Karaoke Portable System
(KPS)

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04/27/2018

FINAL SUBMISSION

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1. Executive Summary

Karaoke is a relatively newer form of entertainment in America. However, its origins lie within the Asian market, starting with the first Karaoke machine made in the 1970's. The basic premise of Karaoke is to provide a backing track for artists or hobbyist to sing songs without the need for a complete band. Karaoke has evolved from a few hundred units being sold in Japan, to having in-home karaoke machines as home theater systems across the globe. With the increasing popularity in American culture, one can often see Karaoke as a form of entertainment in bars and nightclubs. This is often accompanied by lighting effects and multiple monitors to display lyrics to everyone in attendance, including the performer. Karaoke has also been seen in popular video games such as 'Guitar Hero' and 'Rock band' bringing a gaming entertainment to singing along with popular songs. The Karaoke Portable System (KPS) brings the essence of singing along with numerous songs in a portable package. With the KPS, no longer will it be required to go to a 'Karaoke Night' at a bar, or will it be limited to an at-home experience. The KPS will bring the technology of Bluetooth to Karaoke allowing for Karaoke to exist in any moment, and at any time.

The KPS will be a Bluetooth wireless microphone that will have speakers built into the device. It will connect to any cell phone (Apple or Android platform) as a headset. The progression of Bluetooth technology has allowed significant reduction in power consumption, provides a stable connection, and quickly transmits signals. The effective range of the KPS will be up to 10 meters. The background music to be played from the connected phone will output from the two built in speakers on the side of the KPS. This will allow for the performer to hear the music and will also mix the performers voice. Sound effects applied to the voice will only be echo. However, one will also have the ability to adjust frequency, bass, and volume by having potentiometer type knobs on the interface. The KPS will be made of clear plastic to easily display the Printed Circuit Board (PCB). Music will easily be connected by installing one of many popular applications, such as YouTube, on the mobile device. Convenience, size, portability, and cost are immediate marketing points for the KPS. Aside from the entertainment value the KPS can bring, it can also be a greatly effective aid for instructors, as well as needing to amplify a voice for speaking in meetings, or even used with tour guides. LED lights shall be implemented into the KPS. It is not certain yet if the LED lights will be built into the microphone itself, or act as a separate entity. The LED lights shall provide a direct mapping of the 12-note western scale to an individual color. Aside from an effective lightning show, this could also aid in instruction and guidance for aspiring singers.

Several constraints must be met for the KPS to be a viable mini Karaoke machine. One concern is the weight. The device itself should be comparable to that of an actual microphone. This constraint may exclude LED lights. The lights may be separate from the device itself, especially for prototyping. The main design focus is to allow the KPS to be lightweight and portable, and at the same time provide high fidelity sound quality. The challenge that will be faced is having a speaker, a microphone, and a play along track working simultaneously. To aid in sound quality, noise reduction will play an important role in design. The range of frequency that an average human voice can sing is from 100 Hz to 10 kHz. Any frequencies existing outside of this given range, can be filtered. The Bluetooth capability should extend to all portable devices, and not be just limited to a smartphone. A standard protocol for sending and receiving data through a 2.4 GHz wireless link will be used. This is ideal for short-range, low-power, and low cost wireless transmissions between electronic devices. The KPS will have a 5V DC battery and might include a USB port for additional charging to the device. The KPS will also have LED lights that will display an array of colors that correspond to the music while the singer is performing.

The KPS will strive to capture an audience of performers or singers with easy to use functionalities. There will be one button for powering the device on and off and will also have the capability of listening to music by connecting to Bluetooth with a mobile device. The overall objective of this project is to have a microphone that combines portable audio and voice into one device for karaoke while maintaining a low-cost design.

2. Project Description

The idea of karaoke was inspired by the Asian culture. Motivated behind the inkling of bringing about a portable karaoke microphone that could bring together family, friends and everyone in the community to enjoy a time of singing. To accomplish the design of the KPS goals and objectives needed to be put in place but behind these goals was the motivation behind the design to make the KPS a reality.

2.1. Project Motivation

For many performers, the everyday casual singers or any individual ready to sing to their favorite songs in front of an audience a powerful and unique microphone was required to make performing possible. While performing the singer needs to put on an impressing show, with this the visual aspect comes into play. By incorporating an array of LED lights, the performer can adjust at will. The multi-functional LED display will allow the performer to see the key in which they are singing in, represented by an LED light color mapped to the specific key. This will be a guide for singers to stay in the key of the song. The performer will also be able to set the tempo of the LED display on the microphone with a push of a button adding a unique dynamic to a karaoke microphone. With this product we hope to give singers a microphone that will bring out a great performance.

2.2. Project Goals and Objectives

For the KPS to be a practical mini Karaoke machine we need to meet the constraints to make the audio clear and have visible clean lyrics display from a device using a wireless connection. Many singers have different styles of performing. We want to provide performers the ability to access their favorite songs on any device wirelessly. One of the focuses will be to include Bluetooth capability that should extend to portable devices. The lyrics will display from the portable device to allow the singer to follow along to their favorite songs. Singers will also have the versatility of effects to enhance their performance by use of digital effects such as delay and reverberation which will be consist to the song.

While performing the singer will not only be able to sing to songs with several effects but also have the enhancement of a great light show to capture the attention of the audience. By incorporating a multi-functional LED display, the performer can choose flashing patterns to display while singing, flash LEDs to map to the key of their voice or set a flashing tempo. The Display can also be used as a fun display to add dimension to the performance. With the KPS the singer will have the ability. to connect any devices to their microphone wirelessly to gain access to any video or music sharing site or application to perform karaoke style at a park, home gathering, parties, or at any place. And wild an audience with an eye-catching display to give a performance of a lifetime.

