Threshold Noise-Cancelling Headphones

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Abstract

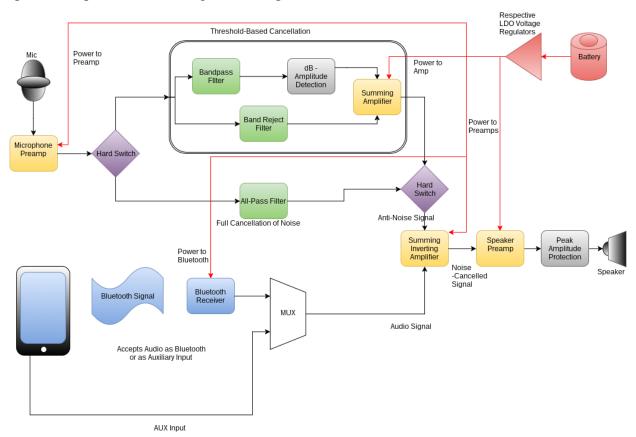
Our project is an adaptation of the conventional noise cancelling headphone design. The cancellation circuit now ignores the human speech band allowing the user to more freely communicate with another person in a noisy environment. We used an entirely analog design to avoid the delays associated with analog to digital conversion and its inverse, which allows for proper noise cancellation. The circuit takes in an ambient noise signal through a microphone, is processed by our circuit's filter system, inverted, and finally added with an incoming audio signal and sent to the headphones. Our results indicate the possibility for complete functionality. However, this design is a prototype and does not meet our original goal of an in-ear device.

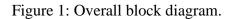
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1 Introduction

Our project seeks to solve the problem of being unable to focus on a single conversation in a noisy environment. The growing trend of open workspaces makes it more difficult for people to concentrate on their work, however someone cannot simply wear headphones and still effectively interact with the people around them. The solution is to only cancel out background noise and to only pass through sound in the human speech range. This project is a prototype for a future model that will be completely in-ear and discreet, much like a hearing aid. There was a number of engineering challenges involved with this project including phase delay, proper amplification, and correctly timing the output so that perfect cancellation occurs. The most difficult aspect of the design was the implementation of the cascading filters which select both specific frequencies and voltage levels to pass.





Our project was divided into three core blocks: Microphone + Amp, Noise Cancellation Filters, and Summing-Inverting amplifier. As seen in Figure 1 there are many sub-components to each block which will be further discussed in the design section below. In order for our circuit to function each block has certain requirements. In order to fit the industry standard of line level, the microphone preamplifier has to step up the mic's output so that the maximum input voltage correlates to a 2.2 V peak-to-peak signal at its maximum output. The filters must be designed in such a way that the cutoff frequencies single out the human speech range of 300 Hz to 3400 Hz. The filters must also not diminish the original signal by an appreciable amount, or they will have to be later amplified which would add more unnecessary noise to our anti-noise signal. And finally, the Summing-Inverting amplifier must set the gain such that the anti-noise signal is an inverted image of the noise signal at the same sound pressure level. It also must set the delay such that the anti-noise signal appears at the ear canal at the same time as the original.

2 Design

2.1 Design Theory

Before we could begin designing our circuit we first had to thoroughly research noise cancellation and how it can be implemented in a circuit. The first problem comes from the speed of the circuit as compared to the speed of sound. In figure 2 below we see that the microphone will receive the noise and process it before it is output to the speaker. We must invert the noise signal and output it precisely when the sound wave reaches the speaker cone. The problem lies in controlling the speed of the circuit to match the speed of sound and the distance between mic and speaker to ensure this timing.

