Noise Canceling Headphones with Hearing Test and Equalization

High Level Design Document

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# Introduction

The ability to properly perceive audio is a precious gift. Properly functioning hearing allows us to communicate efficiently with others, experience rich and beautiful music, and sense important information about the world around us. However, our hearing is one of our most delicate senses, prone to significant and irreparable damage. Those who experience significant hearing loss lose the ability to perceive much of the beauty and character of the world around them. We would like to help these people by providing them with a product that tunes itself to their personal hearing and allows them to reclaim some of the auditory experience that they have lost. Just imagine buying this gift for a grandparent who has become hard of hearing. Imagine for them the experience of the sounds of their youth in a better way than they have ever heard the music. It would bring back memories and the feeling of awe that you get when listening to a fantastic piece of music.

The problem attempted to be solved is the annoying experience of rolling over your headphone wires with your desk chair. The problem is getting tangled up as you try to do anything like cook or clean. Reaching up and around your body pulling off your headphones mistakenly and interrupting your experience is a constant struggle. The problem is the inherent decay of the human experience through physical trauma or age. The problem is that your headphones always fall off as you try to go on a run, or they are just way too uncomfortable to use while exercising.This product seeks to turn back the clock on damage done, and stop the incessant distractions away from a perfect listening experience.

The idea of the bluetooth headphones was really to just do something that was really cool. Build something that was difficult and involved some incredible technology and also produce a product that we would want.

In order to build the headphones we had to solve the problems. The first problem to solve would be the implementation of noise cancellation. This aspect would be taken care of by taking an incoming signal i.e. ambient noise from the environment and converting it into a digital signal by way of some transducers known as microphones. This digital signal could then be inverted by use of an inverting amplifier, or internal processing in a dsp chip. The next problem to be solved is the removal of wires. The typical audio jack was to be replaced by a bluetooth signal built for audio functionality. This would eliminate the wire problem that I mentioned earlier. Audio data would be transmitted via high quality audio A2DP standard. This is also known as “Advanced Audio Distribution Profile”. The transfer of audio signals between any two devices can be made if they both support this standard.

Another aspect to be considered is the implementation of digital equalization. The nature of the equalization curves are to be discovered through an audio examination delivered by an phone application that would go along with the headphones. The examination would test the person's ability to hear at various frequencies across the auditory spectrum from 20 Hz all the way up to around 14-15kHz. This is the range of the audible hearing spectrum. It can go as high as 20 kHz but for all purposes 14-15 kHz will do the job. These different data points would then be compiled into a set of equalization values that would be used by our digital signal processing chip to implement the curves on the music played. The processing chip must be powerful enough to handle the need to equalize and boost sound signals as well as process the data coming from the microphones, invert the signal and boost it to appropriate levels to cancel out the ambient noise.

# Problem Statement and Proposed Solution

Our goal is to provide those listening to our headphones with an accurate and clear perception of whatever they choose to listen to. There are a number of obstacles to this goal that we will address.

First is the issue of environmental noise. A low signal to noise ratio can be distracting, covering up more subtle elements of the audio and making voices hard to understand. Passive noise canceling, realized by earbuds that physically block sound from entering the ear, is one solution to this problem, but it introduces many problems of its own. In our design, we will focus on an active noise canceling system. In its most basic form, active noise canceling works by picking up external noise with a microphone mounted on the outside of the headphones and outputting the inverted version of the signal from the headphone speakers with the exact amplitude of the incoming noise signal. If timed properly, the two signals will cancel and the noise will not be perceived.

Second is the issue of individual hearing loss. Hearing loss has a variety of causes, from exposure to loud noises to natural aging. Often this loss is limited to a specific frequency range, resulting in a gap in the individual's perceptual frequency response. In order to correct for this as much as possible, we will implement an app-based hearing test that the user will take when first setting up the headphones. This will generate a profile that will allow the headphones to boost the frequencies the user has trouble hearing, giving them an experience similar to a person with normal hearing.