2.3. The Engineering-Marketing Tradeoff Matrix

The house of quality or the engineering-marketing tradeoff matrix provides and classifies the desires of the customer while identifying the important key engineering requirements which will meet the desires of the customer and deliver an overall well-made technical product. The diagram below demonstrates the tradeoffs for the Karaoke Portable System (KPS) between marketing and engineering requirements. The purpose of the diagram is to provide indication as to what the KPS is targeting for its consumers.

Figure 1 depicts the house of quality matrix. Along with the marketing requirements and the engineering requirements of the system. The relationship of both requirements are presented in the figure with either a positive or negative relationship. The downward and upwards arrows represent what is needed for that particular marketing requirement in terms of the engineering requirement. The targets for the enigneering requirements are also listed. The values describe how much is needed to meet the requirement. Such as meeting at least a seventy percent or greater statisfaction in the quality of the product. Having a response time of about 1 millisecond when the performer uses the microphone and the output is heard. The overall purpose of the functionality of the House of Quality is to provide a means of a planning process to meet the customer needs. The left side has the customer’s needs, the ceiling has the design features and tehnical requirements. The roof is the matrix that describes the relationship between the design features. Basically to show the interaction between the requirements. The foundation is the benchmark or the target values used to rank how things will be completed. The actions that will be taken to statisfy the customer.

2.4. Requirement Specifications

The design cost of the KPS is $300. To meet this amount extensive research will be done before ordering any components. If the correct or suitable components are not chosen correctly this could lead to fatal design flaws. Since there will be trail runs while designing the microphone more than one PCB board will be ordered to ensure good quality. The potentiometers, wireless device, LED lights and other expenses will also be taken into careful consideration for budgeting the KPS project.

Time will be leading factor in producing a good quality microphone. Milestone deadlines must be kept and checked through the course of the semester by each member of the group. There are 25 weeks to design the device, so to meet each goal all members of the group will work diligently and keep a timely schedule as planned. Table 1. below depict the specifications for the microphone to produce clear audio, the frequency range of the microphone, the LED display which will receive an input from the microphone or other source to determine what output will display according to the choice of the performer.

Table 1. Requirement Specifications

Weight	Less than 5 lbs. Led display may be separate from the microphone depending on the size.
Volume	The microphone will have range of volume from 1 dBA to 25 dBA
Frequency	The microphone will have a frequency range from 100 Hz to 10 kHz
Battery	(12 V 9800mAh lithium inside and easy to remove). As of now this battery could change depending on design constraints.
Connection	The KPS device will have the ability to receive data; (audio, voice) via wireless connection
LEDs	The LEDs will either be on the microphone or separate from the microphone that displays an array of colors the customer can adjust. This requirement is subject to change based on the constraints of the KPS design.
Components	The KPS will be using components which will generation low heat.
Output power	The output power for the KPS is about 25 Watts.
Setup and Installing	Easy to use and install
Supporting systems	Android and IOS
Music quality	Produce a clean sound up to at least 70% satisfaction for the customer
Dimensions	The dimension of the KPS will be about 5 x 5 x 12 in
Cost	Less or equal to 300 US dollars
Response time	Quick. The specific response time of the KPS is yet to be determined.
Volume amplifier	Adjusting the volume by potentiometer
Powerful echo reverb	Adjustable echo length

