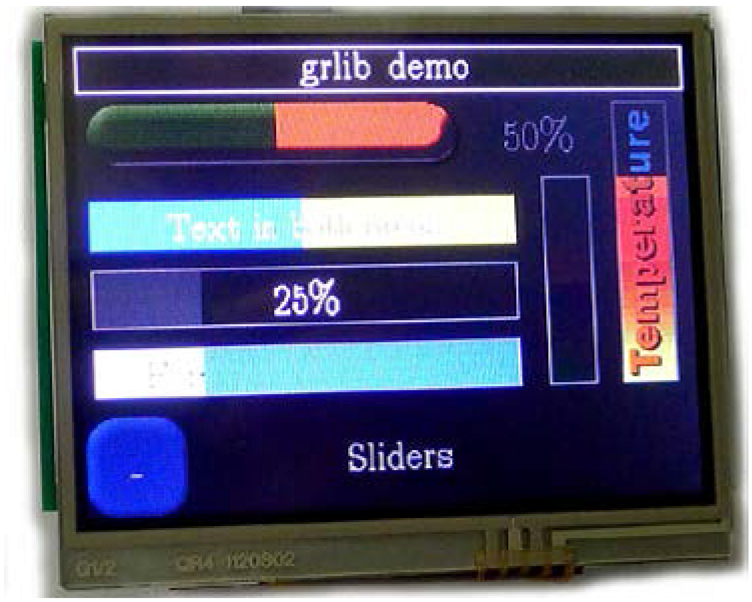


Figure 17: Kentec EB-LM4F120-L35 Stellaris LaunchPad boosterpack [5]