# System Requirements

1. To invert incoming ambient sound waves and reproduce the inverse of the waves within 110us +/- 10us.
2. To provide battery life that would last for at least 2 hours at a reasonable volume.
3. To communicate with another device via the Advanced Audio Distribution Profile. This is the means of playing music through a bluetooth technology system.
4. Low noise/high quality. The music streamed over the bluetooth connection must have as little a quality loss as possible making it almost indiscernible to tell the difference between a wired connection and the bluetooth based connection.
5. To apply equalization of a specified nature on a music profile and ensure its accuracy and consistency.
6. To provide a hearing equalization exam that tested at a range of frequencies to determine hearing loss and rectify the loss to the point where the original sound was intended to be heard.
7. To provide a headset that offered comfort and durability that could be worn for hours.

The headset would be sized such that it could fit all of the hardware without being cumbersome and off balance.

**5. Detailed Project Description**

## **5.1 System theory of operation**

The baseline aspect of the system is the production of sound. The production of sound begins at its recording. A voltage is induced in a microphone which is simply described as a pressure sensor. It receives the longitudinal waves and acts as a forced oscillator. The frequencies of the physical oscillation are converted into voltage signals by way of Maxwell’s equations, namely Faraday’s law which governs the way in which voltages are produced by occurrences of varying magnetic flux in an arbitrary closed loop.

The voltages produced by the microphone are then sampled at more than twice the max frequency of the highest frequency sound wave that the microphone will interpret. This means that the sampling frequency of the digitally recorded signal would be at least 30kHz or so to prevent aliasing. The minimum requirement of sampling at double the frequency of the signal to be recorded is known as the Nyquist frequency. Many problems arise in this stage of the music process. Especially the fact that the device will have resonant fourier series components that could end up interfering constructively with other signals producing a recognizable amount of distortion.

The recording hardware will not only be equipped with the appropriate sampling frequency, but it must also have a high enough resolution. The resolution that I am referring to is the number of bits that will be used to describe the range of voltages produced by the microphone. The simple example to understand this is that the number of different values to be stored by a bit resolution is equal to 2 raised to the power of the number of bits available per sample. So if you have 8 bits available for sampling the number of values you can use to represent the range of input voltages would be 2^8. This is 258 different available values to interpret voltage signal levels.

"A published estimate [Stevens, S.S., and Davis, H. *Hearing, its psychology and physiology*. New York: John Wiley, 1938] gives 330,000 as the approximate total number of monaurally distinguishable tones of all frequencies and intensities." This value indicates that a 32 bit resolution would be the absolute maximum required resolution to discern all available sountds to the human ear. At least with respect to monaurally distinguishable tones. This would be 1000 times more available recording values compared to the human auditory perception.

Now that a digital signal is recorded with a hypothetical 32 bit resolution 30 kHz minimum microphone and recorder, we need to compress the data. In order to efficiently send the data by way of A2DP bluetooth communication profile we need to compress the audio data. This is done by recognizing that the psychoacoustics of the human ear can not perceive as much of the data that is recorded by these high precision digital devices. It is also done by reducing perceptual redundancy and the frequency components that are outside of the human auditory range of 20 - 15 kHz. The very low and high frequencies can be the most easily removed as the optimal range of human hearing is from 200 Hz to a few kHz.

The compressed audio data is then transmitted by bluetooth rf signal from a phone or audio player that is bluetooth compatible to our headphone’s bluetooth processor. Bluetooth is a heavily involved software and hardware protocol that allows for the wireless communication of data. I will provide an introductory explanation to its functionality for the purpose of this document. For bluetooth devices to connect certain protocols are followed much like the examples of SPI and I2C that we know from senior design 1.

First, there is an agreement between the two bluetooth devices. The agreement specifies how much data will be sent per sending period, when the data will be sent, and which devices are masters and slaves. Also included in the agreements is frequency hopping protocols where the master and slave agree upon a certain sequence of frequency changes. By having constantly changing frequencies in the communication protocol there is less likely a chance for interference and it increases the amount of data that can be sent over bluetooth. A master bluetooth master device can connect to up to 7 slave devices each with a settable address.

