Heart and Lung Sound-sensing shirt

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Introduction

Objective

According to the World Bank and WHO, half the world lacks access to essential health services and 100 million are pushed into extreme poverty because of health expenses. There are many factors for this such as the lack of good doctors and the poor quality of medical diagnostic devices. Often, this causes people to travel large distances to talk to a doctor, which can be hard at times. This inspired us to take on a senior design project that may potentially solve this problem. Our goal is to design and build a shirt that is capable of detecting heart and lung sounds which a user can access on his smartphone and then send to a professional doctor.

Background

Many people may be stressed, sometimes even afraid to go to the doctors, even if it's just for a checkup. In many countries, people do not have access to quality healthcare and often have to travel large distances meet a doctor. Even after investing time and money, they might not be able to get good quality treatment. In this case, many people would benefit if they had were able to consult a doctor without physically going there. Therefore, this project will provide convenience for many people and cause them less stress when the need to see a doctor arises.

In the market, there are shirts that can detect a user's breathing and heart rate. Other products like Littmann Electronic stethoscopes offer good biological sound detection but do no capture the sounds from different areas in the body at the same time like our device. These stethoscopes are also significantly expensive than standard auscultation devices. This shirt will detect the sounds of both the heart as well as lung, and by implementing a Bluetooth feature for communicating with a smartphone, the results will be available to be sent to an expert.

High-level requirements

- The recorded sounds must be able to recreate the stethoscope experience for the doctor
- Recorded sounds must help in accurate diagnosis of patient's condition
- Bluetooth transmission should not damage the audio file
- Doctor should be able to play sounds from several microphones