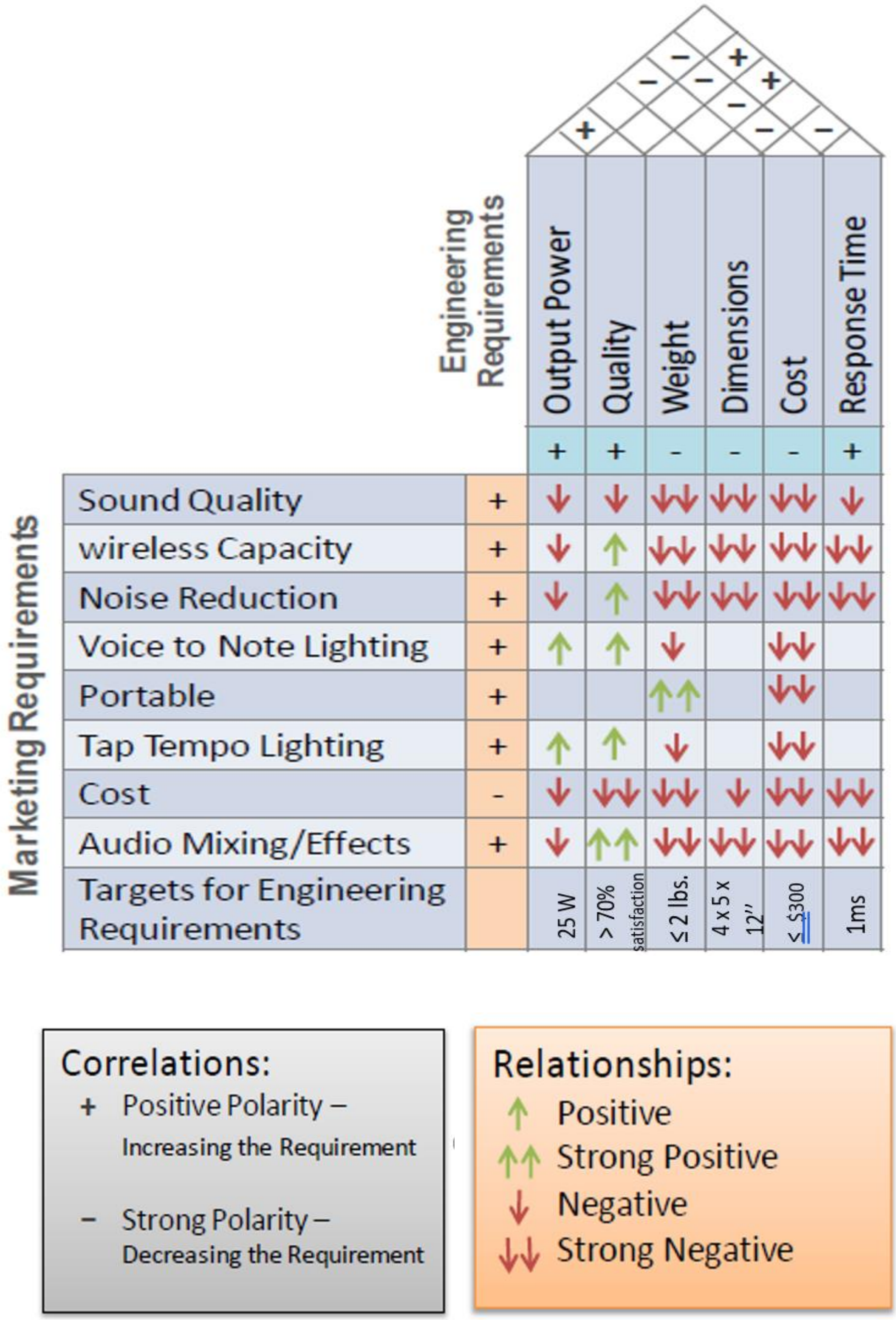


Figure 1. House of Quality

2.5. Project Hardware Block Diagram:

Figure 2 below depicts the process of the hardware design of the system. The division of labor between each group member is also indicated in the figure with a key to note what each member is working on. The project is modular and therefore the system is made up of two modules. One is the microphone module and the other is the LED system module that is an enhancement feature to microphone. The system is divided into modules to maintain a steady workflow among all team members. If a portion of the system where to fail or not work as intended the other parts that make the system will be separate and not hindered by the unfunctional portion of the system. For the LED system and microphone module there will be IC's that will be built and several components that will be purchased.

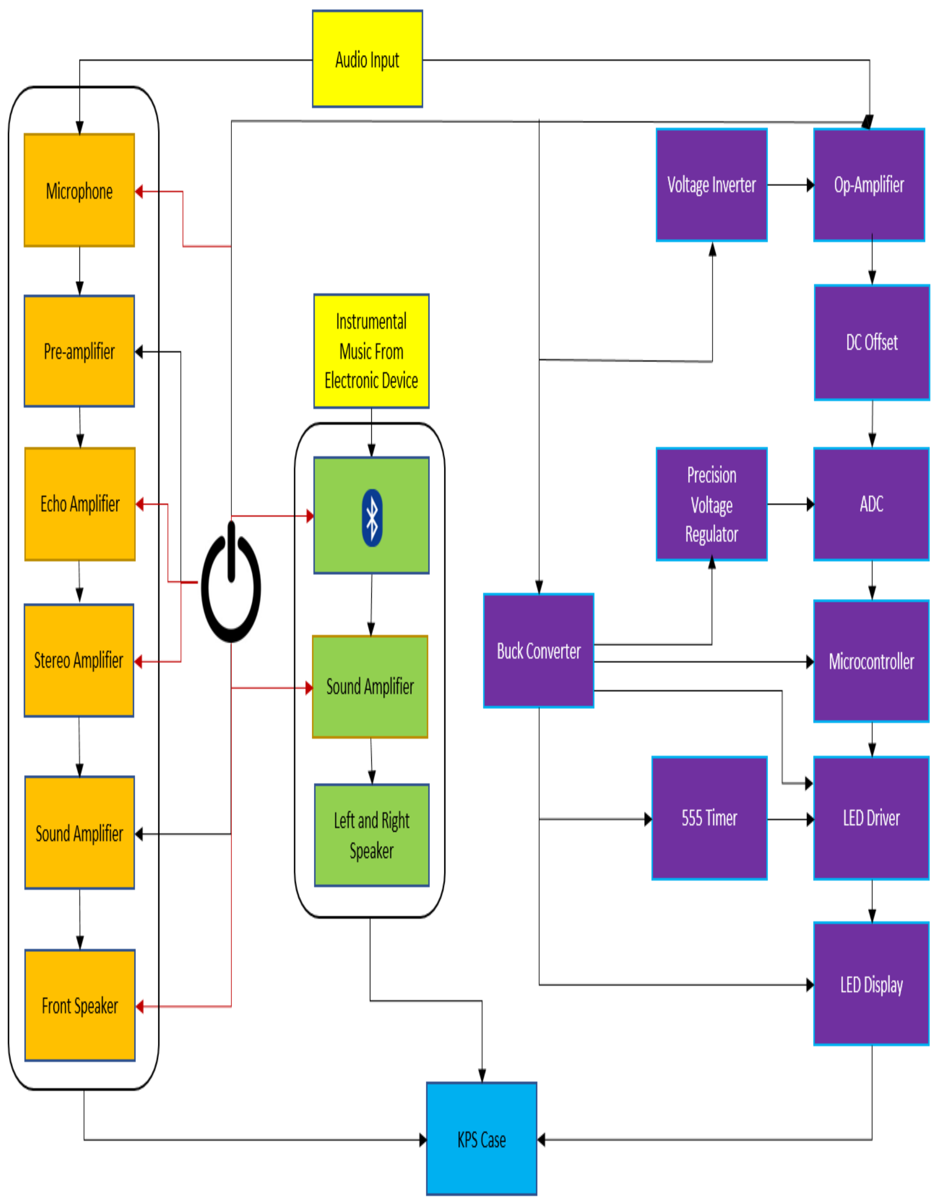
As shown in Figure 2 there will currently be five op amps in the microphone module that will be used for effects. These include bass, echo, treble, reverb and volume. Since the microphone is intended to be portable a wireless connection will be made to the microphone from a cellular device or personal computer. Hence a Bluetooth module will be purchased and is included in the microphone module. To take the digital wireless signal a DAC will be built to communicate across the microphone and device. The LED module runs parallel to the microphone. The LED system module will also have an op amp to amplify the voice signal. Since both modules are intended to run separate from each to cover all areas of error the op amp is crucial to the LED module. An ADC will be built to take the voice signal and convert it to digital to communicate with the microcontroller. LED drivers will also be built and configured to regulate power to the LEDs.

