

Silent Rave: The Ultimate Aural Experience

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1. Introduction

Headphones allow a user to enjoy music while keeping the sound from disrupting others in surrounding areas. However, typical headphones come with some trade-offs. Some environments are constantly changing or the user is entering and exiting different environments, changing the aural inputs that one is receiving.

Depending on the scenario, the user is presented with different issues. For example, when running, it is not uncommon to enter and/or leave areas of increased noise intensity. When the user is running, however, it is difficult to adjust the volume of the music simultaneously. The runner either must stop to adjust the music or continue running with a disrupted music experience.

Another scenario may occur when another individual may approach the user listening to music. In this situation, due to the headphones, the user may not realize that he is being spoken to, which leads to a scenario in which the other individual is ignored.

A third scenario may occur when the user wants to be able to listen to his music and receive environmental input at the same time.

Many alternative headphones have been developed in order to address one of the issues presented. For example, the Shure Push-to-Hear Modules allow the user to start and stop the music from an additional push switch without removing the headphones. Unfortunately, this product still requires the user to press a button. Other alternatives currently on the market also currently only answer to one of the three scenarios.

SilentRave proposes a solution that addresses the unique problems of all three situations by introducing three modes. By adding microphones to the outside of the earbuds, the device takes in the external sound input and adjusts the user's aural experience according to his selected mode. As long as the user pre-selects the correct mode, no external input, such as pressing a button, is required!

The final device met almost all requirements from a hardware perspective. By integrating the board and its associated parts, microphones, headphones, and music player, SilentRave's device is able to adjust the digital signal processing for each mode; hence, a user has a solution to all three tricky scenarios described! However, the device the device did not fulfill an additional feature: alert the user when the battery was low. Further, although the device could be powered from an external source, it was unable to from a USB. The reasons for and solution to a USB connection are described in the future enhancements section.

Moreover, although most functional requirements were satisfied, the aesthetic outcome of the device's case was not ideal; the size and weight of the case was slightly larger than expected. Additionally, noise was present in the audio; although the presence of noise doesn't invalidate any system requirements, this is an outcome that affected the user's experience. Like every product on the market, there is definitely room for improvement

in the design. These improvements are further described in the future enhancements section.

2. System Requirements:

In all cases where the device is mixing in sound picked up by the microphones, the device will:

- Process the external sound through a stereo enhancement algorithm to simulate natural stereo imaging.
- Compress and limit the external sound to prevent overloading and protect your ears (e.g. if someone yelled into one of the microphones).
- Equalize the external sound to seem natural and comfortable through your headphones.

Additional features:

- Alert the user when battery power is low.
- Allow the music to be played through headphones when the device is off (or in the case of device failure).

The functional system will demonstrate three modes, each with different features:

- **Running Mode (Total Isolation):** The device will play music normally and employ noise cancellation until it senses that the environmental noise has increased past a threshold at which noise cancellation is ineffective. At this point, the device will increase the volume of your music (with dynamic compression to prevent distortion and clipping) to compensate for the increased ambient noise.
- **Conversation Mode:** The device will play music normally until a voice is detected. Upon this external input, the device lowers the volume of the music and mixes in the sound picked up by the microphones. When the device no longer detects external input, the music is restored to its normal level.
- **Ambient Mode (Safe Running mode):** The device will automatically mix in environmental sound with the user's audio.

Although not considered a mode, in bypass 'mode', the music will be unaltered by the device. This allows the user to still listen to his iPod when the device has run out of battery.

The device structure:

- The device is integrated into an iPod case. An external case houses the board, charging circuitry, and necessary devices. The case has openings that allow the microphone and iPod input to be sent to the board and the processed music to be sent back out to the user's headphones.
- Since the device is a separate unit, no installation process is required. The customer can easily connect the device with his iPod, and the device is ready to be used as if he were using unmodified headphones.

Power of Device:

- The device will not draw power from the iPod and instead be externally powered by a battery.
- The battery should provide a steady supply of current and voltage to all subsystems.

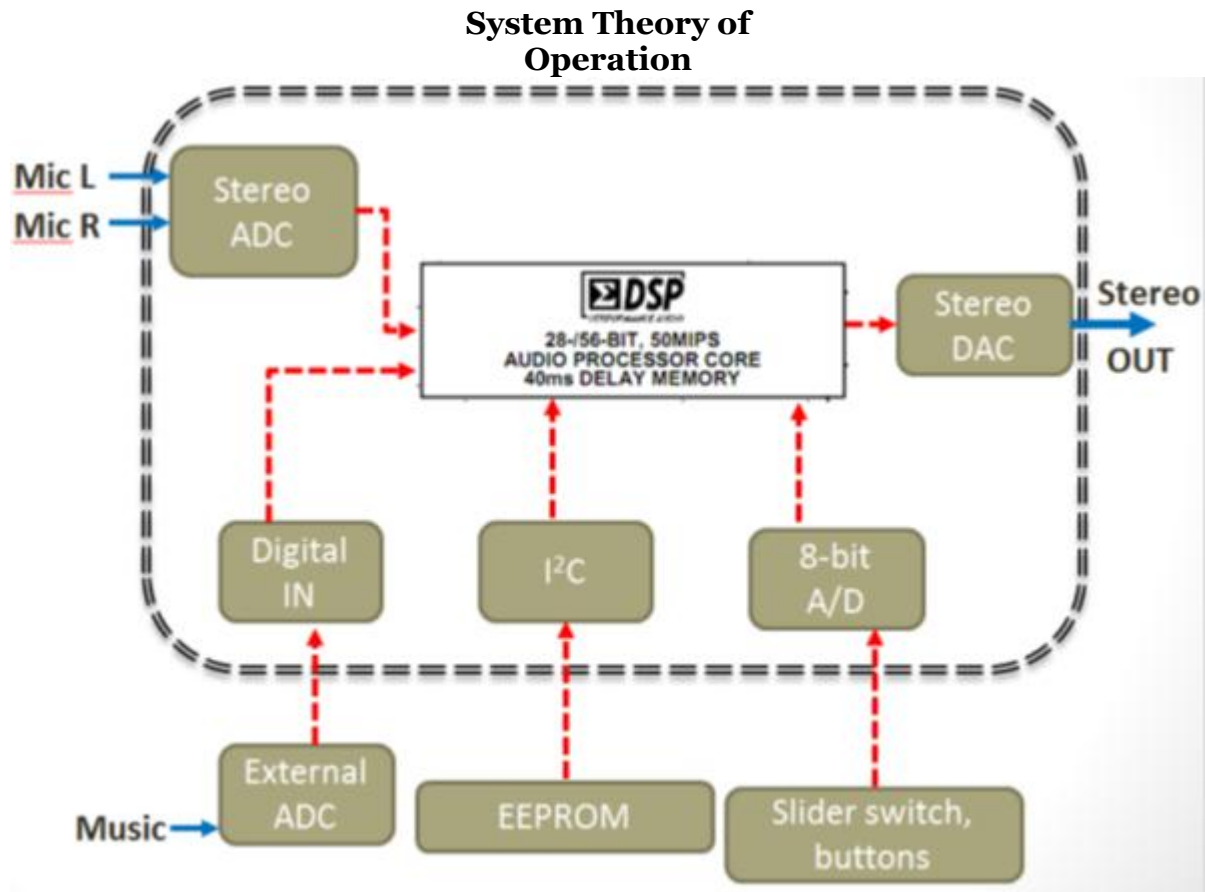
- The battery lifetime should be at least 5 hours, a length of time that accommodates typical iPod usage.
- The battery is easily rechargeable through a USB connection.