2.3 Additional Schematics and Pin Assignments

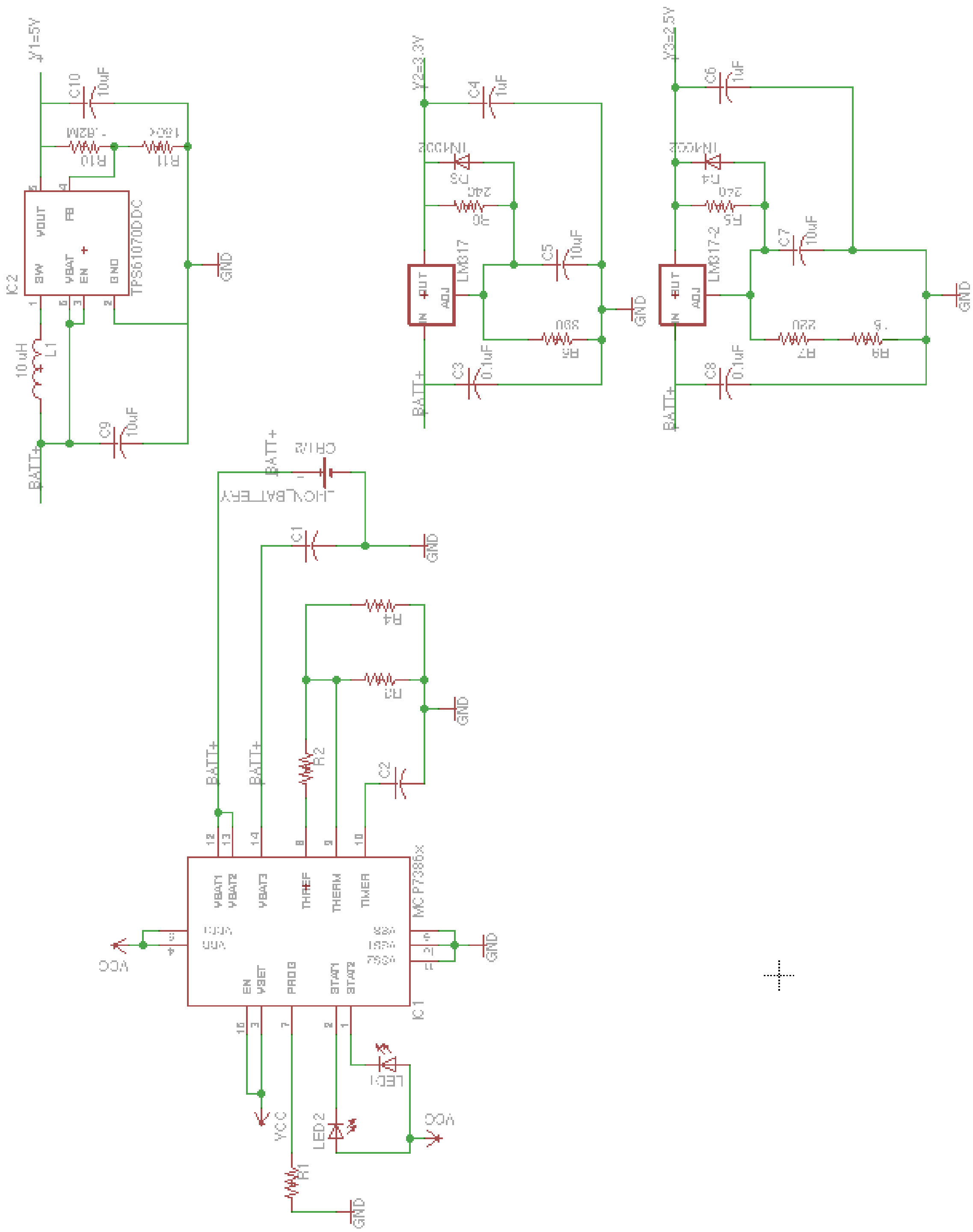