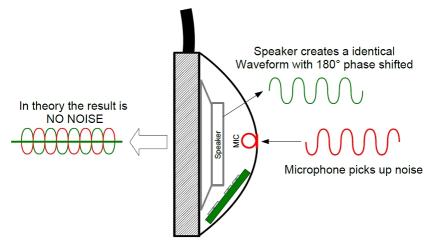


Figure 2: Noise Cancellation Theory [1]

2.2 All-Pass Filter

An all-pass filter is the simplest method to create group delay and also preserve unity gain. This method allows us to set the values of R, R1, and C in order to achieve any amount of delay that we choose. We set our initial distance from the headphone to the mic to be 2cm meaning our time delay will be $(0.02m)/(343m/s) = 58.3 \mu s$. This is calculated assuming that the speed of sound in air is approximately 343m/s.

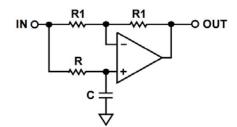


Figure 3: All-Pass Filter Design [2]

Phase Shift
$$(\phi) = -2 Tan^{-1} \left(\frac{RC}{2\pi F}\right)$$
 (1)
(2)

$$Group \ Delay = \frac{2 \ RC}{\left(2 \ \pi \ FRC\right)^2 + 1}$$

 $Delay at dc = 2 RC \tag{3}$

Given a phase shift of φ at a frequency = F

$$RC = 2\pi F TAN\left(-\frac{\phi}{2}\right)$$
(4)

The concept for complete noise cancellation is that the circuit outputs an anti-noise signal-- air pressure rarefactions at the peaks of the original noise and vice versa--at the same time as the incoming noise signal reaches the ear. In a simple circuit, this is achieved through two parts: A delaying amplifier to achieve accurate output timing and a summing-inverting amplifier to combine the audio signal intended to be heard with the created antinoise. This setup has unity gain as well as phase shift across frequencies, meaning it will cancel completely no matter what frequency or sound pressure level (SPL) we take in at our input. To begin we designed the full cancellation circuit as our starting block. This is divided into three simple sections: Microphone + amp, all-pass filter, and summing-inverting amplifier.

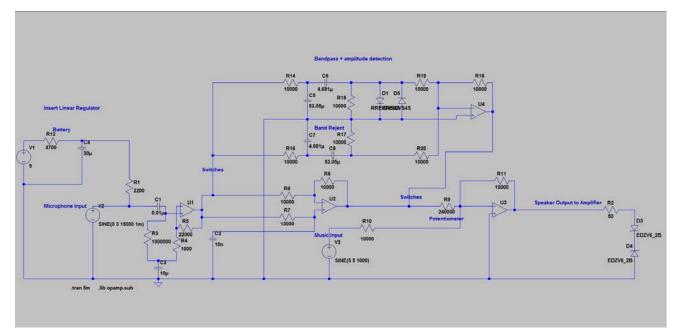


Figure 4: Initial LT Spice Simulation Circuit (Has since been improved).

Initial simulation of the all-pass circuit yielded excellent results. The noise signal added correctly to the anti-noise and generated an almost-perfect output waveform. Our goal is to recreate just the music signal. However, since our circuit is very simplistic in nature, the output was not perfect. Furthermore LTspice did not have the exact op amps we would be working with and as a result we needed to conduct actual tests before we could confirm that the circuit is functional.

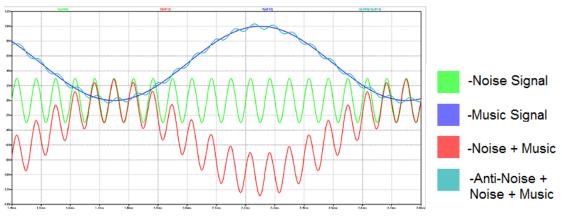


Figure 5: LT Spice Simulation for Full-Cancellation.

2.3 Peak Voltage Protection

The only danger involved with using this circuit is in the case of too much voltage being applied at the music input terminal. To prevent damage to the ears we implemented peak voltage protection at the output in the form of zener diodes. Peak to peak voltage for line level is 2.2 V. We used two zener diodes with breakdown voltages of 1.1 V each to ensure that our output voltage never exceeds 2.2 V peak to peak. Figure 6 shows the basic theory of the zener diode voltage clipping circuit.