Safety Concerns:

- The device shall not harm the user through unintended electric shock during normal operation.
- The device shall limit and compress audio levels so as not to expose the user to unsafe audio levels unintentionally.
- Any generated heat from the device will be dissipated to prevent unintended burns or harm to the device.

3. Project Description:

An overview of the device and integration of the subsystems is depicted in the image below:



The final device is comprised of several subsystems that determine the overall device functionality. Schematics, code, and other technical information for all subsystems are included throughout this section.

3.1 Subsystem 1: Battery & Charger

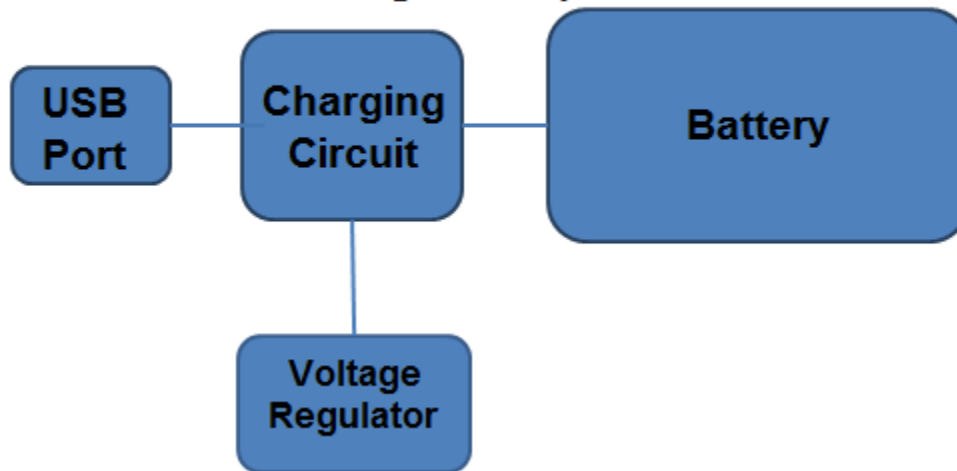
This subsystem is to provide stable power to all necessary components of the system over a target lifetime of at least 5 hours. The circuitry will allow for external charging and powering of the system.

- This subsystem should be able to demonstrate that the battery can successfully charge and discharge.
- The battery subsystem should be able to indicate the charging status, such as low charge or the charge is complete.
- The subsystem should be able to power the entire system from an external source (USB).

The specific battery (PRT-10472) was chosen based on the following:

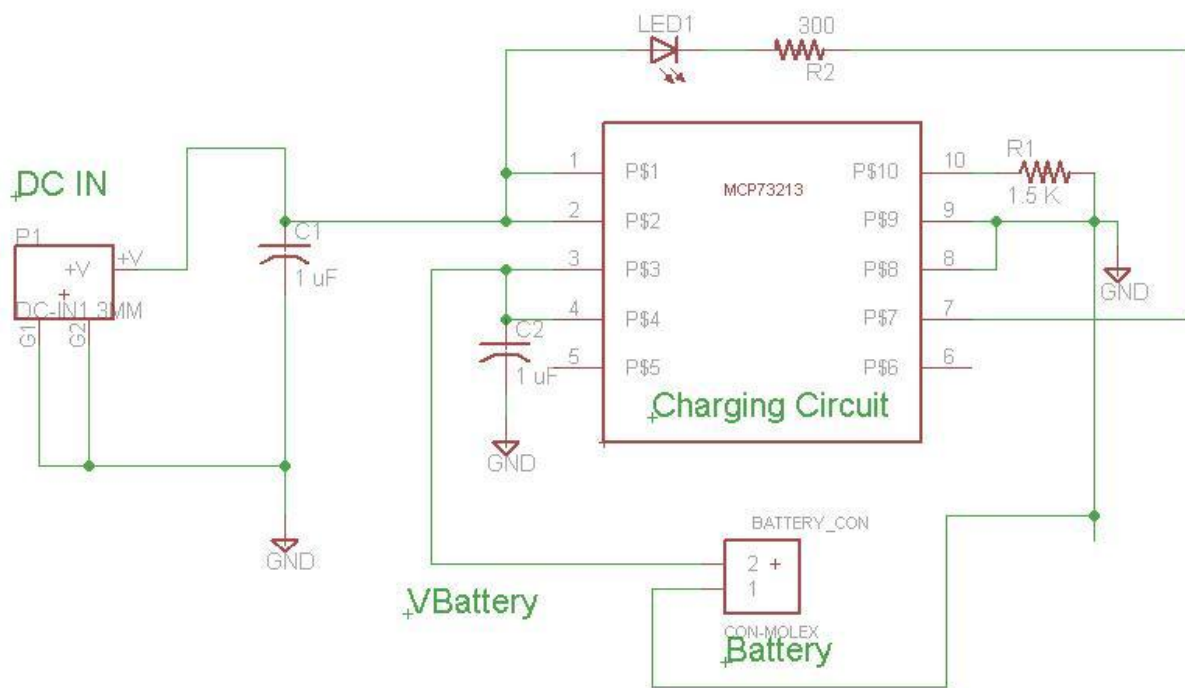
- Li-Ion Polymer battery can provide a rechargeable battery solution in a compact package that meets the necessary electrical specifications.
- A rechargeable battery was chosen as a specification of user preference and convenience.
- At under \$17 for the battery and the charging circuitry, this solution fits the budget.
- At 70mm x 35mm x 18mm and 85g per unit, the physical dimensions and weight of the battery are appropriate for a mobile device case to be used with iPods.
- The electrical characteristics (voltage, current draw, etc.) of the battery are sufficient to power all subsystems over the target lifetime.
 - More specifically, the 7.4V output is sufficient for the supply voltages required by the voltage regulator and other subsystem components.
 - Additionally, the ADC and DSP have the following current characteristics:
 - ADC (max analog = 45 mA, max digital = 22 mA) - max 67 mA, typically 58 mA
 - DSP (max analog = 85 mA, max digital = 60 mA) - max 145 mA, typically 90 mA
 - The 1000 mAh and 25C continuous discharge rate will enable the desired lifetime mentioned previously to be attained.
- The battery is readily adaptable for use with USB via the associated charging circuitry. The USB interface was chosen as it is a commonly used standard.