Figure 18: Power Supply Schematics

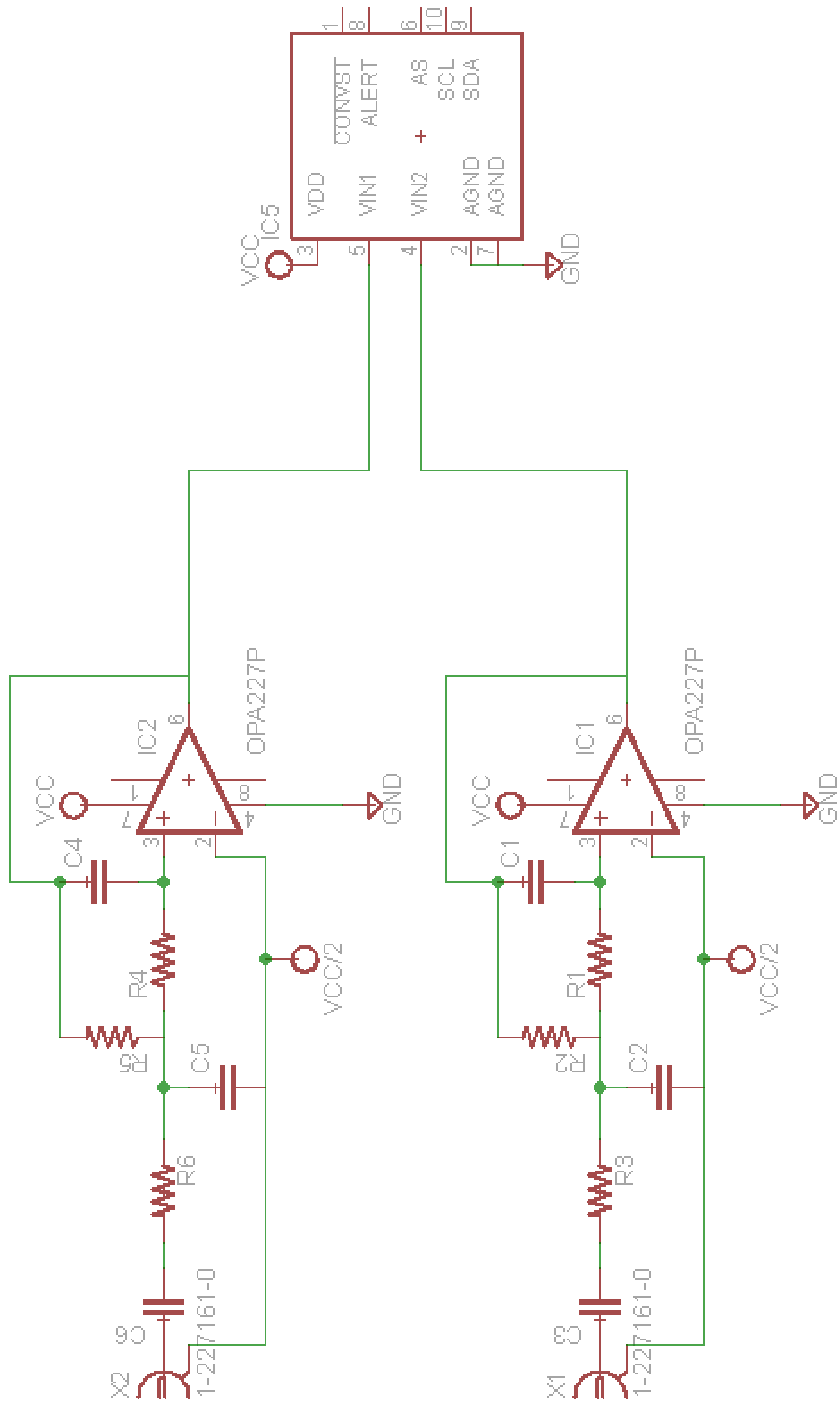


Figure 19: Analog Schematic 2 Input