Design

Block Diagram

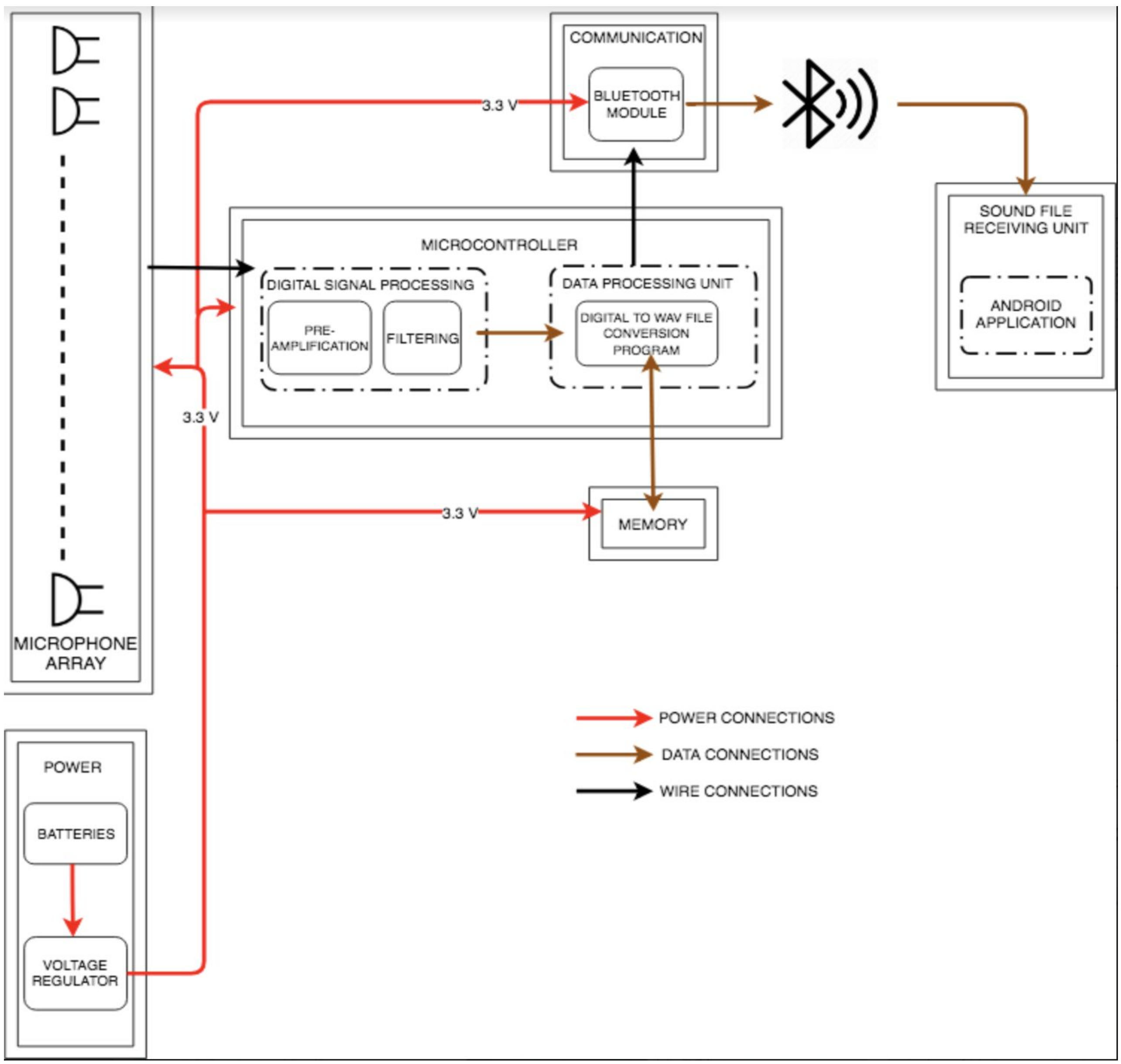

Figure 1: Block Diagram

The Block diagram in the above figure demonstrates how different units in the system interact with each other to fulfill the high-level requirements of this project. The ten microphones will be placed at different positions on the shirt. The output signal from each of these digital microphones will be transferred to a Microcontroller for Digital Signal processing and format conversion. An external 256 MB memory stores our WAV files. The files from the system will also be made available to the user on a smartphone and can be sent to a doctor for analysis.

Physical Design

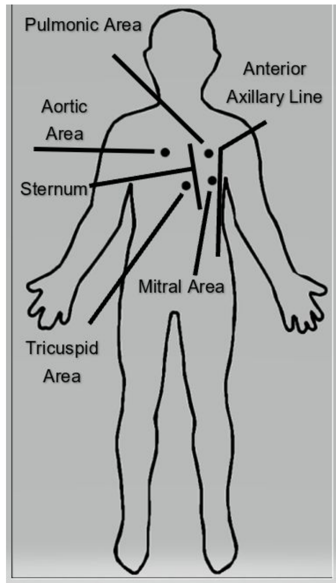

Figure 2: Physical diagram

The figure above demonstrates the locations of six of the digital microphones on the chest and abdominal area of the shirt to detect the heart as well as lung sounds. Two of the microphones are going to be placed on the back across the Aortic Area and Tricuspid Area. Another two will be placed placed near Mitral Area and Pulmonic Area to capture ambient noise. The shirt would consist of two layers. The microphones, power circuit and microcontroller will be placed between the inner and the outer layer to isolate the patient's skin from the electronics. The microphone heads will be placed in 3-D printed headers which will make contact with the patient's skin to capture signals and the shirt will be snug enough to keep the microphones in place.

Functional Overview

Microphone Array

Microphones are used in our system to sense the heart and lung sounds. 10 TDM MEMS microphones will be placed in an array on the chest as well as upper back region of the shirt and interfaced with a microcontroller as shown in Figure 4. All microphones in the array sample their acoustic signals synchronously, enabling precise array processing. Statistical analysis showed that the major concentration of energy, for both first heart sound (S1) and second heart sound (S2), is below 150 Hz which may indicate that both sounds are caused by vibrations within the same structure, possibly the entire heart. However S2 spectra have greater amplitude than S1 spectra above 150 Hz, which may be due to vibrations within the aorta and pulmonary artery. In subjects with healthy lungs, the frequency range of the vesicular breathing sounds extends to 1000 Hz, whereas the majority of the power within this range is found between 60 Hz and 600 Hz. Other sounds, such as wheezing or stridor, can sometimes appear at frequencies above 2000 Hz. Based on this data, the microphones should be able to detect sounds within the frequency range 50 Hz - 2500 Hz. Two of the microphones will be used to capture ambient noise. The digital output from the microphones be sent to the microcontroller for amplification, filtering, noise cancelling and data processing. The following block diagram in Figure 3 was taken from the datasheet for Invensense ICS-52000 [6].