2.6. Project Software Block Diagram:

Figure 3 below depicts the current process of the software design of the system. The figure represents the division of labor of the software design. For the LED module there will be programming of the ADC with the microcontroller to ensure the analog voice signal is converted to a digital signal for mapping of the LEDs. The LED module is intended to have several features such as mapping a single specific note to a specific LED, an efficient program must be written to work with the frequency and map correctly.

The microphone module will have a wireless connection. A programmable code to ensure a wireless connection is made from the microphone to any device will be included. The code must run efficiently to ensure a good connection between devices. This will allow the user or performer to use any device such a cellular phone or computer to connect to the microphone.

Figure 3 depicts the overall software design and division of labor between each team member. To ensure the system runs well and functions as intended. The figure is divided by color for both the microphone module and LED system module of the KPS system.



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Note: Red lines indicate the voltage source has been through voltage regulator reducing to the limit voltage of each component.

Figure 2. Hardware Block Diagram

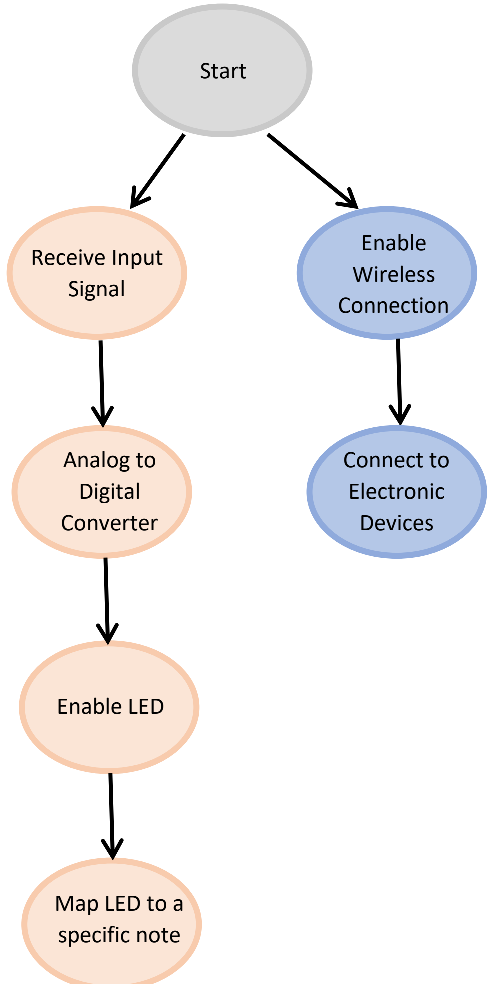


Figure 3. Software Block Diagram

Key:

Lam Hai Dinh

Tuan Dao

Jennifer Franco

Project Block Diagram Status:

- All the items from the block will be purchased
- Each block is currently being investigated
- All blocks are currently in design process
- None of the blocks are being prototyped

3. Project Research

In this section, we are in charge to research all our components and theories that we need to be ready to conduct our project. The research is including: components, battery, theories, the power delivery, and the connection between the components and power supply. Any problems may arise when we are researching.

3.1. Sound

Sound is a pressure wave which is created by a vibrating the objects. It is also the series of vibrations that caries energy and traveling through the air [1]. The vibrations set the medium particles in the surrounding air with vibrational motion, and the energy is transported through the medium of the air. Sound is created by anything that creates the vibrations or waves, is referred to as the sources. The sources can be a string, a bell, a voice, a speaker, a working machine, or anything that generates a vibration within our hearing range [1]. For example, when the stone is dropped into the water, there will be the vibrations in the water. That vibration is as a source that create the sound and waves. There are also a series of ripple waves in the water. The ripple waves are created by areas of molecules that are being pushed together. When these molecules expand by the pressure, they create flatter areas like harmonic waves. Sound travels just like this, by compression and rarefaction [1].

Compression is the area where molecules are pushed together in high pressure, and rarefaction is the area where molecules are moving away or pulling apart, or expanded, in the wave like harmonic with lower pressure than compression [1]. The way of the sound moving by compression and rarefaction is referred to a longitudinal wave. The longitudinal wave is that the mediums oscillates back and forth about their individual equilibrium position [2]. The oscillation in a spring is an example for longitudinal waves. When stretching out from one end of a spring, a wave can be created by stretching back and forth horizontally that same direction of spring. A wave will subsequently be seen traveling from one end of the spring to the other. As the wave moves along the spring, the energy is transported to the other end also.

3.1.1. Sound Implementation

In the music, the sound that humans can hear has to affect them. The definition of music is making a best sound that human like to hear with suitable frequency between 20Hz to 20Khz. The idea of music is change years by years, but the function of music is still the same. Humans want to hear music because it can change their state of mind to be positive or negative. The music affects human's emotions so deeply. Humans can listen to music in their language or other languages and their emotion still is affected. It means that the music is also a same language in tern of emotion. The human brain is an information processing system. An information processing system has four basic components: input, output, process, and storage. When applying these basic components for analyzing of music. The input is what the music that human can hear.