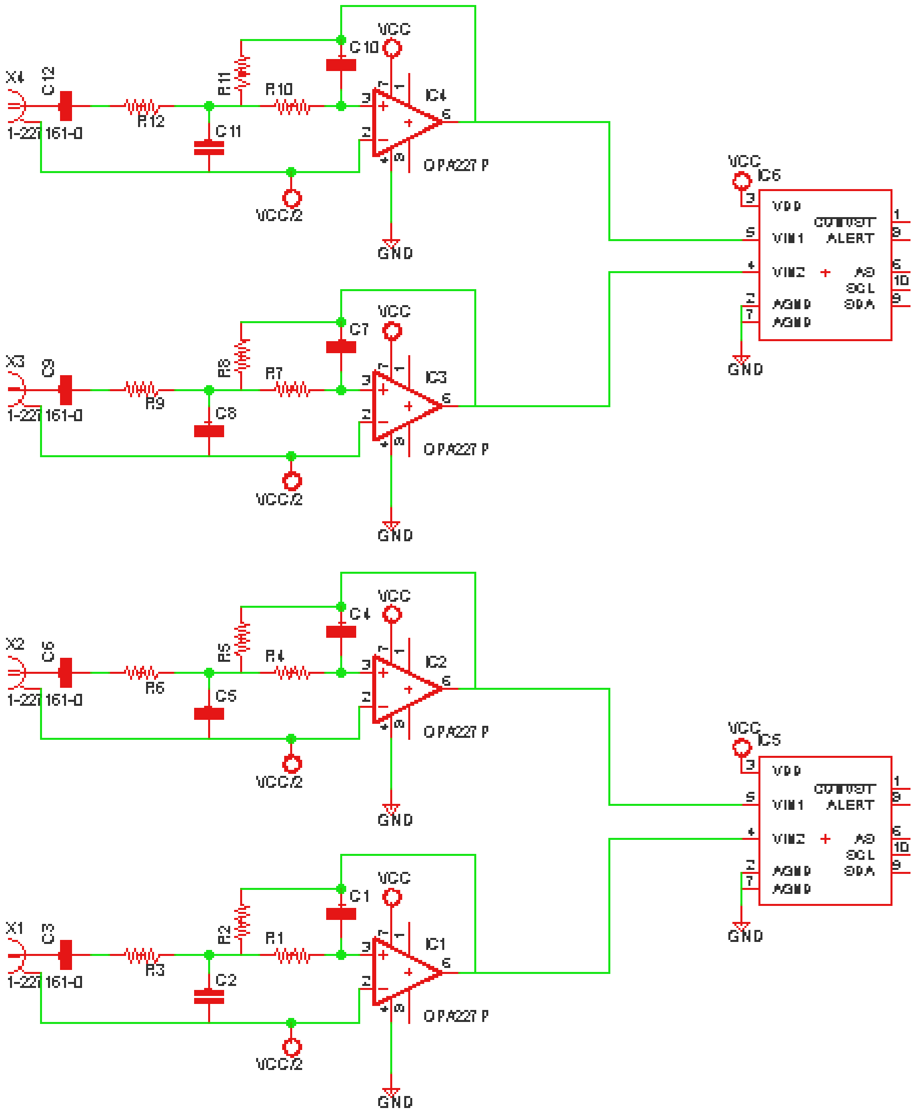
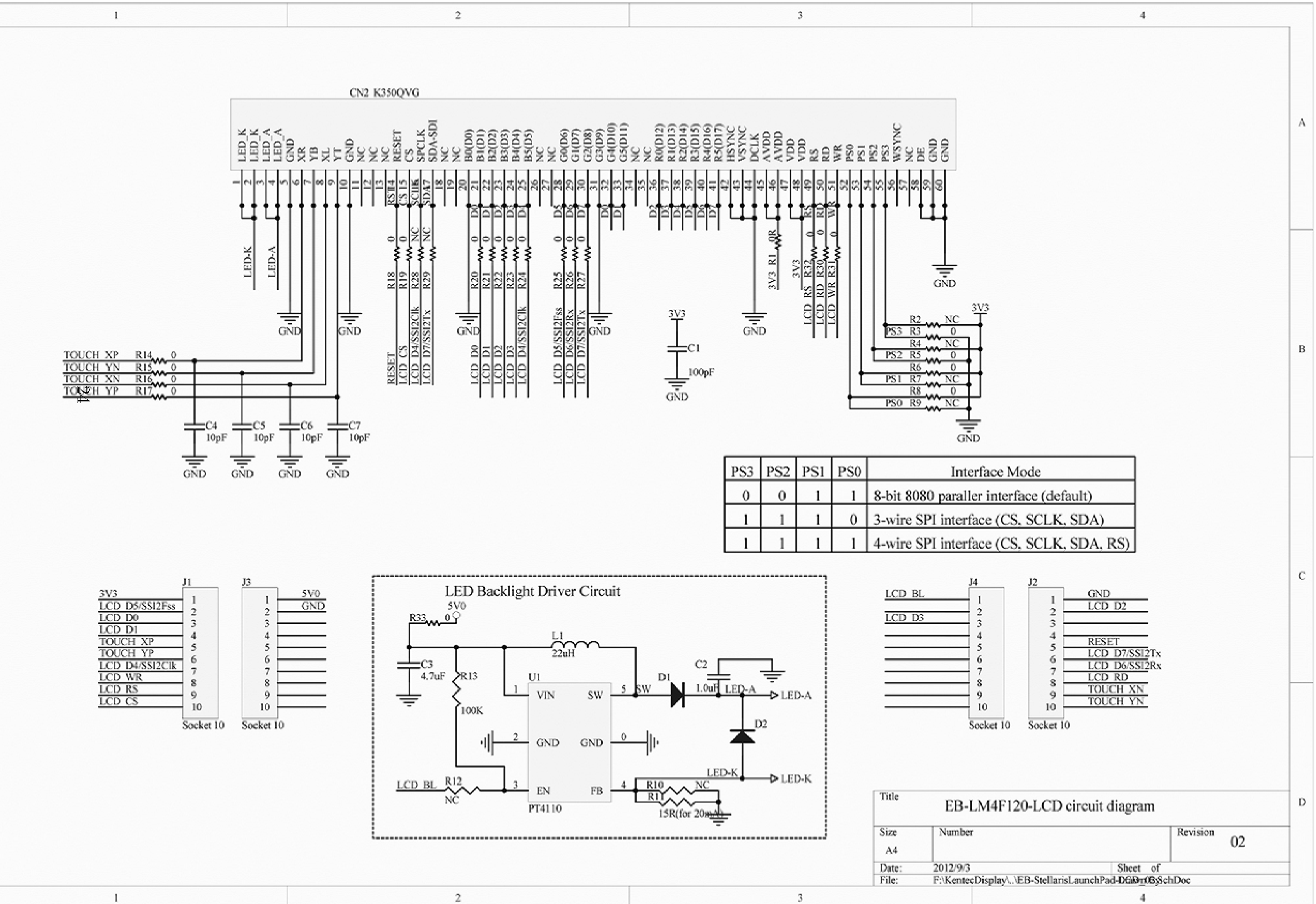