Battery and Charger Subsystem Block Diagram Representation



In addition to the parts described in the block diagram, the schematic below includes 330Ω and 2k Ω resistors, two 4.7μF capacitors, and an LED to indicate charging status.

Battery Charging Circuit

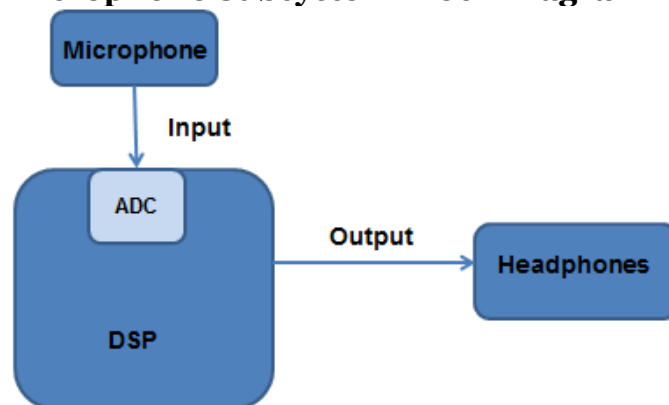


3.2 Subsystem 2: Microphones

This subsystem will obtain ambient noise through the microphones installed on the headphones and pass the signals through to the DSP for processing.

- The subsystem should show that it can successfully mount to the printed circuit board.
- The subsystem should demonstrate that an intelligible or useful output signal can be generated.

Microphone Subsystem Block Diagram



The specific microphones (Digikey part P9961-ND, Manufacturer part WM-55A103) were chosen based on the following:

- With a diameter of 9.70mm and height of 5.00mm, the MEMS devices are small enough to be discretely mounted to ear bud style headphones as well as over the ear style headphones.
- With a 250 μ A current draw (per microphone), the required battery resources are of minimal effect..
- Running at 1.5~10V, the microphones are suitable to function with the battery and other components.
- They are capable of being surface mounted to a PCB.
- The microphones do not require additional supply line resistance to set the output impedance, freeing up more space in the confined area around the headphones.
- They are very low cost, at \$2.60 per microphone.
- The microphones have a sensitivity of -38dBV, suitable for direct input to an ADC without prior amplification, eliminating the need for another circuit component.
- The microphones have a high SNR of 62 dB (A-weighted), sufficient for consumer level audio application.

3.3 Subsystem 3: Digital Signal Processor & Analog to Digital Converter

This subsystem is central to the functionality of the overall system. Its purpose is to take the separate raw audio signals generated by the iPod and the microphones installed on the headphones, process the signals according to the mode selected by the user, and output the modified signal to the headphone drivers.

- The DSP should be able to handle user input in the form of mode selection and load and run the program associated with each.
- The DSP should be able to communicate with on-board memory using the I₂C protocol to load and execute programs.
- The DSP should be able to perform the audio processing depending upon the mode of operation selected.
- The A/D shall convert the analog signal from the microphone into a digital signal at a sampling rate necessitated by the DSP for processing ($f_s=48$ kHz).
- The DSP will receive four distinct channels of audio. Two channels shall be analog and converted by the onboard ADCs. The other two shall be digital audio converted from analog by an external ADC.

The specific DSP (ADAU1701) was chosen based on the following:

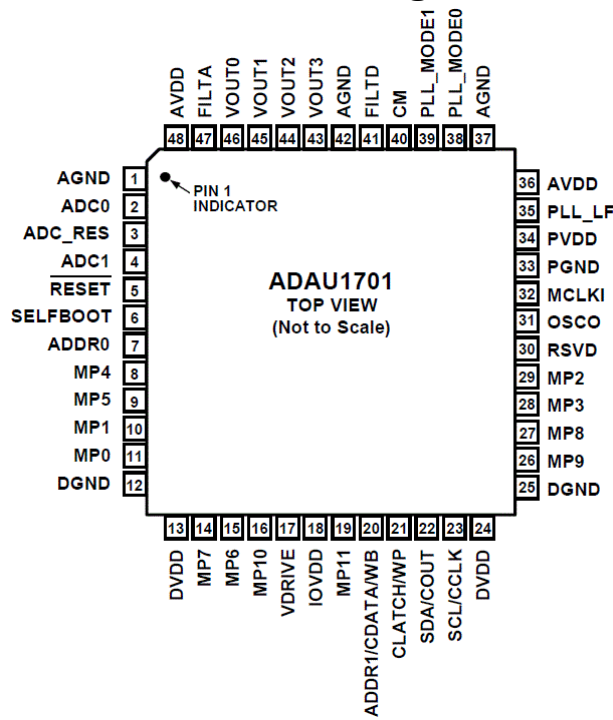
- It is a fully programmable DSP that can utilize SigmaStudio software to graphically configure a custom signal processing flow and implement basic functions in a simple manner.
- Programs can be loaded from a serial EEPROM through its own self-boot mechanism.
- It can communicate through an I₂C bus.
- The low power dissipation of 286.5 mW will allow the device to meet our previously specified device lifetime.
- The operating voltage of 3.3V can be readily supplied by the selected battery.
- The DSP has twelve multi-purpose pins that can be programmed to be used as serial data inputs, serial data outputs, digital control inputs/outputs to and from the SigmaDSP core, or inputs to the 4-channel auxiliary ADC. When set as an

input, the multi-purpose pins can control DSP program settings such as volume. As digital outputs, the pins can be used to drive LEDs or other logic to indicate status of internal signals and devices. In summary, the DSP can handle the user interface processing for mode selection without the assistance of another device.

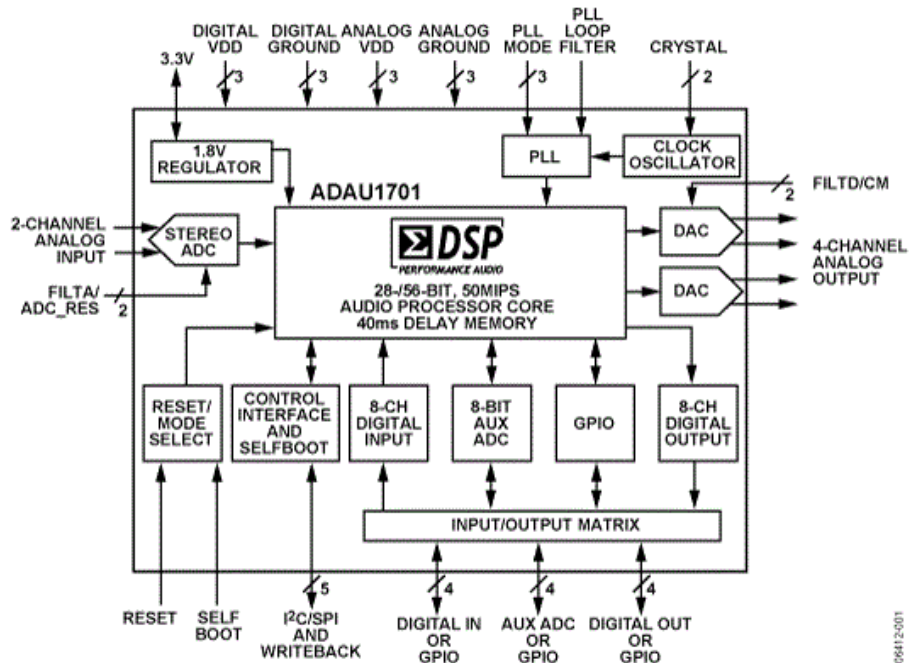
- The DSP is capable of processing ten distinct channels of audio, and therefore satisfies the requirement for four channels of audio.
- The DSP can receive digital audio over a variety of common protocols, including the I²S format that has been chosen for this project.

The functional block diagram and package pin-out for the DSP are shown below. For a full description of pin functions, a datasheet containing more information is available online at the link provided in the work cited.

ADAU1701 Pin Configuration

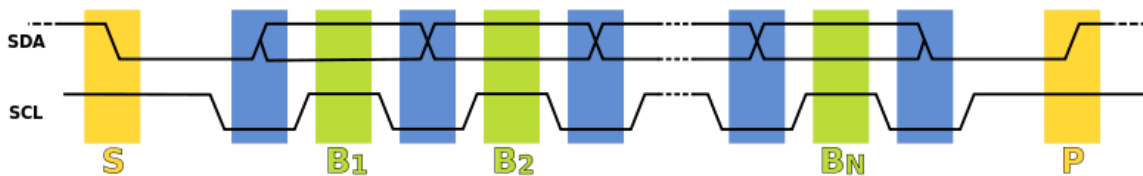


Functional Block Diagram of ADAU1701



The DSP will communicate with the EEPROM using the I²C protocol which is detailed below.

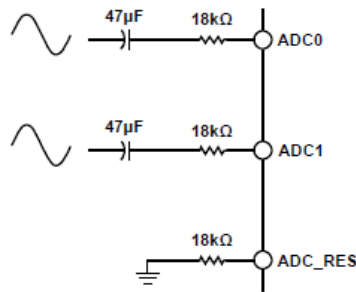
Timing Diagram for I²C Protocol



The data stream is initiated by the Serial Data line (SDA) signal going low while the Serial Clock line (SCL) signal stays high. The SDA then sets the transfer bit when SCL goes low as indicated in blue. Once SCL goes high the data for byte 1 is read. This process continues until all bytes have been transmitted and the stop bit is sent, which is indicated by the SDA signal going from low to high while the SCL signal remains high.

The configuration required for the DSP ADC is shown in the following schematic.

Audio ADC Input Configuration

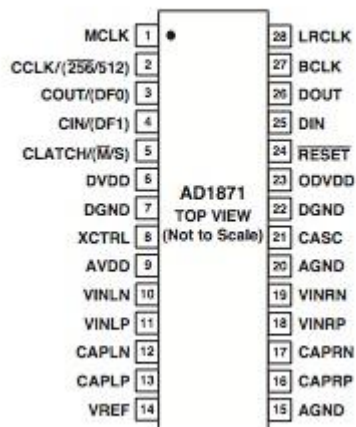


The specific external A/D (AD1871) was chosen based on the following:

- The device can convert two channels of audio on a single chip.
- The device can sample the audio at 48 kHz which is suitable for signal reconstruction in the audio range.
- The device operates on a voltage of 5.0V which is capable of being supplied by an on-board battery.
- The device can output digital audio over a number of serial protocols, including I²S which has been chosen for the digital audio communication to the DSP.
- The operating mode of the device can be set using hardware pins, eliminating the need for an SPI interface to set internal registers during assembly.