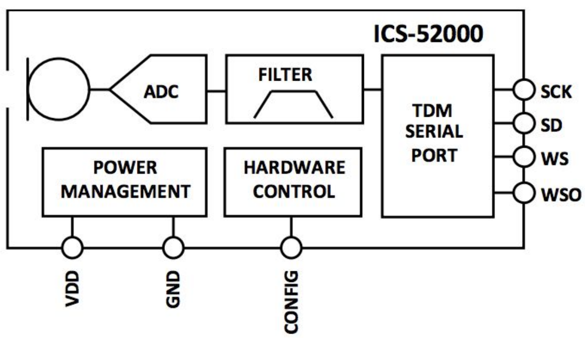

Figure 3: ICS-52000 Block Diagram [6]

Pin Name	Function
WSO	WS output, connected to the the WS of the next ICS-52000 in the array
CONFIG	Pulled to VDD. The state of this pin is used at power-up
GND	Connected to ground
VDD	Power, 3.3 V. This pin will be decoupled to GND with a 0.1 µF capacitor
WS	Serial Data-Word Select for TDM Interface
SCK	Serial Data Clock for TDM Interface
SD	Serial Data Output for TDM Interface. This pin tri-states when not actively driving the appropriate output channel. The SD trace will have a 100 kΩ pulldown resistor to discharge the line during the time that all microphones on the bus have tri-stated their outputs.

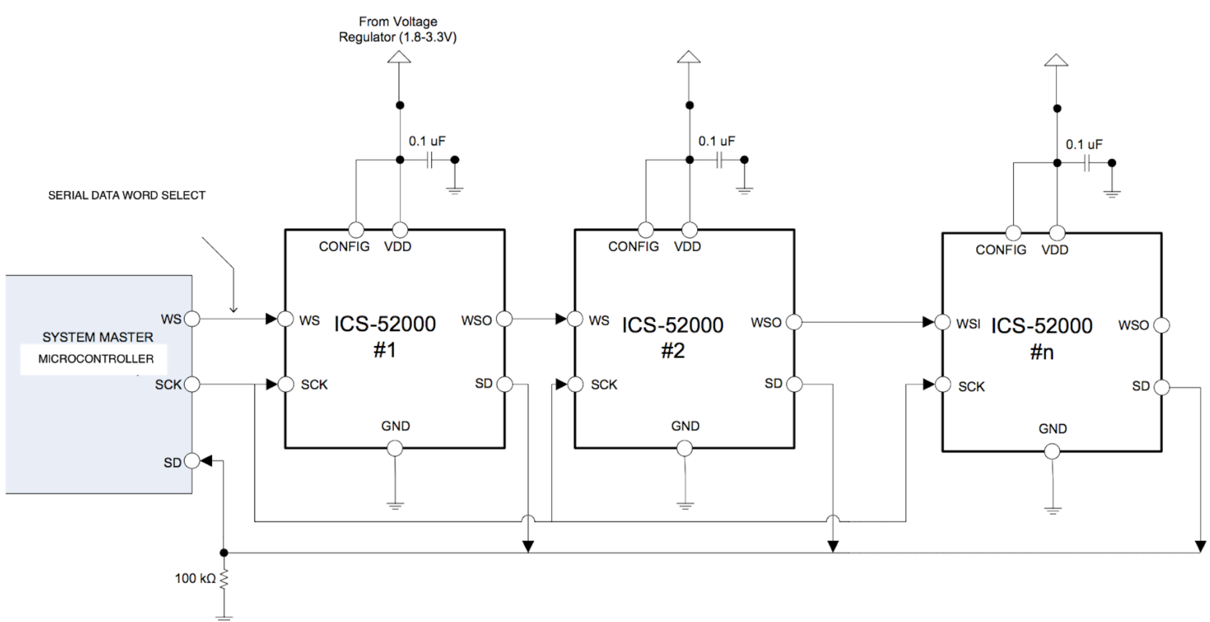

Figure 4: The Microphone array interfaced with the Microcontroller

Requirements	Verification
● Have a Frequency response in the range 50 Hz - 20 kHz ● Operate on 3.3 V +/- 0.1 V	● Produce audio and use the oscilloscope to plot the voltage response ● Power the microphones with voltage in the range 3.2 V - 3.4 V

Power Unit

The Power unit will of a single power line which supplies 3 - 5 V in order to power the microcontroller, microphones, memory and the Bluetooth chip. There will be a voltage regulator to ensure there are no voltage spikes which can damage the components of our system.