1. How much data will be sent at a time?
2. How will devices speak to each other (protocol/profile)? Bluetooth provides agreement on how this will be done. Products have to agree on when bits are sent, how many will be sent at a time, and how message confirmation and checking will occur

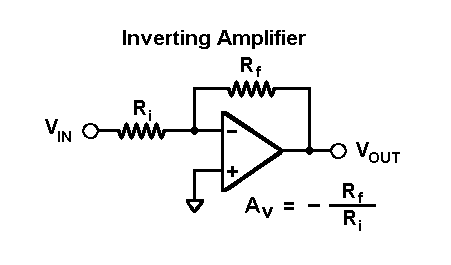
Bluetooth has two layers. The first layer called the lower layer makes up the hardware and radio transmission components of the devices. This describes physical characteristics of the radio. This is the layer that performs modulation of the 2.4 GHz band. Above the radio is the link manager protocol also known as the LMP. This aspect of the bluetooth stack is responsible for data formatting, too and from the radio. This part of the layer defines the characteristics of the communication. From timing and framing to the size of packets and flow control. This part of the protocol also receives instructions from the given HCI commands.

HCI is also known as the host controller interface. This acts as an interface between the host controller and the hardware that is to be implemented. The last major component of the stack to be described is the L2CAP (logical link control and adaptation protocol) which performs communication between upper and lower layers of the stack. It is the data control center controlling where and when data is to be transfered.

Each different type of communication to be sent will have an associated communication profile. This profile describes how to implement bluetooth for a particular usage. The one that we are interested in is the A2DP profile as mentioned above it is the Advanced Audio Distribution Profile. For example the A2DP applies the interconnection of the layers listed above by taking advantage of the LMP and L2CAP protocols. It uses these to control the radio interface and transmit the digital audio stream. The link below provides the specification for A2DP communication:

<https://www.bluetooth.org/docman/handlers/DownloadDoc.ashx?doc_id=260859&vId=290074>

Inversion of an input signal

Signal inversion can come in different forms. Electrical engineering training points first in the direction of an inverting op - amp that will produce an output signal that is the inverse with a controllable amount of gain. A visual depiction of the set up is bel

The gain at V out is proportional to ratio of -Rf / Ri. All you have to do is make this ration a little more than - 1. This is determined by discerning the range of input voltages and current ratings of a microphone and optimising it so that the max voltage that we would see, would be close to the max voltage allowable by the audio processor. The audio processor could then amplify even more as the inverted noise signal gets driven to the speaker drivers.

The voltage signal gets sent to the speaker drivers. It is here that the electrical signal is turned back into a sound wave. The voltage is applied to a coil that in turn applies a magnetic force to a diaphragm in the speaker. This diaphragm will oscillate with all of the frequency components presented in speaker coils, in addition to the extra frequency components created due to non-linearities. The mark of a good speaker driver is how well it is able to reproduce the original sound which means any non-original frequency components in the music would be minimized. We will be sending both the audio signal and the inverted signal to the speaker drivers and the speaker drivers will have no problem playing all of the frequency components simultaneously as the movement induced in the diaphragm is a superposition of all of the voltage signal components presented in the output waveform.

# System Block Diagram

## Overall System:

* Headset unit
* Main circuit board
* Battery and charging circuitry
* Smartphone app
* Bluetooth interface

## Subsystem and Interface Requirements:

### Headset

The headset unit itself will consist of a plastic frame, two drivers, two external noise canceling microphones, and two internal error correction microphones. The microphones will be digital MEMS microphones. The exterior microphones must be mounted at a well-defined distance from the speaker drivers to properly time the production of the signal. The headset will also act as a case for the main circuit board. To accommodate this requirement, the speaker cavities will be 3d printed.

### Main circuit board

The main circuit board will include our primary microprocessor and any supporting components. Depending on the choice of microprocessor, a variety of components including a bluetooth adapter, DACs, ADCs, amplifiers, and a crystal oscillator may be required. The primary configurations under consideration will be detailed in the high level design decisions section. Regardless of component selection, the board itself will have to be made small enough to fit in the headset. The firmware downloaded to the main processor will need to implement a bluetooth stack capable of interfacing with our app and streaming audio. It will also need to implement our equalization filter.

### Battery and charging circuitry

The headset will be powered by a lithium-ion battery embedded in the opposite speaker cavity from the main circuit board. The battery will have to be small enough to fit in the headset, but high capacity enough to power the device for several hours. The recharging circuitry, if such functionality is not built into the main chipset, will be on a separate board next to the battery.