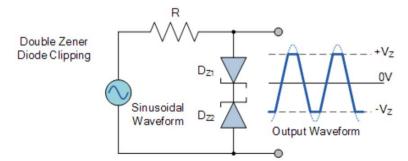


Figure 6: Diode Clipping for Peak Output Protection [11].

2.4 Threshold-Based Cancellation Filter

After building a basic noise-cancelling circuit we could now apply our modification to ignore cancellation lying in the human speech band and above a certain decibel level. This part of the circuit would run as an alternative to the all-pass filter since it will also be creating a delay. The human speech band we will be using is defined as from 300 Hz to 3400 Hz [3]. The true human speech range is much wider than this, but we chose this band because it will still remain intelligible while also cancelling out a larger range of frequencies. Our first idea was to sum together a bandpass filter with a band-stop filter while also limiting the output of the bandpass using a clipper circuit. We chose to use the Twin-T band stop design due to its simplicity for implementation. While ideal filter frequency response would have perfectly straight lines for the cutoff frequencies, real filters have a curve where they fall off from the stop band to the pass band. Figure 7 shows this effect. Our implementation uses a bandpass filter and a Twin-T band stop filter to divide the frequency content of the noise received at the input.

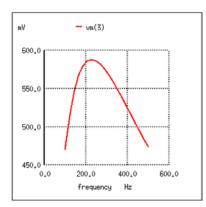


Figure 7: Simulated bandpass filter using R1 = 200 Ω , C1 = 2.5 μ F, C2 = 1 μ F. [4]

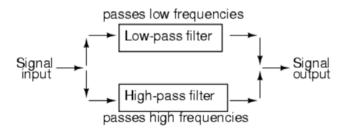


Figure 8: Block diagram of band-stop filter. [4]

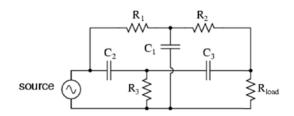


Figure 9: Twin T circuit diagram. [4]

$$R_1 = R_2 = 2(R_3) \tag{5}$$

$$C_2 = C_3 = (0.5)C_1 \tag{6}$$

$$\mathbf{f}_{\text{notch}} = \frac{1}{4\pi R_3 C_3} \tag{7}$$

The bandpass clipper circuit is the core part of the uniqueness of the circuit. It was designed using a simple high-pass into low-pass circuit with a clipper attached in series in order to limit the decibels. The output from our microphone preamp was 67.5 mV peak to peak at 70 dB (SPL) meaning we want to remove any cancellation above that voltage. 70 dB (SPL) being our reference for human conversation. Diode clipper circuits function very similarly to the

voltage protection output where the voltage is simply cut off at a certain threshold. Figure 9 shows this effect. We selected the value for R1 and R2 to be 10 k Ω since that was the recommended value for resistance when working with the LT1056 Opamps. The values for the respective capacitance was chosen based off of equation (8) with f_{cutoff} being set to 300 Hz for lowpass and 3400 Hz for highpass.

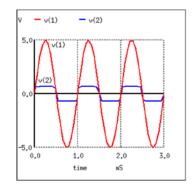


Figure 10: Clipper circuit simulation. [4]

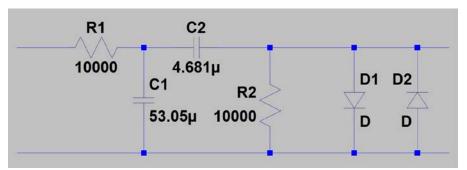


Figure 11: Bandpass clipper circuit in LT spice.

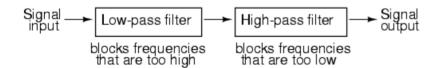


Figure 12: Block diagram for bandpass filter. [4]

$$\mathbf{f}_{\text{cutoff}} = \frac{1}{2\pi RC} \tag{8}$$

To combine these two filters together we used a non-inverting summing amplifier. Ideally we wanted this part of our circuit to be unity gain, however that would not be possible. Instead we chose to have this part of our circuit have a gain of 2 through Rf = Rf1 and making the gain variable later on with a potentiometer in Section 2.6.