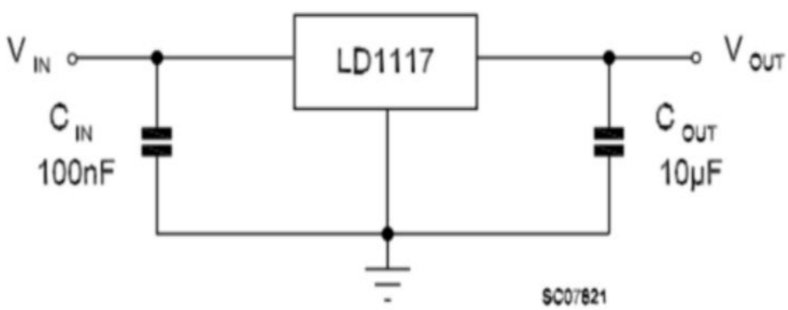

Figure 5: Power supply circuit [4]

Requirements	Verification
● Generate 3.3 V +/- 0.1 V ● Can operate at currents 0-200 mA ● Batteries provide 3000 mAh of power	● Measure the output voltage from the voltage regulator and ensure that it stays within the 3.2 - 3.4 V range ● Use a constant current circuit to draw 200 mA from the power supply and voltage regulator ● Ensure that the batteries run for 15 hours at maximum current (200mA)

Signal Processing Unit

The Digital Signal Processing Unit filters out the noise and amplifies the relevant sound signals. The outputs from 8 MEMS microphones and the ambient noise captured from 2 microphones are fed into an adaptive filtering unit in the microcontroller. The signals of interest are compared with the noise and the ambient noise is subtracted. We are using an adaptive algorithm, the Least Mean Square (LMS) algorithm, to get a better estimate of the signal by changing the value of the filter coefficients. You can see a visual representation of the algorithm in Figure 6. LMS starts by filtering the reference input using weights (w) of the adaptive filter and creates an estimate of the primary input. It then creates an error signal using the equation,

$$\text{error} = \text{primary input} - \text{estimate of primary input} \quad (1)$$

The error signal is then sent into the adaptive filter to update the weights (w) and increasingly make the algorithm more efficient. The weights are updated using,

$$w(n+1) = w(n) + 2*\mu*e(n)*x(n) \quad (2)$$

where w(n + 1) is the new vector of filtering weights, w(n) is the current vector of filtering weights, μ is the size step parameter (determined through experimentation), e(n) is the error signal, and x(n) is the vector of recorded reference signals x with length of n.

For the heart, the frequencies of the sounds we want to detect in lie in the range 60 - 150 Hz and for the lungs, the signals of interest are in the range 50 - 2500 Hz. This can be achieved by implementing a Digital Bandpass filter with cut-off frequencies between 50 and 2500 Hz using the microcontroller.

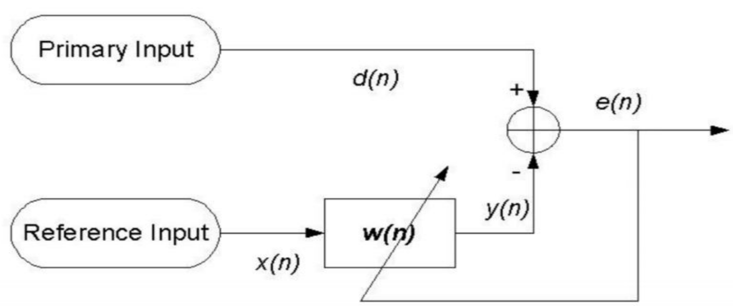

Figure 6: Block diagram of adaptive filter[2]

Requirements	Verification
• -3 dB Frequency Response below 25 Hz and above 300 Hz for filtering out	• Use signal generator to generate signal at 25 Hz and below. Measure

heart sounds ● -3 dB Frequency Response below 50 Hz and above 2500 Hz for filtering out lung sounds	Frequency response to verify is -3 dB below. Repeat the process for signals 300 Hz and above ● Use signal generator to generate signal at 50 Hz and below. Measure Frequency response to verify it is -3 dB below. Repeat the process for signals 2500 Hz and above

Data Processing Unit

In addition to Signal processing, the microcontroller also performs data processing by converting the digital signal to a WAV file format and send it to the communication unit. The header of the WAV file is 44 bytes long and has the following structure.