The music is storage in the human's brains, then it is calculated in the brain. The output is the human's state of mind. Years by years, human make music changes the quality of sound also. There are some characteristics of the sound wave, such as amplitude, frequency, phase, velocity, wavelength, harmonics, envelope, and noise [1]. Understanding these characteristics is essential to make the music better in the future. It also helps the human to create a better music speaker, headphone, or karaoke system.

3.1.2. The Characteristics of Sound Waves

In the speaker, the volume can be adjusted low or loud by changing the amplitude of the wave. The tone of the music can be adjusted by changing the range of the frequency. It is very important to understand these two characteristics "amplitude" and "frequency". They also help humans to know or decide or make the music even better and suitable the human's ears. The harmonic and envelope are also important to help humans identify which sound from and different to the others. The rest of characteristics including velocity, wavelength, and phase help human know how fast and how many cycle, and phase of the sound traveling through the air.

First, amplitude is the volume of the speaker or stereo, and in physics, it can be the height or magnitude of the wavelength [1]. In the wavelength, the volume can be adjusted by increasing or decreasing the height of the wavelength. By changing this, the pressure is changing too. High volume or loud sound has the high height, high pressure, and more area under the curve of the wavelength. Low volume or quiet sound has small height, low pressure, and less are under the curve of the wavelength.

The amplitude of the sound is determined by acoustic decibels [1]. A decibel dBA is a unit of measurement that indicates how loud of the sound is. Humans can hear the sounds between 0 and 160 decibels acoustic which between 20Hz to 20Khz [1]. 0 dBA does not mean that there is no sound, but there is a little bit humans can hear it. 0 dBA is also called the threshold for the human's ears. If humans can hear more than 160 dBA, it is too painful for human's ears and may damage the human's organs. The table blow shows how loud sound is [1]. In audio, the engineers is using decimal acoustic (dBA) to talk about the loudness of the sound or noise. When the engineer say to boost the dBA, that means the loudness is increasing (positive dBA). However, when an engineer says to cut the dBA, that means the loudness is decreasing (negative dBA). According to the author: "Timothy A. Dittmar" of the book title "Audio Engineering 101" that was printed by "Focal Press" in the Unites States. He gave some of the examples for the range of the acoustic decibels (dBA) that human can hear: [1]

Look at the table example for the range of dBA for human hearing, the humans use the amplitude to measures how forceful the wave is. It is measured in decibels or dBA of sound pressure. 0 dBA is the softest level that a person can hear. Normal speaking voices are around 60 to 75 dBA. Sounds that are 85 dBA or above can permanently damage your ears.

Table 2: Acoustic Decibels (dBA) for Human Hearing [1]

Acoustic Decibels(dBA)	**Human Hearing**
0 dBA	Almost quiet or near silence
40–50 dBA	Room Ambience
50–60 dBA	Whisper
60–75 dBA	Typical Conversation
80–85 dBA	A Blender, Optimum Level to Monitor Sound
90 dBA	Factory Noise, Regular Sound can cause hearing damage
100 dBA	Baby Crying
110 dBA	Leaf Blower, Car Horn
120 dBA	Threshold of Pain, can Cause Hearing Damage
140 dBA	Snare Drum Played Hard
150–160 dBA	Jet Engine

Second, the frequency of a sound wave refers to how often the particles of the medium vibrate when a wave passes through the medium [1]. The frequency of a sound wave is measured as the number of complete back-and-forth vibrations of a particle of the medium per unit of time or the number of cycle per second that is completed by the sound waves [1]. The humans can hear from 20Hz to 20kHZ, If there is higher or lower, humans cannot hear them. Any sound with a frequency below the audible range of hearing (less than 20 Hz) is known as an infrasound**,** and any sound with a frequency above the audible range of hearing (more than 20 kHz) is known as an ultrasound [3]. In sound waves, the frequency is determined by Hertz (Hz), named after the German physicist, Heinrich Hertz. In the United State, 1 kHz = 1000 Hz. Humans can change the frequency for making a best sound in music or identify where the source from. For example, in the guitar, there are six strings, and each string has its own frequency. The first string (note E) is about 329Hz, the second string (note B) is about 246 Hz, the third string (note G) is about 195 Hz, the four strings (note D) is about 146 Hz, the fifth string (note A) is about 110 Hz, and the last string (note E) is about 82Hz [4]. Also, in the karaoke speaker, humans can change the bass, volume, and treble through the amplitude and frequency.

In audio, the frequency can be divided into three range of the frequency including: low or bass frequency, mid or midrange frequency, and high or treble frequency [1]. In low or bass frequency, the range of frequency is between 20 Hz to 200 Hz.

These frequency is omnidirectional, make sound better, and provide energy [1]. In the mid or midrange frequency, the range of the frequency is between 200 Hz to 5 kHZ [1]. This frequency has more directional, so the sound can go to anywhere [1]. This range of frequency is the best that human wants to hear. In the midrange frequency, there can be divided into three additional area: the first one is low-mid, which the frequency is around 200 Hz to 700Hz, and make a sound more darker and hollower tones. The second one is mid-mid, which the frequency is around 700 Hz to 2 kHZ, and make more better like a live tone [1]. The last one is high-mid, which the range of the frequency is about 2 kHZ to 5 kHZ, and make the music more brighter and shiny. The last one of the range of the frequency is high or treble frequency [1]. In this frequency, the range of this frequency is from 5 kHZ to 20 kHz. and this frequency is extremely more directional. This range makes sounds airy, bright, shiny, or thinner. In music, the sensation of the frequency id called as pitch of a sound or a high frequency of sound wave [1]. The sound from two sources or different frequency from two sources that can combine by using superposition. Certainly, when two sound waves when played simultaneously produce a particularly pleasant sensation frequency when heard, are said to be consonant.