Figure 20: Analog Schematic 4 Input



PS3	PS2	PS1	PS0	Interface Mode
0	0	1	1	8-bit 8080 paraller interface (default)
1	1	1	0	3-wire SPI interface (CS, SCLK, SDA)
1	1	1	1	4-wire SPI interface (CS, SCLK, SDA, RS)

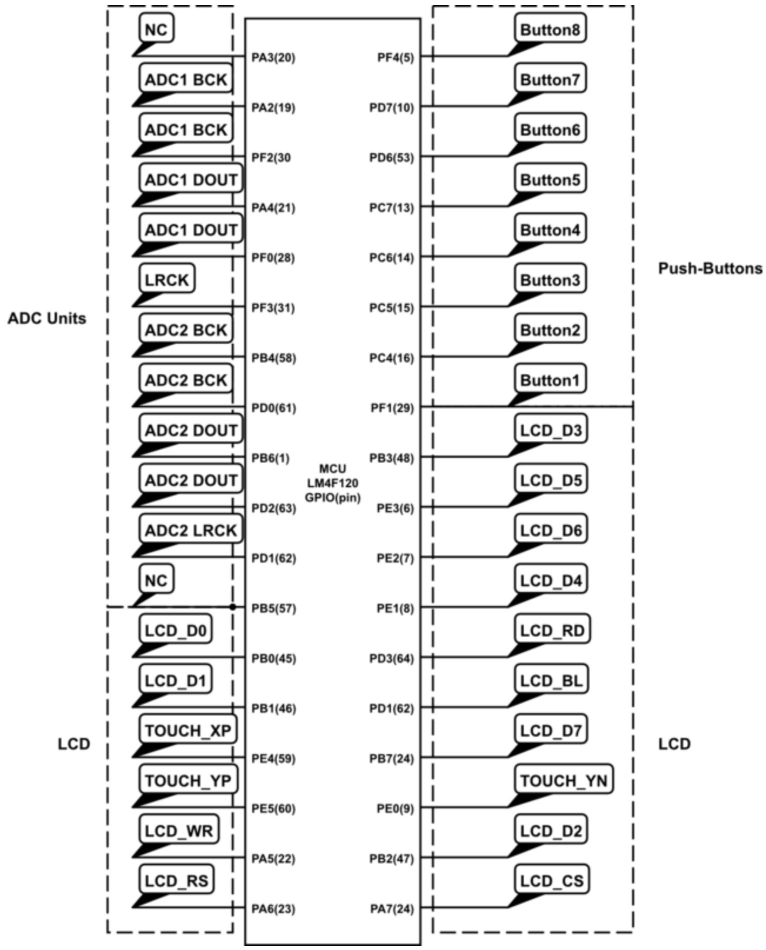


Figure 22: MCU Pin Assignment

2.4 Performance Requirements

1. *Power*

   The overall voltage of the power supply ideally should be 2.5, 3.3V, or 5V (but range could be 1V). The module should be able to provide a stable voltage that the other modules can use. In terms of efficiency the power supply should have an efficiency rating of over 50.

2. *Analog*

   This module should be able to take input from the microphone and send a usable signal to the MCU. This should prevent unwanted current/voltages from reaching other modules.

3. *ADC*

   This module needs to successfully digitize the input audio data and send it to the microcontroller without errors in I2S format.

4. *Microcontroller*

   This module should receive data from the ADCs without error, process the data, and then output it to the LCD module without error. It also needs to be able to accept user input and act accordingly depending on what pushbutton is pressed and what is being displayed on the LCD currently.

5. *User Interface*

   There should be ample room for all the buttons to sit on the actual device. The button layout should be easy to use and prevent multiple buttons from being pressed accidentally.

6. *LCD*

   This module needs to display requested processed audio data correctly and to the users inputted settings.

2.5 Testing Procedures

1. *Power*

   Requirements Summary This should provide enough/stable power and voltages to each of the other modules that require power.

   Verifications Summary Use multimeters to probe the circuit to see if the correct voltage is created for the different modules. We will also check that voltage/power over time for each module for stability purposes.

Requirements	Verifications
1) Check voltages for 5V, 3.3V, and 2.5V (Tolerance should be desired voltage $\pm$ 0.5V).	1) Use a multimeter to check the outputs of the boost converter (5V) and the two voltage regulators (3.3V, 2.5V).
2) Supply sufficient voltage each of the modules over time. (With a voltage ripple of $\pm$ 1.0 V).	2) Use an oscilloscope to see if there is steady voltage being given to individual module within the voltage ripple.
3) Check individuals modules if receiving correct voltage.	3) Check individual modules a. Connect all modules to power supply b. Verify if modules have the correct input voltage with a multimeter
4) The Charging Circuit i. The fully charged battery should be at maximum 4.2V ii. The battery should only at minimum 3.0V at full discharge. iii. With a charge rate of 1000mA, the battery (with rating of 2600mAH) should be fully charged in 2.6 hours ($\pm$ 20 min).	4) Charging Circuit i. Using a multimeter, check the voltage of battery in initial testing (max 4.2V). ii. Using a multimeter, check the voltage of battery after being connected to various loads (See Requirement (1)) and must be above 3.0V iii. Connect the battery to the wall outlet and check the amount of time it takes for the battery to be charged to a max 4.2V.

2. *Analog*

Requirements Summary: Needs to take in the correct data from the microphone probes and send this to the microcontroller unit.

Verifications Summary: Use an oscilloscope to takes measurements from the audio module. We will also run stress tests to see if voltages and current are within the tolerances of key components.