Positions	Sample Value	Description
1 - 4	"RIFF"	Marks the file as a riff file. Characters are each 1 byte long.
5 - 8	File size (integer)	Size of the overall file - 8 bytes, in bytes (32-bit integer). Typically, you'd fill this in after creation.
9 -12	"WAVE"	File Type Header. For our purposes, it always equals "WAVE".
13-16	"fmt "	Format chunk marker. Includes trailing null
17-20	16	Length of format data as listed above
21-22	1	Type of format (1 is PCM) - 2 byte integer
23-24	2	Number of Channels - 2 byte integer
25-28	44100	Sample Rate - 32 byte integer. Common values are 44100 (CD), 48000 (DAT). Sample Rate = Number of Samples per second, or Hertz.
29-32	176400	(Sample Rate * BitsPerSample * Channels) / 8.
33-34	4	(BitsPerSample * Channels) / 8.1 - 8 bit mono2 - 8 bit stereo/16 bit mono4 - 16 bit stereo
35-36	16	Bits per sample
37-40	"data"	"data" chunk header. Marks the beginning of the data section.
41-44	File size (data)	Size of the data section.
Sample values are given above for a 16-bit stereo source.		

Figure 7: WAV file structure [5]

Requirements	Verification
● WAV file conversion of the digital signal	● Run a sample digital signal through the data conversion program and ensure output format is WAV and the audio can be played

Memory

An external memory will store the patients data. The data stored will contain the heart and lung audio file of the patient as well as the time it was recorded. This unit is needed so that the doctor can access the patient's history at any given time. This unit will store several megabytes of WAV files.

Requirements	Verification
• Write Program memory at 100 kBit/s • Have a total usable memory of > 256 kbit • Operate on 3.3 V +/- 0.1 V	• Record the time it takes to write a 10 MB WAV file • Try to fill the memory with > 256 Kbit • Attempt to power the chip with 3.2 - 3.4 V

Communication

This unit will be the interface of our device to the outside world and will consist of a Low Power Bluetooth module. It will make sure that the patient's information is sent to the right people. This unit will consume little power as a Power Enable pin on the chip allows us to power the chip only when required and will only need to be powered when information is being sent.

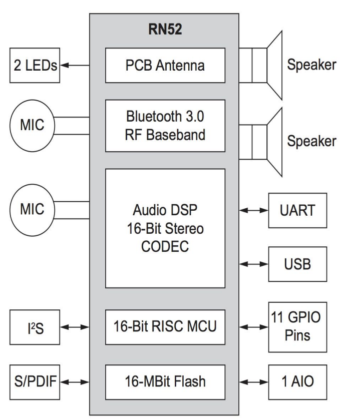

Figure 8: RN-52 Bluetooth module

Requirements	Verification
• Ability to communicate through UART or SPI • Have at least 2 MB/s data communication rate for WAV file transmission • Ability to transfer the WAV file while maintaining good quality • Operate on 3.3 V +/- 0.1 V and consume	• Check Datasheet to confirm the communication protocol required for the Bluetooth chip and verify by sending sample data to the chip. • Program to send an WAV file from microcontroller to Bluetooth chip • Convert sample audio to WAV format and transfer the file to a smartphone • Attempt to power chip with with voltage between 3.2 - 3.4 V

Sound File Receiving Unit

The receiving unit will consist of a smartphone through which the user will be able to access the sound files generated from our system using an Android application. These sound files can then be sent to a doctor for an accurate diagnosis of the patient's condition.

Requirements	Verification
• The Android application must provide the capability to play audio files from all the microphones corresponding to different regions of the body	• Run sample WAV file and transfer it via Bluetooth to the smartphone