### Smartphone app

Our smartphone app will run on Android and provide basic configuration functionality along with a hearing test. On initial setup, the hearing test will run and generate a profile for the user. This profile will be downloaded to main microprocessor over bluetooth and used to configure the main equalization filter. Additional functionality, as detailed in the future enhancement requirements section, will also be included in the app and any additional configuration will be downloaded to the headphones over bluetooth.

### Bluetooth Interface

The primary wireless interface between the smartphone app and the main microprocessor will be bluetooth low energy for data transmission, and bluetooth classic for audio streaming. The low level functions of the interface will be implemented in hardware and built-in device firmware.

## Future Enhancement Requirements

In future releases of our product, there are a couple elements we would like to add. First is a microphone input, mainly meant for talking with others online or over the phone. We would also like to add color changing LEDs for aesthetic purposes. We would also like to add a hearing aid feature that would amplify human speech and attenuate all other frequencies. This would allow the headphones to be advertised to multiple demographics, from young gamers to people who work in loud environments to the elderly.

# High Level Design Decisions

### Headset

The headband, pads, and speaker drivers for our headset will be taken from the Sennheiser HD 201 headphones. The speaker cavities will be custom 3d printed to allow the main circuit board and battery to be embedded. The ICS40619 MEMS microphones that will be used have been found in the senior design IC bin.

### Main circuit board

The most important component on the main circuit board is the main microprocessor. We are still looking into bluetooth capable from providers such as Qualcomm and TI. A major consideration is that many models support bluetooth low energy for data transfer, but do not support bluetooth classic for audio streaming. Qualcomm's CSR8xxx series is a good option. If this SoC is chosen, very few additional components will be necessary, as it is designed to be integrated into bluetooth noise canceling headphones and has most of the necessary functionality built in.

### Battery and charging circuitry

A 1800 mAh battery with the correct proportions to fit in the headset was found in the senior design room. The BQ2057CSN li-ion or li-po battery charging IC was also found in the senior design room and will be used to charge the battery.

### Smartphone app

The smartphone app will have a couple different features. The first is a way to create and name a hearing profile. At the end of testing your hearing you can download the profile to the headphones. In addition there will be a page to choose different equalization settings on top of that for different types of music. Lastly, if we have time, the app should interface with various music playing apps (namely Spotify and Play music).

### Bluetooth Interface

If the Qualcomm SoC is selected as the main processor, the Qualcomm development environment and bluetooth stack will be used to program the main board. The smartphone app will use Android's bluetooth API to interface with the smartphone's bluetooth hardware and interface with the main board.

# Open Questions

As of right now, there are three major issues that will need to be addressed. The first is the speed of the active noise cancellation processor. The DSPic we are using right now is not fast enough. As such, the distance between the microphone and the driver needs to be .2 m apart to work optimally. For our final product next semester, the processor needs to be faster so that the microphones can be incorporated into the headset.

The next foreseeable issue is the signal processing and bluetooth connection for sending music to the headset from the phone. The connection has to be fast enough for high quality music, have a reliable connection, and be able to equalize the music. As such we may decide to use different microprocessors for the music than the ANC.

The final issue is incorporating everything into one headset. As of right now, we plan on 3D printing the chassis for the chips and drivers and putting that into a set of headphones we take apart.

# Major Component Costs

The components we need for our boards are not very expensive and many of them already are in the senior design room. As such we probably only need 20 dollars for the various components. The board itself will most likely cost around 40 dollars at the most. Batteries and a charging jack will most likely be around another 20. And conservatively, whatever microprocessor we get should be less than 30 dollars. 3D printing through the engineering department is free and we currently have a set of headphones that we plan on stripping for parts. So even being very conservative our budget is 110 dollars.

# Conclusions

Overall, there will be some issues we will encounter in our mission to design noise cancelling headphones. However, we have a fairly good idea of what they will be and what we can do to surmount them. We are excited to begin the project in earnest next semester.

**References:**

Stefan Liebich et al., "Active Noise Cancellation in Headphones by Digital Robust Feedback Control," accessed at <https://www.eurasip.org/Proceedings/Eusipco/Eusipco2016/papers/1570251613.pdf>