Third, Phase is a point in a sound wave's cycle, and is also related to frequency, it can be positive or negative phase depending on where the points on the wave’s cycles [1]. It is measured in degrees and is used to measure the time relationship between two or more sine waves. At a start point is zero-degree phase, and a complete cycle is defined as 360 degrees of phase. Leading phase refers to a wave that occurs ahead of another wave of the same frequency. Lagging phase refers to a wave that occurs behind another wave of the same frequency [5]. When adding two sound waves that have the same positive or negative phase, the magnitude of the superposition wave is increased. The phase can be 180 degrees out of phase. In this instance, the combination wave can be canceled out if they are 180 and -180 degrees. Phase is also very important to do when listening in multiple speaker or stereo. a sound may be completely out of phase and kind of cancel out, so the result would be unwanted sound.

Four, velocity of sound is the speed of the sound that travel through the particles, and depending on temperature [1]. In 20^{o}c, the speed of sound is 344 meters per second. At low temperature, the speed of sound is lower. At high temperature, the speed of sound is higher. The speed of sound is not only depending on temperature but also depend on material and altitude. For example, at sea level in a standard atmosphere, at a temperature of 15^{o}C, sound travels 761 miles per hour [6].

Fifth, the wavelength of the sound wave is the distance between two peaks next to each other or can be identified by the length of one cycle [1]. The wavelength can be calculated by taking speed of sound divide by the sound frequency. At the high frequency, the wavelength is much shorter. At the low frequency, the wavelength is longer.

Sixth, every music sound has harmonic. Harmonic series is the series of the frequency that can be multiplied by a positive integer with a first frequency [1]. The first frequency is called fundamental harmonic, and then the second harmonic is two times of the first harmonic. Additionally, harmonics are divided into evens and odds harmonic. Even harmonics are smoother and can make the listener feel comfortable and better. However, odd harmonics often make the listener feel bad [7]. In music, each instrument has their own musical makeup of a fundamental frequency plus additional own harmonics to that instrument. This is how humans can distinguish the music instruments.

Another characteristic is the envelope. The envelope is known as the varying level of a sound wave over time [1]. The envelope helps the listener know where the sound or voice from the other. The envelope includes four different characteristics: attack, decay, sustain, and release [1]. *A*ttack is the first point of a note or sounds envelope. It is known as the area that rises from silence node to its peak volume [1]. Decay is known as the next area of the envelope that goes from the peak to a medium level of decline [1]. *S*ustain is known as the portion of the envelope that is constant in the declining stage [1]. Release is known as the last point in the envelope where the sound returns to silence node [1].

The last characteristic is noise. Noise is any unwanted sound that is usually non-repeating, and humans do not want to hear [1]. Noise is bad sounds that accompany a sound wave when it is mixed or recorded. Noise comes from a variety of sources besides the instrument, such as an air conditioner, running machine. There is a way to compare the desired sound and noise is taking the ratio of desired sound and noise. The higher value of the ratio is the more quality of the sound. In recording, the noise often from the tape that recording the analog signal, and in digital signal, the noise can come from the low fidelity or the least range of bits number that use DAC [8].

3.2. Sound Effect

In this section, we will discuss the sound effect such as Bass, Treble, Echo and how to amplify those effect.

3.2.1. Bass

Bass describes the low-end of frequency response and ranges from 20 Hz to 250 HZ [1]. In the bass frequency, it is divided to areas. First is the sub bass that provides the first unused low frequencies on most recordings and ranges between 20HZ to 60HZ [1]. Many instruments struggle to enter this frequency range to the music instruments, but it is difficult to hear any sound at low volume level around the sub bass. The second is the real bass and range between 60 HZ to 250 HZ [1]. The frequencies around 250 Hz can add a feeling of warmth to the bass without loss of definition. In music, the bass gain is measured by decibels (dBA). Above zero dBA is boost the bass, and below zero dBA is cutting the bass. The base in the music can be using an analog signal or a digital signal to make a bass in the music.

3.2.1.1. Bass Amplifier

Bass refers to tones with lower frequency, pitch, and range [10]. Bass range from 20 to 250 Hz. They are the lowest part of the harmony in musical compositions. Bass voice refers to a type of classical singing voice which has the lowest range of voice types. In choral music, the bass sound is added by adult male singers. These are low sounds that when loud you can feel. The Bass control will cause the speaker to boost or cut these frequencies in what the humans are listening to. It's important to realize that a bass boost is not creating new sound, but instead it is boosting the music better. The amplifier for bass is fluctuation of frequency.

3.2.1.2. Analog Bass Amplifier

The signals or the human voices go through the air as analog signals to a low pass, this amplifier usually boost not a constant voltage or current but a fluctuating signal of some kind. By fluctuating, it means that it changes at a certain frequency (so many times per second, measured as so many hertz, Hz). Audio signals (ones we can hear), for example, the broad frequency range from about 20 Hz to 20,000 kHz that humans can hear. In the bass amplifier, the design is changing the low frequency between 20 to 250 Hz.

3.2.1.3. Digital Bass Amplifier

The voices or signals travel through the air go to the microphone. After this, the signals are gone to the low pass that is designed the frequency from 20 Hz to 250 Hz. After this, a device known as an analog-to-digital converter (ADC) receives the discrete voltages from the sample and ascribes a numerical value to each amplitude. This process of converting voltages to numbers is known as quantization. Those numbers are expressed in the chip as a string of binary digits (1 or 0). The resulting binary numbers are stored in memory. To play the sound back, we read the numbers from memory, and deliver those numbers to a digital-to-analog converter (DAC) at the same rate at which they were recorded. The DAC converts each number of binary to a voltage, and communicates those voltages to an amplifier to increase the amplitude of the voltage. After the signal goes to the DAC, there are some high frequency, so the system needs another low pass to get the desired signal or sound.