Additional Circuits

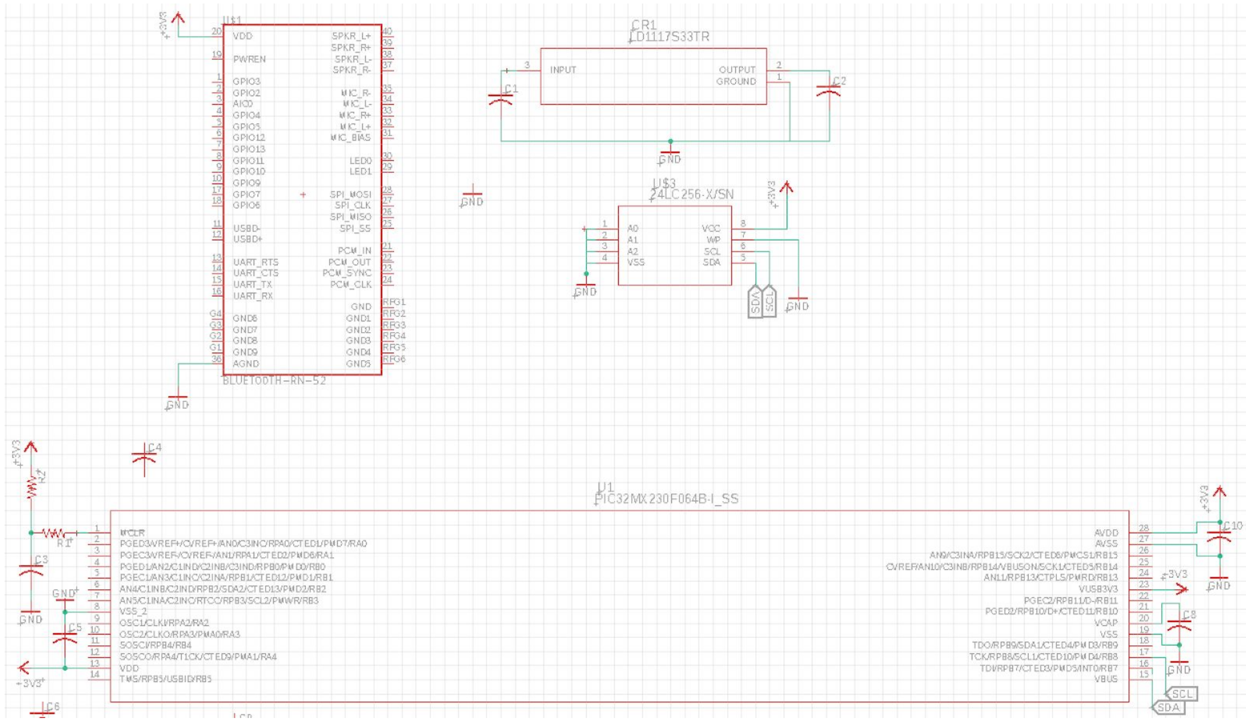

Figure 9: Microcontroller + Bluetooth + Memory circuit

Tolerance Analysis

The outputs from the digital microphones in our project are crucial for the success of this project. By design, all of the ICS-52000 microphones on a common bus will sample their acoustic signals simultaneously and output their individual data words in their respective time slots in the TDM bus. Synchronized sampling is critical for the performance of a multi-microphone array because the algorithms typically used with these arrays require accurate synchronization between signals for optimum performance. Without the ICS-52000’s integrated synchronization at start-up, these DSP algorithms would require a complicated series of delays and/or buffers to pre-process the signals to achieve the best algorithm performance.

It is also very important to consider the Total Harmonic Distortion of the output of the microphone in our analysis. A higher THD measurement indicates a higher level of harmonics present at the output of the microphone. The THD of the MEMS microphones is calculated from the first five harmonics of the fundamental. The input signal for this test is typically at 105 dB SPL, which is 11 dB above the reference SPL of 94 dB. THD is measured at a higher SPL than other specifications because, as the level of the acoustic input signal increases, the THD measurement typically increases as well. A rule of thumb is that the THD triples with every 10 dB increase in input level. Therefore, THD less than 3% at 105 dB SPL means that the THD will be less than 1% at 95 dB SPL.

Cost and Schedule

Cost Analysis

Labor

Our fixed developmental costs are about $40/hr, 10 hrs/week for three laborers.

$$\frac{\$40}{1\ hr} * \ 2.5 \ * 10\ weeks \ * \ \frac{10\ hours}{1\ week} = \frac{\$10{,}000}{laborer}$$