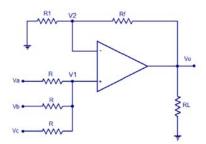


Figure 13: Summing Amplifier [5].

$$Vo = (1 + (Rf/R1)) ((Va+Vb+Vc)/3)$$
 (9)

2.5 Microphone and Microphone Amp

The first stage is to power the microphone in accordance with its datasheet. The datasheet did not specify the output voltage level, but this was determined through testing to have a small DC offset. In order to remove this DC offset we sent the output signal through an amplifier which functions as a unity gain amp for DC and a non-inverting amp with a set gain for the rest of the frequency spectrum. This allows the output of our noise-cancelling circuit to be centered around 0V to be processed later in the circuit. The datasheet specifies $R = 1 \text{ k}\Omega$ and $C = 0.01 \mu\text{F}$ [6].

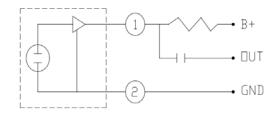


Figure 14: Connection diagram for ECM-60PC-R microphone [6].

The microphone preamp follows the same configuration as a standard non-inverting amplifier. A capacitor was added in series with R1 in order to completely remove any remaining DC offset. We set our gain to be 20 in order to boost the output to our desired level.

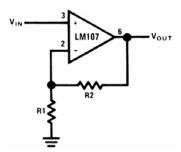


Figure 15: Non-inverting amplifier [7].

$$V_{OUT} = \frac{R1 + R2}{R1} V_{IN}$$
(10)

$$R1 \parallel R2 = R_{SOURCE}$$
(11)

2.6 Summing-Inverting Amplifier

The summing-inverting amp is where the noise cancellation takes place. The input music signal is added to the noise signal and inverted to be sent to the headphones. Following the diagram, our V1 is the output of the filter circuit and R2 is the music input. R1 is chosen to be a 200 k Ω potentiometer and R2 is set to be 10 k Ω . By making R1 variable we can easily change how much cancellation is taking place in the circuit. Since we are not using a more complicated form of noise cancellation that uses a microphone at the speaker to detect the cancellation output level we have to use this more controllable design instead. As in Figure 4, this adds and inverts the two signals together to be sent to the speakers. This stage also acts as our speaker amplifier since we chose R2 to equal R4 to create unity gain from the music source.

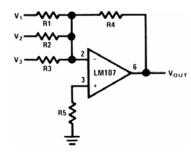


Figure 16: Summing-Inverting Amplifier Diagram [7].

$$V_{OUT} = -R4 \left(\frac{V_1}{R_1} + \frac{V_2}{R_2} + \frac{V_3}{R_3} \right)$$
(12)
R5 = R1 || R2 || R3 || R4 (13)

2.7 Bluetooth Chip and Music Input Mux

Our bluetooth datasheet indicates that it requires 3 V input voltage to function and outputs an analog signal at the same level as an aux cable from a phone. Minimal work was required here and this module was added on with the intention of learning more about bluetooth. To easily switch between bluetooth and aux connections we implemented a simple mux with a control switch.

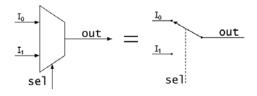


Figure 17: Multiplexer functionality diagram [8].

2.8 Power System

We needed a power system to meet the following requirements: op amps require +5 V and -5 V minimum across the terminals in order to function properly, bluetooth chip needs 3 V, microphone circuit need +5 V and ground, and the filters need a 0 V ground reference to refer to. To solve this problem we put two 9 V batteries in series to create an 18 V driving voltage which was then sent to a 12 V regulator. Voltage divider would be insufficient since our circuit involves numerous other resistors. Thus we used an opamp to create +6 V, -6 V, and virtual ground using the circuit in figure Y. We chose 1 M Ω as the resistance for R in order to reduce power consumption by the resistors. V_{cc}/2 then becomes virtual ground while V_{cc} and ground in the diagram become +6 V and -6 V respectively. For the bluetooth chip we use a 3 V voltage regulator which is powered by the +6 to Virtual ground portion of the opamp.