Pin Configuration for AD1871

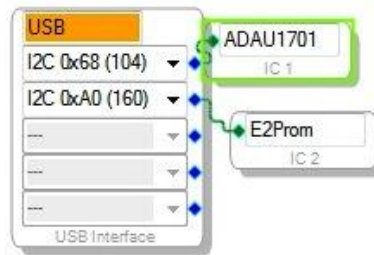
PIN CONFIGURATION



The following pins settings are used to set the operating parameters for the part:
 Pin 1: 12.288 MHz clock (synchronous to the DSP), Pins 2,3,4,5: GND, Pin 6: 5VDC, Pin 7: GND, Pins 8,9: 5VDC, Pin 15: GND, Pins 20,21,22: GND. Pins 26-28 connect to the DSP, and all other pins are used for analog audio interfacing.

DSP Programming

The following is the proof-of-concept subsystem demonstration DSP hardware configuration:

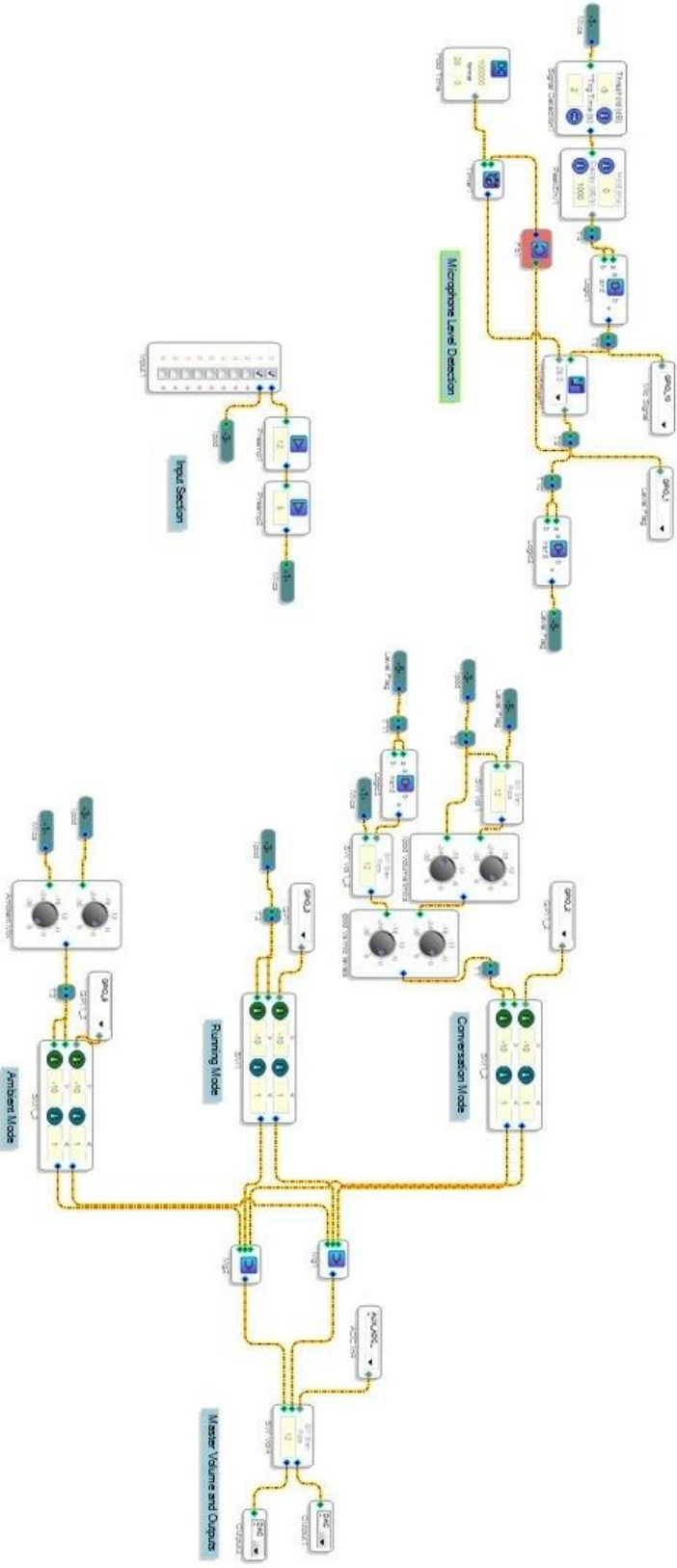


The following is the proof-of-concept subsystem demonstration DSP register settings:

The screenshot shows a 'GPIO' configuration window with a table of pin settings. The window has a checkbox for 'Forced By SPI' and a 'Debounce 20ms' dropdown. The table lists pins from MP0 to MP11 with their respective values, directions, and inversion settings.

Pin	Value	Direction	Inv
MP0	Low	Input GPIO Debounce	<input type="checkbox"/>
MP1	Low	Output GPIO	<input type="checkbox"/>
MP2	Low	Input GPIO Debounce	<input type="checkbox"/>
MP3	Low	Input GPIO Debounce	<input type="checkbox"/>
MP4	Low	Input GPIO Debounce	<input type="checkbox"/>
MP5	Low	Input GPIO Debounce	<input type="checkbox"/>
MP6	Low	Input GPIO Debounce	<input type="checkbox"/>
MP7	Low	Input GPIO Debounce	<input type="checkbox"/>
MP8	Low	ADC3	<input checked="" type="checkbox"/>
MP9	Low	Input GPIO Debounce	<input type="checkbox"/>
MP10	Low	Output GPIO	<input type="checkbox"/>
MP11	Low	Input GPIO Debounce	<input type="checkbox"/>

The following is the proof-of-concept subsystem demonstration DSP program:



4. System Integration Testing

4.1 Subsystem Testing

During the assembly and verification of the prototype, each of the subsystems were tested individually to ensure their proper functionality before the complete integration of all subsystems.