Requirements	**Verifications**
1) Check to see that the frequency response is similar to the simulation.	1) Check several frequencies and match with simulations a. Using a function generator and an oscilloscope check the filter output. b. Measurements should start at 20Hz, 100Hz, 500Hz 1kHz then repeat every octave. c. Conditions of sucess are shown below - If datapoints deviates less than ±1dB from simulations - If datapoints between 20Hz and 10kHz do not deviate more than 0.2 dB from each other
2) Check to see that the filter gain is similar to the input with a small passband ripple.	2) Input a 5kHz signal at 1Vpp and check the output. The output should also be roughly 1V ± 50mV. The passband ripple should be less than .1Vpp Afterwards input a 15kHz signal and check the output. The output should be roughly .6v ± 50mV.

3. *Microcontroller Unit*

Requirements Summary: This module should receive data from the ADCs without error, process the data, and then output it to the LCD module without error. It also needs to be able to accept user input and act accordingly depending on what pushbutton is pressed and what is being displayed on the LCD currently.

Verifications Summary: We will check for valid connections from each of the various modules to the microcontroller. Afterwards, we will use a simple program known to be bug free with audio input and check for expected outcome. Further tests will include ability to take inputs from the User Interface as well as properly displaying text from the MCU.

Requirements	**Verifications**
Data Collection: 1) Proper ADC I2C to dual SPI Communication	Data Collection: 1) Ensure ADC I2C to dual SPI Communication a. Connect Vref1 to AGND. b. Connect Vref2 to AGND. c. Connect VinL to AGND. d. Connect VinR to Vcc. e. Run Data Collection program once saving the left and right channel data. f. Through the onboard ICDI and code composer studio view the data Verify left channel data to be 0x0000 and right channel data to be 0xFFFF.
Digital Signal Processing: 2) FFT processes data correctly.	Digital Signal Processing: 2) For a predetermined set of audio data (constant tone of known frequency), verify FFT processing: a. Insert 2048 samples data set of a 5kHz sine wave into memory b. Run FFT code. c. Extract output data set to computer using ICDI and code composer studios manually. d. Plot data in Matlab and verify a frequency spike at 5kHz $\pm$ 100Hz.

User Input: 3) Push-button depress triggers GPIO	User Input: 3) Verify depressing a push-button causes a GPIO interrupt: a. Add a line of code into interrupt code to toggle onboard LED. b. Run code. c. Depress pushbuttons one by one and verify LED status.
4) Push-buttons change parameters (current screen shown, DSP parameters) depending on current parameters.	4) Verify push-buttons change the proper parameters based on current parameters: a. Manually change the default parameters b. Run User Input program c. Depress each push button d. Check parameters afterwards using ICDI and code composer to check if the correct parameters changed to the correct values. e. Repeat as necessary.
Display Controller: 5) Data can be shown correctly on screen.	Display Controller: 5) Verify Data stored is shown on screen correctly (LCD screen must be verified first): a. Manually change processed data of FFT to a progressive linear increase of values from 20Hz to 10kHz. b. Run display controller code with LCD boosterpack attached. c. Verify screen shows an increasing histogram type bar graph starting at 20Hz to 10kHz.

4. *ADC Module*

   Requirements Summary: This module needs to successfully digitize the input audio data and send it to the microcontroller without errors in I2S format.

   Verifications Summary: Given a certain input to the ADC, we can take the data received by the microcontroller and check if the data received matches the expected output for the given input.

Requirements	Verifications
PCM1801: 1) ADC chip is powered correctly with appropriate parameters set. 2) ADC output signal is I2S format	PCM1801: 1) Ensure device is powered and output parameters are set: a. Connect multimeter probe GND to ADC AGND. b.Attach multimeter positive probe to ADC VCC. Voltage should be +5V nominal. Voltage should not be below 4.5V. Voltage should not exceed 5.5V. c. Connect oscilloscope probe GND to ADC DGND. d.Attach multimeter positive probe to ADC VDD. Voltage should be +5V nominal. Voltage should not be below 4.5V. Voltage should not exceed 5.5V. 2) Ensure proper I2S output format: a. Connect VREF1 to AGND. b. Connect VREF2 to AGND. c. Connect VINL to AGND. d. Connect VINR to VCC. e. Connect logic analyzer GND to ADC to ADC DGND. f. Connect logic analyzer probe to ADC DOUT. g. Connect function generator GND to ADC DGND. h. Connect function generator output to ADC BCK. i. Set function generator to generate a 3MHz square wave with amplitude of 3.3V. j. Connect ADC LRCK to ADC DGND. The logic analyzer should show 16 bits of audio data representing left channel which should be 0x0000. k. Connect ADC LRCK to 3.3V. The logic analyzer should show 16 bits of audio data representing right channel which should be 0xFFFF.