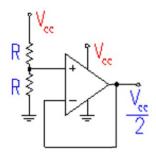


Figure 18: Virtual ground generating amplifier [10].

3 Design Verification

The tests conducted for each section can be found in the Requirements and Verifications table in Appendix A.

3.1 Filters

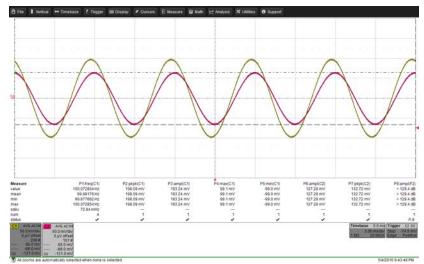


Figure 19: Summing-Inverting amp at 100 Hz Input signal level (Yellow): 198 mV_{pp} Output signal level (Pink): 133 mV_{pp}



Figure 20: Summing-Inverting amp at 430 Hz Input signal level (Yellow): 215 mV_{pp} Output signal level (Pink): 111 mV_{pp}

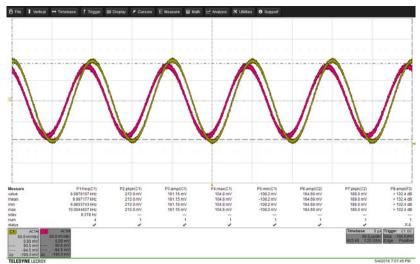


Figure 21: Summing-Inverting amp at 10 kHz Input signal level (Yellow): 213 mV_{pp} Output signal level (Pink): 189 mV_{pp}

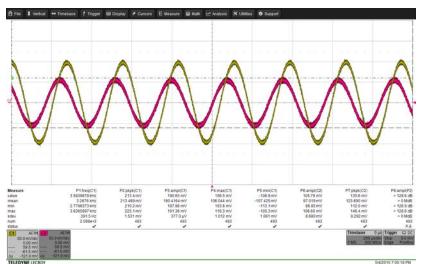


Figure 22: Summing-Inverting amp at 3.6 kHz Input signal level (Yellow): 213 mV_{pp} Output signal level (Pink): 123 mV_{pp}

These four tests were conducted to show the functionality of the summing inverting amp. We see from figures 21 through 24 that the output of the noise is reduced around our bandpass frequencies 300 Hz and 3600 Hz. These were tested to prove our noise cancelling circuits' functionality.

The tests were done by sending a sample waveform through the filters at specific frequencies which were expected to pass, then changing the frequency until the -3dB point was reached.

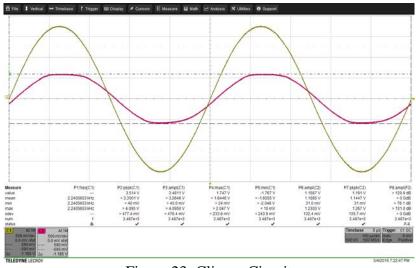


Figure 23: Clipper Circuit Input signal level (Yellow): 3.3 V_{pp} Output signal level (Pink): 1.14 V_{pp}

This graph shows the output of the clipper circuit and its clipping functionality past a certain voltage threshold. It will not exceed the roughly $1.1V_{pp}$ as set by our diodes.

This test was done by increasing the voltage of the signal from the previous tests until the output signal hit its maximum, this is the maximum level of human noise which will be passed to the cancellation module.