4.1.1 Battery, charging circuitry, and regulators

The charged battery should produce a voltage of 7.4 V, which is easily checked at the output terminals. The charging IC and circuitry should restore a depleted battery voltage, power the device while charging, and indicate the charging status of the battery. The regulators should be able to take an input voltage between 4 V and 10 V and produce a steady voltage of either 3.3 or 5 Volts. The regulators can be tested by simply measuring the voltage of the output when powered using a voltage meter or oscilloscope.

4.1.2 Microphones

With 3.3V DC supplied to the microphones from an external bench power, each microphone shall reproduce a 1kHz sine tone played at 1 Pa (measured in the plane of the microphone) with voltage between 9 and 18 mVrms, measured at the 2.5mm connection terminal. The above sine wave shall be free of noticeable distortion and clipping when viewed on an oscilloscope.

4.1.3 ADC

With a line level audio signal (nominally -10 dBVrms) connected to the ADC and 3.3V and 5V DC power supplied to the device from an external power supply, and a 12.288 MHz clock provided from an external function generator, the ADC shall begin outputting audio data in I2S format with the Left channel data triggered on a falling edge of LRCLK upon start up. This data will be observable on an oscilloscope. When a reset signal is applied to the reset pin, the device shall cease sending data and restart itself, again meeting the above requirements.

4.1.4 DSP

With a 3.3V DC supply connected to the DSP, the device shall generate its own clock and start up to operating mode.

Once running, a simple program shall be able to be downloaded successfully to the DSP using the provided USB interface to the header pins. The simple program shall do the following: generate a sine tone at 1 kHz and play it through DAC2 and DAC3. The analog audio inputs shall also be routed to these outputs.

To test the I/O capability of the ADC, the DSP will execute the above program. The test will be successful if an undistorted sine wave is heard in both ears when headphones are connected to the DAC2 and DAC3 outputs via the 3.5 mm jack, and when an iPod or other device is connected to the analog inputs, the content of the iPod is heard undistorted and free of unwanted noise.

To test the mode selection function of the DSP, a separate program will be used. A 500 Hz tone will connect through a mute controlled by the Conversation mode selection to the headphones, a 1 kHz tone will connect in the same manner but controlled by the Running mode selection, and a 2 kHz tone will also connect in the same manner but controlled by the Ambient mode switch. The overall volume of these signals will be controlled by the digitized values from the potentiometer. To verify the mode-switching hardware functionality, a different pitched tone will be heard in each mode and rotation of the potentiometer will adjust the volume accordingly.

4.2 Integrated System Testing

The regulators will be able to maintain the desired output voltage (3.3 V and 5 V) regardless of any current drawn from other subsystems, such as the DSP powering on. All subsystems dependent on a voltage supply (microphones, ADC, DSP) will operate with power from the battery and from the wall charger.

With the proper registers set, the DSP will receive two channels of distortion and noise free audio from the ADC on inputs 2 and 3.

The system will be able to detect a 40 dB SPL signal through the ear-mounted microphones and decrease the music audio volume, in order to demonstrate Running Mode. The same test will demonstrate Conversation Mode, but the detection shall only occur when the external signal is in the speaking frequency range. To demonstrate Ambient Mode, the device must be able to pass a signal from the microphones through to headphones, mixing the external and music signals. The device must also be able to play the music signal unchanged when in Bypass Mode.

4.3 Overall System Testing

The system was tested with the following setup: an iPod connected to the device with the 3.5mm cable, headphones and microphones connected to their designated ports on the device, the mode selection switch in ambient mode. Immediately upon connecting battery power to the board, the device passed music playing from the iPod to the headphones as well as stereo sound from the microphones at an equal level. When switched to conversation mode, the device would lower the music to a still audible level and pass the microphone audio when a voice was detected. The voice recognition was tested with varying voice qualities (male vs. female, high pitch vs. low pitch) speaking to the user at a distance of 3 ft. The tests were performed in a variety of settings: quiet indoor space, “industrial space” simulated by fans and equipment (electronics, computers, power tools) operating outside of a 10 ft radius, and “crowded space” simulated by additional people talking at a normal volume outside of a 10ft radius. In all cases, only the sound of the tester’s voice speaking to the user and the user’s voice triggered the actions of conversation mode. The above tests were also used to test the running mode algorithms, in which case all of these situational noises triggered the volume increase. The device passed all operational tests.

5. User's Manual

5.1 User Configuration

The device requires no installation procedures but a quick and painless set-up. All the user needs to do is insert his or her iPhone SilentRave's device and make all necessary connections.

5.2 Proper Device Operation

The user is alerted when the battery is charging by an LED. The user selects whether the product is on or off by changing the 2-way switch to the desired selection. The user can tell if the product is working by testing the device. For example, the user can set the device in the running mode and determine whether the headphone output increases if he enters a noisy environment.

5.3 User Troubleshooting

The device is designed for typical consumer use in which the user has no knowledge digital signal processing and lacks the ability to program or adjust the device. User troubleshooting is expected to be similar to that of the troubleshooting for an iPod. If the user is having problems, it is suggested that all connections are checked. These connections include the microphones, headphones, and charging circuitry. The user can also ensure that the LED is lit when the battery should be charging.

6. To-Market Design Changes

Although SilentRave's device met all requirements, to-market design changes could be made. First, the case could be greatly improved. Marketed as a combined case and headphone device, a future case would focus on and minimize the size and weight to ensure that having modified headphones would not come as a drawback. In upcoming models, smaller parts overall would be chosen that would allow for a less bulky case. The dimensions for the vacuum-formed container were also larger than necessary; inside the container, there was a lot of room for additional parts.