5. *User Interface*

Requirements Summary: Needs to be responsive to depressing of buttons on the device and have the correct functions.

Verifications Summary: To test this section of the device, we would go through various menus and functions by depressing the respective buttons. If the correct operation is occurring on the screen, then the buttons seem to be working with the microcontroller. We will also test if one button executes one operation or if more than one functions occurs.

Requirements	**Verifications**
1) Depressed buttons show correct operations.	1) Use a multimeter to check the voltages of each button. Check if high (depressed) or low (not depressed)

6. *LCD*

Requirements Summary: The LCD screen needs to clearly show the menus and data from audio. It needs to be able to show fluid changes of text/graphs on screen.

Verifications Summary: The LCD module can be verified using the graphical library example program made by Stellaris to showcase the graphical library. If the example program runs and the expected graphical elements are shown on screen, then the LCD Module works.

Requirements	**Verifications**
1) LCD powers on and displays properly	1) Verify LCD screen operation with LaunchPad evaluation kit: a. Plug in LCD screen boosterpack to Launchpad evaluation kit b. Upload and run graphical library example code c. Verify operation of graphical library example program as shown in figure 17.

2.6 Tolerance Analysis

The most critical portion of this project is the method of converting all input signals into usable, distinguishable data to show on the display screen. Thus, the actual programming portion in tandem with the microcontroller unit will be the most work intense. The goal of this tolerance analysis is to verify that the program is correctly defining each audio input and has the correct calculations/algorithms to show acoustical data. Since we are also performing an FFT on the data, this needs a different calculation that needs verification. The original unit from the offices of Professor Swenson and Dr. White shows the data they want to be shown and analyzed, we will use this as a basis for our unit (since we are basically creating a miniature version of this). The acceptable tolerance will be a 10% error of the acoustic intensity shown from what is acceptable by Professor Swenson and Dr. White.

2.7 Ethical Issues

As with all engineering projects we have to make sure that we are properly following the proper ethical guidelines. In order to check that we do not have any Ethical Issues Associated with out project we will go through the IEEE code of ethics for any potential breach in ethics.

1. *to accept responsibility in making decisions consistent with the safety, health, and welfare of the public, and to disclose promptly factors that might endanger the public or the environment*

   Initially we were worried about what the implications of our project would be if used by an army research facility; but, after some discussions with the potential end users, we understood the basis of Professor Swensons and Dr. Whites research. In general, the army generates a lot of noise and they are researching solutions to minimize the armys footprint in their area of operation. This research actually benefits the public because noise pollution and hearing damage is a daily challenge for some people. Since our device would potentially assist them in identifying and minimizing noise sources, we do not violate the first code.

2. *to avoid real or perceived conflicts of interest whenever possible, and to disclose them to affected parties when they do exist.*

   Working with the US army research lab could bring up several conflicts of interests that is based on nationality. However, that conflict of interest is minimized since all three group members are indeed US citizens. To add to that, this particular research base does not deal with any weapons of war since most of the research deals with base functionality, such as structure and acoustical research.

3. *to be honest and realistic in stating claims or estimates based on available data.*

   Our group has been transparent with the research group so they understand our capabilities, as well as our projected product. All information about our project should be available in this Design Review report as well.

4. *to reject bribery in all its forms.*

   Since this is a purely academic project, the risk of any of our group members being bribed is minimal. To ensure that this still does not occur, our group will maintain transparency regarding the project and keep each other in check.

5. *to improve the understanding of technology; its appropriate application, and potential consequences.*

   Before designing our project, we looked into all potential applications and consequences that might be associated with this. Since this is going to be research equipment, we believe that the user will use our product appropriately and responsibly.

6. *to maintain and improve our technical competence and to undertake technological tasks for others only if qualified by training or experience, or after full disclosure of pertinent limitations.* We made sure to split up our work according to our previous experiences and have consulted with others that have similar, if not higher technical background than ourselves.

7. *to seek, accept, and offer honest criticism of technical work, to acknowledge and correct errors, and to credit properly the contributions of others.* All group members have grown up in a diverse environment and will treat each other as well as the rest of our colleagues, fairly.

8. *to treat fairly all persons regardless of such factors as race, religion, gender, disability, age, or national origin.* All of our group members are respectable and will respect the property of others as we expect the same courtesy from them.

9. *to avoid injuring others, their property, reputation, or employment by false or malicious action.* All of our group members are respectable and will respect the property of others as we expect the same courtesy from them.