3.2.1.4. Types of Bass Amplifier

There are two types of bass amplifier including analog and digital bass amplifier. A bass signal is commonly recorded using an analog signal and digital amplifier emulation [10]. To control the sound, the analog signal is given the exact the signal when it goes through the low pass of the bass amplifier. The signal is kind of exact the same. However, in the analog amplifier, if the human wants to ignore some unwanted signal, the analog cannot do it because it is much easier to reach the exact sound that the human wants. To solve that, the digital can do it. Digital bass amplifier makes the signals to be processed so that the information that they contain can be displayed, analyzed, or converted to another type of signal that may be of use. In the real-world, analog products detect signals such as sound and manipulate them. All in all, the analog signal goes through a device as an Analog-to-Digital converter then take the analog signal and turn it into the digital format of

1's and 0's [10]. From here, the digital signal processing takes over by capturing the digitized information and processing it. By these processes, the bass digital amplifier helps the humans can select the signals or sound that they really want. Another thing is that there is the noise in these two bass amplifiers. The bass analog amplifier is reduction more noise than the bass digital amplifier.

3.2.1.5. Comparison and Conclusion

By comparing a digital and analog bass amplifier that want to have the exact same quality of sound and design, the human would prefer to digital bass amplifier since the digital emulation almost always matches up in terms of quality of sound. In other way, the digital bass amplifier can help the human to achieve all different styles of music. All in all, the bass digital emulation is the much more practical route when it comes to the bass.

3.2.2. Treble

Treble describes the high-range spectrum of frequency response and ranges from 2.5 kHz up to 20 kHZ [1]. The high tones are the sharper sounds you can hear in music. Boosting in this range makes sounds brighter, shiner, or thinner. In music, the treble gain is measured by decibels (dBA). Above zero dBA is boost the treble gain, and below zero dBA is cutting the treble gain. The most often that the treble frequency is used in music for the best sound is above 10 kHZ [1]. The treble music is using to control the tone of music. By combining the treble and the bass in the music, humans can control the tone and the quality of the music better.

3.2.2.1. Treble Amplifier

Treble is referred to the part of high frequency, and the range of that is from 2.5 to 20 KHz. In the amplifier, when turn up the amplifier, it typically adds more air to the sound. If the human turns up too high, the sounds can be painful for the human’s ears. High frequencies don’t travel as far as low frequencies [11]. This is why the humans mostly hear the bass from a faraway of the concert or from the party next door (bass can travel through walls). In the music, treble frequency is very important to make the sound a lot of better. The treble is the very high-pitched sound or tone and is the higher part in a recording. Usually, the treble amplifier is made from treble digital amplifier. When increasing the amount of gain above 0 dBA or at the high frequency, it will boost the treble. However, when decreasing the amount of gain below 0 dBA, it will cut the treble. Treble gain is usually applied to frequencies higher than 1000 Hz, with the most gain being applied to frequencies above 10000 Hz [11]. Also, in treble voice, a treble voice is a voice which takes the treble part of the highest-pitched part [11]. The term is most often used today within the context of choral music in reference to youthful singers.

3.2.2.2. Treble Analog Amplifier

The signals or the human voices will go through the air as analog signals to a high pass that can let all the high frequency go through, typically, the range is between 2.5 to 20 KHz in treble amplifier, this amplifier usually boost or cut not a constant voltage or current but a fluctuating signal frequency of some kind. By fluctuating, it means that it changes at a certain frequency (so many times per second, measured as so many hertz, Hz). Audio signals (ones we can hear), for example, the broad frequency range from about 20 Hz to 20,000 kHz that humans can hear. In the treble amplifier, the design is changing the high frequency between 2.5 to 20 KHz. A high-pass filter (HPF) attenuates content below a cutoff frequency, allowing higher frequencies to pass through the high pass filter. It is often used to clean up all low frequency signal and noise, remove unwanted sounds in audio signals, the allow the higher frequency signals to appropriate speakers in the sound systems. Human usually uses passive high pass filter for audio such registers, capacitors or inductors.

3.2.2.3. Treble Digital Amplifier

The voices or signals travel through the air as the sound go to the microphone. This is a analog signal like sine waveform. After this, the signals are gone to the high pass that is designed the frequency from 2.5 KHz to 20 KHz. Any signal with low frequency and noise is cut at the cut off frequency (frequency at -3dB). After this, a device known as an analog-to-digital converter (ADC) receives the discrete voltages from the sample and ascribes a numerical value to each amplitude. This process of converting voltages to numbers is known as quantization. Those numbers are expressed in the chip as a string of binary digits (1 or 0). The resulting binary numbers are stored in memory. To play the sound back, we read the numbers from memory, and deliver those numbers to a digital-to-analog converter (DAC) at the same rate at which they were recorded. The DAC converts each number of binary to a voltage, and communicates those voltages to an amplifier to increase the amplitude of the voltage. After the signal goes to the DAC, there are some low frequency and noise, so the system needs another high pass again to get the desired signal or sound. in digital audio, Equalization is the process of adjusting the balance between frequency components within electronic signals [12]. The most for using of the equalization is in sound recording and production sounds in the music. The circuit is used to achieve equalization is called an equalizer [12]. These circuits can boost (increasing) or cut (decreasing) the energy of frequency that carried when travelling in specific bandwidth. In sound recording and reproduction, equalization is the process that used to adjust the frequency response of an audio system using linear filters. Equalization uses the simple filters to make bass and treble adjustments.