In addition, by selecting different parts, the aesthetics of the case could be improved. The three-way switch selected is much smaller than its purpose called for. Due to the switch's minimal projection, an additional part is needed so that the user could switch the modes from the exterior of the case. If time had allowed, a different three-way switch would have been chosen to replace the current piece. Additionally, the potentiometer was smaller than desired; the volume wheel required a bottle cap so that the user could turn the switch and adjust the volume. A different potentiometer would have been chosen that would eliminate the need for an attached part. With these new parts, a much more aesthetically pleasing product could be created.

By reducing the size of the individual parts and in turn the overall size of the case, a much smaller and aesthetically pleasing product could be created. Consequently, the case would be more suitable for attachment to the user's body when running or in other conditions where the user may be active. Further, as a device used when running or in active conditions, a sturdy case is desired. In the future, the materials for the case could also be chosen so that it could withstand falls and provide some protection from impact to the device.

Additional to-market design changes would be necessary to reduce the noise problems apparent in the proof-of-concept demonstration. Additions of a headphone amplifier on the output of the DSP's DACs would correct for distortion created by output clipping and allow for better volume control. Input gain staging, all passive electronic components, on the analog inputs to the AD1871 would help to correct the low SNR of the music from an iPod by better bridging the impedance to the headphone amplifier built into the iPod for maximum voltage transfer.

Future designs would also be able to charge from a USB. The model produced was unable to use a USB because a regulated 5v power supply for the external ADC was required; this type of power supply isn't possible over USB. In future designs, chips that perform the same job but also run on 3.3v would be chosen. This type of chip requires an SPI interface to set its registers, however. Due to time constraints, this was not possible.

Furthermore, new modes programmed for different scenarios and adjustable thresholds through user input and/or ambient noise monitoring could also be created for future products.

7. Conclusions

Headphones allow a user to enjoy music while keeping the sound from disrupting others in surrounding areas. However, typical headphones come with trade-offs. Some environments constantly change or the user may enter & exit various environments, altering the aural inputs that one is receiving. Depending on the environment that the user is in, he may want his microphones to react in different ways.

Unlike other devices on the market, which propose a solution to a single problem or require the user to produce some external input, SilentRave's device can meet all of these changing needs automatically. The user can select one of three modes that best addresses the problem at all hand by having different programs run depending on the design of the mode.

In all modes, the device processes the external sound through a stereo enhancement algorithm to simulate natural stereo imaging, compresses and limits the external sound to prevent overloading and protect one's ears (e.g. if someone yells into one of the microphones), and equalizes the external sound to seem natural and comfortable through the headphones.

In a scenario when the user is in a safe but dynamically noisy area, he may want to be in the running (total isolation) mode. In this mode, the device automatically increases the sound when external audio levels increase. Consequently, when the external noise is eliminated, the sound level is also reduced.

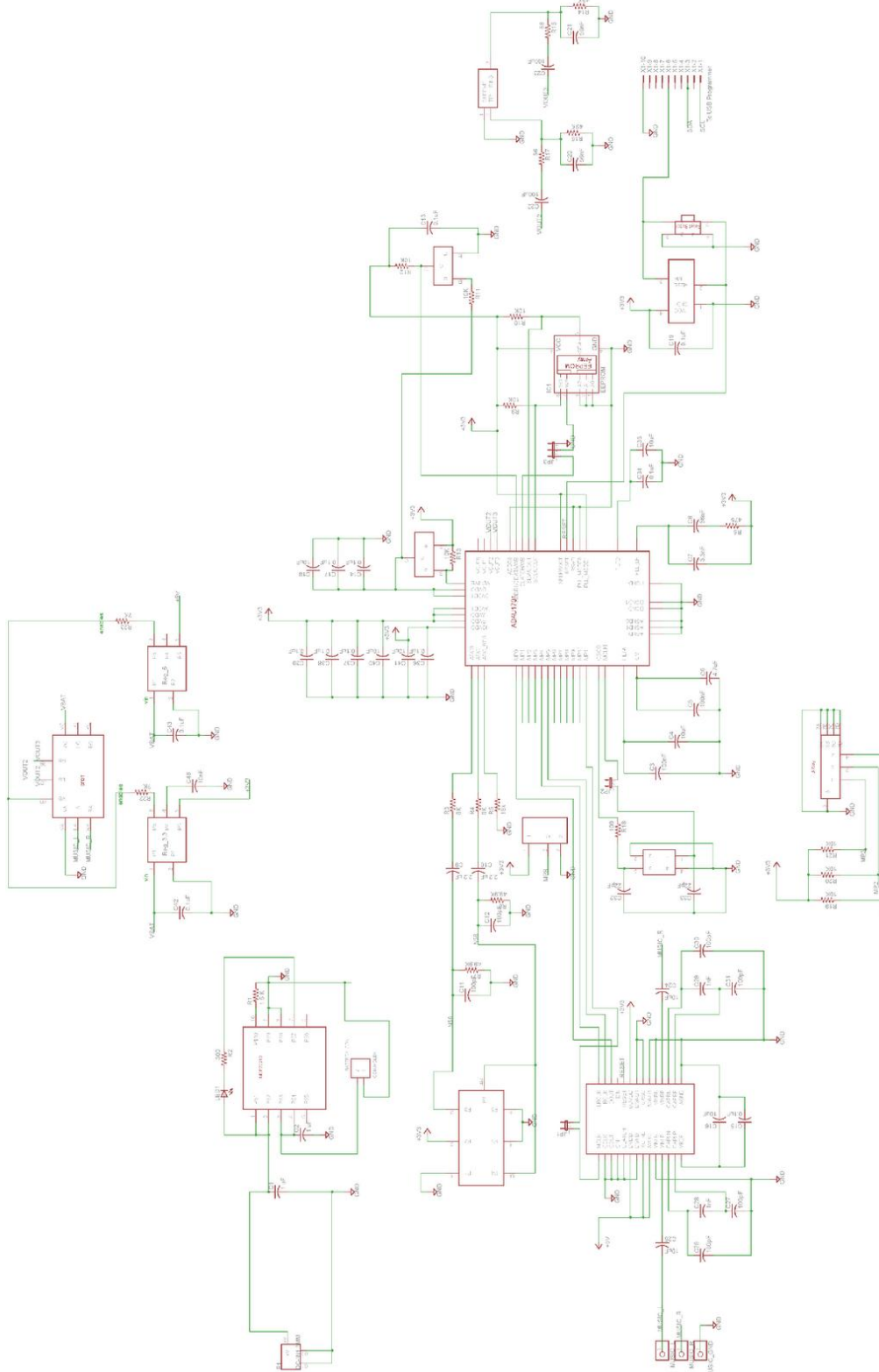
In scenarios in which the user expects interruptions but also wants to be able to enjoy music in between interruptions, the user may want to use the conversation mode. In this mode, when a sizeable level of human voice (20 Hz – 20 kHz) is detected, music volume is decreased and external audio boosted. The user does not need to be aware of others or deliver any input to the device. When the external noise is eliminated, the volume of the user's audio is increased again.