10. *to assist colleagues and co-workers in their professional development and to support them in following this code of ethics.* While going over the ethical rules and guidelines, we have read and proofed each others work and therefore understand the IEEE code of Ethics. We will keep each other, as well as the rest of our peers, in check as we continue onward with our project.

2.8 Safety Issues

The biggest safety issue we have for our project is our use of a LiPo battery. LiPo batteries are one of the more unstable batteries in existence and has been known to heat up rapidly and catch fire if not used properly. In order to avoid this problem we have bought a pre-made battery management system board which has been tested and used by various individuals and is known to work properly. By having this intermediate module within our power module we will guarantee that our battery will not short of ignite due to human errors.

3 Cost and Schedule

3.1 Cost Analysis

Table 2: Labor

Member	Hourly rate	Total hours	Subtotal x 2.5
Kristine Cabrera	$35	130	$11375
Kevin Chen	$35	130	$11375
Joseph Shim	$35	130	$11375
Total			$34125

Table 3: Parts

Parts	Quantity	Cost per Unit	Total	Order Status
Stellaris LM4F120	1	$12.99	$12.99	Ordered
KS0108 LCD display	2	$35.00	$70.00	Ordered
5V/1A Wall Adapter	1	$5.95	$ 5.95	Ordered
USB/DC LiPo Battery Charger	1	$17.95	$17.95	Ordered
Polymer Li-Ion Bat 2600 mAh	1	$11.95	$11.95	Ordered
USB 2.0 Cable	1	$0.85	$0.85	Have
Voltage Regulator	2	$1.95	$3.90	ECE Shop
LM317				
Synchronous Boost Converter - TPS61070	1	$4.95	$4.95	Ordered
PCM1801 ADC	3	$6.97	$20.91	Ordered
PLL1705 Clock Chip	2	$4.03	$8.06	Ordered
OPA227PA Op-Amp	5	$0.643	$3.22	Ordered
PCB	2	$0.00	$0.00	Need to Design
LEDs	10	$0.262	$2.62	ECE Store
RCL Components	N/A	$0.00	$0.00	ECE Service Shop
LiPo Battery	1	$22.50	$22.50	Ordered
Mic Probe	1	$0.00	$0.00	At CERL
BNC Connectors	3	$5.57	$16.71	ECE Store
Pushbuttons	10	$3.06	$30.60	ECE Service Shop
Total			$232.31	

Table 4: Grand Total

Labor	Parts	Total
$34125	$232.31	$34357.31

3.2 Timeline

Table 5: Schedule

Week	Task	Member Assigned
2/04	Writeup for explanations and testing	Kristine
	Plan schedule and create parts list	Kevin
	Block diagram and LaTeX coding	Joseph
2/11	Rough Design for Power	Kristine
	Rough Systems Design + Program	Kevin
	Rough Design for Analog Input	Joseph
2/18	Finalize, Prototype, and Test Power	Kristine
	StellarisWare Operation Proof of Concept	Kevin
	Finalize, Prototype, and Test Analog Input	Joseph
2/25	Prototype Interface Design	, Kristine
	Program DSP functions	Kevin
	Prototype Interface Hardware	Joseph
3/04	Design User Interface	Kristine
	Setup LCD Software	Kevin
	Make PCB	Joseph
3/11	Finalize User Interface	Kristine
	Program GUI for Application	Kevin
	Test and Assemble PCB	Joseph
3/18	Finalize Hardware Modules	Kristine
	Finalize Software	Kevin
	Finalize Hardware Compatibility and Interconnects	Joseph
3/25	Additional Functionality	Kristine

	Iron out software	Kevin
	Iron out usability issues	Joseph
4/01	FlexTime*	Kristine, Kevin, Joseph
4/08	FlexTime*	Kristine, Kevin, Joseph
4/15	FlexTime*	Kristine, Kevin, Joseph

4 Contingency Plan

In the event that something occurs that is unplanned or unforeseen, we have made a contingency plan. In other words, we created a Plan B.

- If programming/ planning for up to four different microphone probe signals does not pan out, we will design for only a two-microphone probe.
- If designing the device to be compatible with the other groups microphone (Group #48) seems not plausible, then we will at least make sure that our device is compatible with the two microphone probe in the possession of Dr. Swenson and Dr. White.
- If the power circuit does not perform well with the recharging circuit, then we will put a Li-polymer battery that can be easily replaced (since no recharging of the same battery).
- We will do a lower programming development if it turns out too messy/ not intuitive.
- Remove GUI and work on single application unit (will do a single, simple function with no user input, except the microphone probe).

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