Considering we have 3 laborers, our total labor cost is,

$$\frac{\$10{,}000}{laborer} * 3\ laborers = \ \$30,000.$$

Parts

Description	Manufacturer	Model#	Units	Units cost	Total
Microcontroller	Microchip	PIC32MX230F256B-I/SS	1	$2.53	$2.53
Microphones	Invensense	ICS - 52000	10	$2.85	$28.5
Memory chip	Microchip	24LC256	1	$0.72	$0.72
Bluetooth	Microchip	RN-52	1	$44.94	$44.94
Resistors	Various	Various	20	$0.10	$2.00
Capacitors	Various	N/A	24	$0.30	$7.2
Voltage Regulator	STMicroelectronics	LD1117	1	$1.95	$1.95
PCB	PCBWay	N/A	1	$4.00	$4.00
Cotton Roll	Curad	N/A	1	$5.23	$5.23
Lithium Coin Battery	Renata	CR2477N.IB	5	$1.74	$8.7
Connectors	Various	Various	4	$1.00	$4.00

Oscillator	Various	Various	2	$1.40	$2.8
Total Cost					$112.57

Grand Total *= Labor Cost + Parts Cost*

= $30,000 + $112.57

= $30,112.57

Schedule

Deadline	Abhiyash	Hesham	Marc
3/4	Build the Power circuit and test the voltage regulation.	Order the microphones, bluetooth module, microcontroller and memory chip.	Build the Power circuit and test the voltage regulation.
3/11	Verify the microphones, test the Frequency response.	Program the DSP Filtering and test it with sample digital signals.	Program the DSP filtering and test it with sample digital signals.
3/18	Program the data conversion of digital signal into WAV format and test memory circuit.	Solder the microphones onto Evaluation boards.	Program test benches for data conversion.
3/25	Place Bluetooth module into the communication circuit on a Protoboard for testing.	3-D print the microphone headers and stitch the two-layered shirt.	Program test benches for data conversion.
4/1	Program the Bluetooth chip to transfer audio file to smartphone.	Develop Android application for playing audio file.	Develop Android application for playing audio file.

4/8	Test the final system and doctor for feedback on the shirt.	Test the final system and ask doctor for feedback on the shirt.	Test the final system and ask doctor for feedback on the shirt.
4/15	Prepare final presentation	Prepare final presentation	Prepare final presentation
4/16-4/20	Mock demonstrations	Mock demonstrations	Mock demonstrations
4/23-4/25	Demonstrations	Demonstrations	Demonstrations
4/26-4/27	Mock presentation	Mock presentation	Mock Presentation
4/30	Presentation	Presentation	Presentation

Ethics and Safety

There are several aspects of the project that can pose a safety hazard to the user. The Power Unit consists of batteries providing a total of 3.3 V to our system and it can easily overheat causing discomfort to the user and correlates to the IEEE Code of Ethics #9 [3].
Another possible hazard is water which can short the components on the circuit board and even harm the user which again correlates with the IEEE Code of Ethics #9 [3]. We need to ensure that the user is not sweating while wearing this shirt and it is not worn for prolonged periods of time.
The material of our shirt is another safety concern as certain materials can cause allergic reactions in some people. We are using a shirt made of 100% cotton for this project and are going to ensure that the volunteers testing this suit do not have such allergies. We will make sure that the user is aware of the materials used to build the suit following IEEE Code of Ethics #3 [3].

A crucial factor in determining the success of the suit is whether doctors are able to detect the relevant sounds from the sound files generated by the system. Therefore, we need to make sure that we work closely with a doctor and a patient to make our device better in accordance with IEEE Code of Ethics #7 [3].

Since this shirt is a medical device, we need to make sure Food and Drug Authority (FDA) medical device regulations are satisfied and the need for human participants in this project requires us to follow the Institutional Review Board (IRB) guidelines [1].

References

[1] “ A. Paulson, “Institutional Review Board (IRB),” American Public University System (APUS), 09-Mar-2017. [Online]. Available: http://www.apus.edu/academiccommunity/research/institutional-review-board/index. [Accessed: 10-Feb-2018].

[2] V.K. Iyer, P.A. Ramamoorthy, H. Fan, and Y. Ploysongsang, “Reduction of heart sounds from lung sounds by adaptive filtering,” IEEE Trans. Biomed. Eng., vol. 33, no. 12, pp. 1141–1148, Dec. 1986. [published erratum appears in IEEE Trans. Biomed. Eng., vol. 35, no. 1, p. 76, Jan. 1988].

[3] “ Ieee.org, “IEEE IEEE Code of Ethics”, 2016. [Online]. Available: https://www.ieee.org/about/corporate/governance/p7-8.html. [Accessed: 10- Feb-2019].

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