In scenarios when the user wants to remain aware of his surroundings but still enjoy audio, the ambient mode automatically mixes in a level of external audio (microphone signal) with the user's audio.

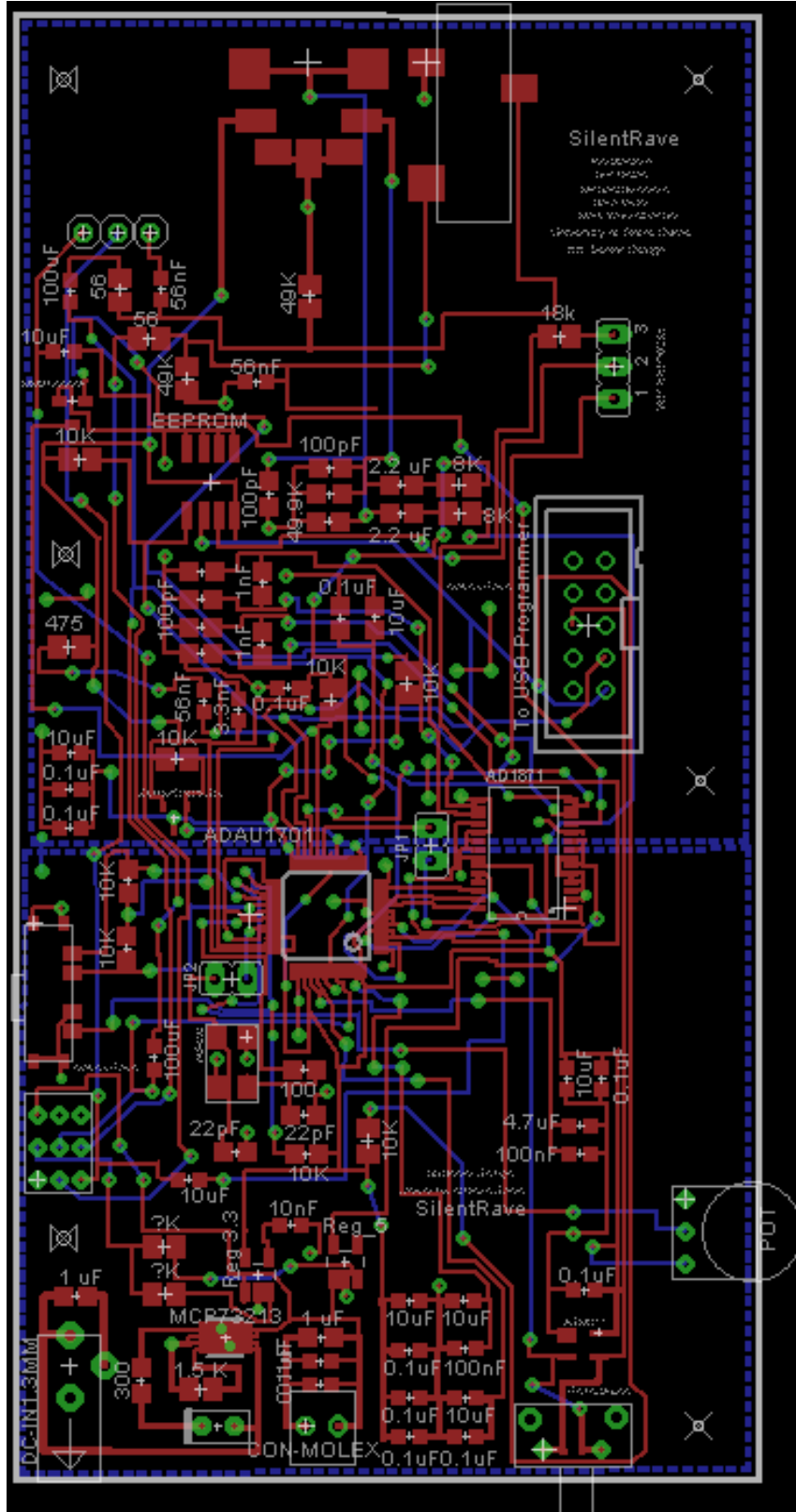
Aside from the aesthetic presentation of the case and noise in the audio, the functionality of the final design was as described in initial documentation. SilentRave's device met nearly all subsystem requirements and each subsystem communicated well to produce one coherent, interconnected system.

8. Appendices

8.1 Complete Hardware Schematic



8.2 Complete Board Layout



8.3 Relevant Parts or Component Data Sheets

Digital Signal Processor

http://www.analog.com/static/imported-files/Data_Sheets/ADAU1701.pdf

Analog-to-Digital Converter

http://www.analog.com/static/imported-files/data_sheets/AD1871.pdf

External EEPROM

<http://ww1.microchip.com/downloads/en/DeviceDoc/21189S.pdf>

Battery Charging Circuit

<http://www.sparkfun.com/datasheets/Prototyping/Batteries/MCP73831T.pdf>

8.4 Additional Works Cited:

- <http://creativecommons.org/licenses/b-sa/3.0>
- <http://ww1.microchip.com/downloads/en/DeviceDoc/22190b.pdf>

8.5 DSP Code

- The code can be seen in the accompanying SigmaStudio file in .dspproj format.