3.2 Summing Inverter

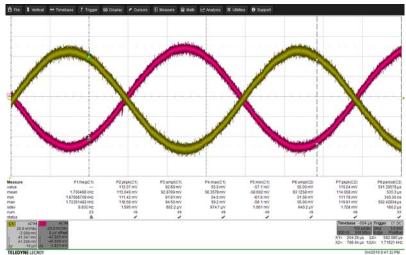
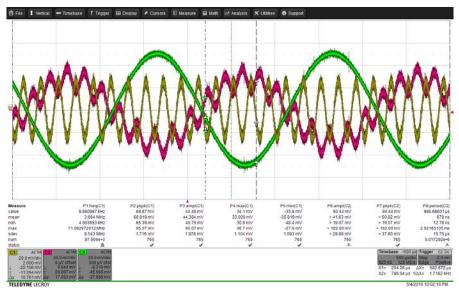
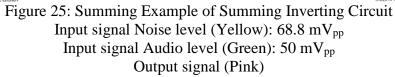


Figure 24: Inversion Example of Summing Inverting Circuit Input signal level (Yellow): 115 mV_{pp} Output signal level (Pink): 115 mV_{pp}





This figure is a similar test as conducted in our simulation in Figure 5. This shows that our summing-inverting amplifier works correctly as specified.

This test implemented both output of the function generator at a mock noise level (5kHz) and a mock audio level (500Hz). The two signals were successfully inverted and added together.

4 Costs

4.1 Parts

Item	Quantity	Cost	Total
Bluetooth Chip (RN-52)	1	\$25	\$25
Opamps (LT1056)	6	\$1.95 each	\$11.70
Test Headphones (Apple Earpods)	1	\$29.00	\$29.00
Resistors and Capacitors:	Resistors: Potentiometer (up to 500000Ω) $15x 10 k\Omega$ $13x 1 k\Omega$ $2x 100 \Omega$ $1x 1 M\Omega$ Capacitors: $4x 10 \mu$ F 2x 10 nF $2x 50 \mu$ F	~\$5	\$5
Diodes	2x Zener 2x 0.7 V Threshold	\$1	\$1
Battery Pack(9V)	2	\$1.25	\$2.50
Microphone (ECM-60PC-R)	2	\$1 each	\$2.00
Aux Connector	1	\$0.50	\$0.50
Linear Regulators	1 x 12 V 1x 5 V	\$0.75 each	\$1.50
Headphone Amplifier (LM4929)	1	\$1.00	\$1.00
2-1 Mux	2	\$0.50	\$1.00
TOTAL			\$80.20

4.2 Labor

Name	Hourly Rate	Total Hours	Total Cost
Nick	\$40.01	225	\$9002.25
CD	\$40.01	225	\$9002.25
David	\$40.01	225	\$9002.25
TOTAL	\$120.03	675	\$27006.75

5 Conclusion

5.1 Accomplishments:

We managed to create a mostly functional system for noise cancellation with adjustable cancellation levels and the option to pass through human speech. While each module was designed with the intention of being as simple as possible, their interactions were very complex. The entire circuit worked as intended up until the summing inverting amplifier. Our original design had the output from the summing inverting amplifier go through a voltage protection circuit and supply power directly to the headphones. Unfortunately, the signal disappeared when the headphones were plugged in. The only explanation that we could come up with was that our opamp needed more voltage to properly amplify the signal to power the headphones or the speaker amp was damaged.

5.2 Uncertainties:

There were a lot of challenges with creating this project. What works in a simulation will not always work in the real world. While the full cancellation circuit works as intended, our modification had many problems with functionality. The filter circuit functions properly by itself, but when attached to the circuit it did not work as intended. As shown in design verification the circuit did have a dip in the 300 Hz to 3600 Hz range, which was desired to prevent human speech range from being inverted.

5.3 Future Work / Alternatives:

This project was initiated with the intention of learning more about noise cancellation and how to modify the basic circuit design. The future of this prototype is to get closer to our original hearing-aid style concept where the circuit will be completely enclosed in the earpieces and difficult to distinguish from far away. The technology for in-ear bluetooth hearing-aid style headphones currently exists and is commercially available. We wanted to take this concept to the next level and add selectable noise cancellation. We believe the trend from smartphones will lead to the eventual popularity of smart-headphones. Our project is a step towards that future.

5.4 Ethical Considerations:

Our project has two primary ethical and safety considerations. The first is damaging the user's ears if the output voltage is high enough. We dealt with this problem by limiting the output voltage to the headphones to 2.2 V peak to peak. Using a standard aux output from a phone the output voltage will be unable to become louder at the output than if the headphones were plugged in directly to the headphone jack. The second consideration is the lack of awareness induced through noise cancellation. For example: if an alarm is going off at a certain frequency and gets cancelled by this circuit, the user will have a diminished ability to hear it. While ideally our circuit can cancel any sound at the same level as our output in SPL, the real-life functionality will allow any alarm to pass through. These are stated to be in accordance with points 1 and 9 of the IEEE code of ethics found in Appendix B.

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APPENDIX A

Module	Requirements	Verification
Noise cancelling circuit	Less than 3 dB loss of volume from audio input (specifically the volume from the input audio, after the anti- noise is added)	 Voltage measurements on oscilloscope can determine this. 3 dB correlates to 50% power Measure audio voltages before and after summing inverting amplifier
	Noise should be reduced by at least - 15 dB on average in 60-250 Hz range, and in 5000-15000 Hz Range	 Generate a sinusoidal signal of 60 and 250 Hz of known amplitude. Measure output of the -180 degree phase shifted sinusoid and convert ratio to dB Use oscilloscope to sum together input and output signals Verify that it is -15 dB or less after conversion Repeat 1-3 for 5 kHz and 15 kHz.
	From 300-3400 Hz Noise below 60 dB (SPL) should be reduced by at least -20dB (voltage) And noise above 60 dB (SPL) should be reduced by no more than -15dB (Voltage)	 Measure a sinusoidal tone at 300 and 3400 Hz below 60 dB (SPL) and play it next to the microphone. Measure voltage level of noise after BandPass filter, and after the summing inverting amplifier. Use oscilloscope to sum signals together and verify that the reduction is at least 20 dB Repeat for a tone with SPL above 60 dB, Verify that reduction is less

		than or equal to 10 dB
Bluetooth Chip and Amplifier	Peak analog audio output through amplified embedded DAC should be within -6 dB of maximum input from auxiliary connection.	 View sample song on aux output and measure maximum output voltage compare to same song output from BT out and convert to dB. dB ratio should be equivalent or +/- 1 dB
Power Supply	Vout = 12V +- 2V at 14 mA	 Attach a 1 kΩ resistor as load. Attach oscilloscope across the load. Apply 18 V across the voltage regulator. Ensure that the output voltage is between 14 V and 10 V.
Music Input Mux	Mux correctly switches between AUX and Bluetooth with AUX being the default input.	 Play music through AUX and Bluetooth separately. Use an oscilloscope to confirm that the output switches with each input. And if both are input at once, Auxilliary input prevails
Full System	SNR above 95 dB through the 20 Hz - 15,000 Hz range	 Input frequency sweep at microphone and voltage measurement at speaker input Re-record voltage output level with no input frequency sweep Divide max voltage of sweep test to max voltage of noise test and convert to dB.
	Total Harmonic Distortion+Noise is -74 dB over 200-5000 Hz	 Input signal with known amplitude and frequency Measure Output with

fundamental

APPENDIX B

IEEE Code of 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;
- 2. To avoid real or perceived conflicts of interest whenever possible, and to disclose them to affected parties when they do exist;
- 3. To be honest and realistic in stating claims or estimates based on available data;
- 4. To reject bribery in all its forms;
- 5. To improve the understanding of technology; its appropriate application, and potential consequences;
- 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;
- 7. To seek, accept, and offer honest criticism of technical work, to acknowledge and correct errors, and to credit properly the contributions of others;
- 8. To treat fairly all persons and to not engage in acts of discrimination based on race, religion, gender, disability, age, national origin, sexual orientation, gender identity, or gender expression;
- 9. To avoid injuring others, their property, reputation, or employment by false or malicious action;
- 10. To assist colleagues and co-workers in their professional development and to support them in following this code of ethics.