3.2.2.4. Types of Treble Amplifier

There are also two types of treble amplifier including analog and digital treble amplifier. A treble signal (high pitch or high frequency) is commonly recorded using an analog signal and digital amplifier emulation. To control the sound, the analog signals are given the exact the signal (high frequency) when it goes through the high pass of the treble amplifier. The signal is kind of exact the same to the signals that pass the high pass filter with some noises. However, in the analog amplifier, if the human wants to ignore some unwanted signal, the analog cannot do it because it is much easier to reach the exact sound that the human wants because the signals are not analyzed or discrete as sample with the bits 1 or 0. To solve that, the treble digital amplifier can do it. Digital treble amplifier makes the signals to be processed so that the information that they contain can be displayed, analyzed, or converted to another type of signal that may be of use. In the real-world, analog products detect signals such as sound and manipulate them. All in all, the analog signal goes through a device as an Analog-to-Digital converter then take the analog signal and turn it into the digital format of 1's and 0's. At this time, with the binary numbers, the signals can be stored in the memory of the chip and will be reproduction after the requesting. From here, the digital signal processing takes over by capturing the digitized information and processing it. By these processes, the treble digital amplifier helps the humans can select the signals or sound that they really want. Another thing is that there is the noise in these two treble amplifiers. The treble analog amplifier is reduction more noise than the treble digital amplifier.

All in all, the main differences in performance between the treble analog and digital amplifier are the bandwidth of the frequency and the signal-to-noise ratio which the digital amplifier produces more noises than the analog amplifier does. In the music, human usually combine the bass and treble digital amplifier together to make the circuit smaller and more convenience. It is called digital tone control circuit. The tone control circuits are used to reduce the noise in the frequency when pass the filter circuit and to enhance (boost) the quality of signal or sound. The tone control circuit are usually used in music for bass and treble adjustment [13]. For example, when listening to the music, the listener can boost or cut the bass or treble frequencies. This process is called the tone control circuit. In modern technology as today, the tone control circuit is also called equalizer circuit, specially, graphic equalizer is a special of this process. The circuit is used to make a digital band pass filter of appropriate cut off frequency which will pass only the desired range of the frequency through the filters.

3.2.2.5. Comparison and Conclusion

By comparing the advantages of treble digital and treble analog amplifier and with the modern technology today, the treble digital amplifiers are more commonly used in today since they take more advantages than the treble analog amplifier. The treble digital amplifier is making the sound more enjoyable to the listeners since the listen can use digital amplifier to adjust the treble or bass frequency with the certain time. Also, the listeners can design what kind of music that they want to hear since they can easier change the treble and bass frequency. By the modern technology today, the chip is running faster, the analog digital converter and digital analog converter are developed very well which the humans can get more sampling when the process of quantization is improved, the listeners can listen to a better music over the using of analog amplifier for bass and treble.

3.2.3. Volume

The volume of music can be measured by acoustic decibels (dBA). There is +/- 30 dBA. When increasing the dBA, the volume is increased [1]. When decreasing the dBA, the volume goes down. In this case, when both increasing bass and treble frequency, and the overall volume is lower, the mid-range frequency is down. Similarly, when reducing both the bass and treble frequency, and increasing the volume has an overall effect of boosting the middle frequencies [1].

3.2.3.1. Volume Amplifier

Speaker are devices that translate signals from an electronic device, such as a receiver or CD player or sound waves. Sometimes, the listeners want to get a sound louder or smaller, then speakers may need a bit of a power boost to produce louder and cut the power to get sound lower. Running an amplifier between the audio device and the speakers will raise the wattage going into the speaker. This amplifier is called volume amplifier. The connection of the speaker and pre-amplifier is important, and without one of them, the listeners cannot hear any sound. the best combination of these relationship is to get a pre-amplifier that has twice wattage of the speaker [16]. For example, speaker can produce 50 watts, so the amplifier should be 100 watts.

3.2.3.2. Analog Volume Amplifier

A volume amplifier is an electronic circuit that boosts an electric current. For example, when recording the music through a microphone, the microphone will receive the sound in analog signals, then convert them into a fluctuating electric current (electric signal) that constantly changes in strength. A transistor-based amplifier takes the electric signal as an input and boosts it many times as a gain before going into a speaker [17]. As right now, the listeners can hear the sound as an output.

3.2.3.3. Digital Volume Amplifier

The digital volume amplifier can be called digital volume controller. When the sound or the analog signals go to the microphone. First, the signal should go to an ADC that convert to digital signal. Now, the signal will be sampled and analyzed into the binary number. The signal will go to the programable processor that reads if the number of binary is large then the sound is louder, but if the number of binary is smaller than the sound is smaller. After this, the digital signal will go to a DAC that convert to analog signal, and the listeners can hear through the speaker. For the perfect sound and desired loudness of the listeners, make sure the chip should carry at least 24 bits to cover enough the range of loudness or it will clip the signal [18].

3.2.3.4. Types of Volume Amplifier

The volume amplifier has two types of amplifiers including: analog and digital volume amplifier. In the analog amplifier, the electric signals (the electric current) is boosted to make sound louder. The gain is determined by the divider of output signal over input signal. As an analog volume amplifier, the noise when boost and cut the volume is less than the digital volume amplifier. Since the analog signal is fluctuating waves (like a sine wave), the noise can be adjusted when the volume turns down. However, in the digital volume amplifier, the noise is a fixed noise since the digital signals are equitization signals. A digital volume amplifier has its own problems. When each 6dB reduction in volume from the maximum setting then using 1 bit. For example, when reduction about 30 dB then there is 5 bits using. The lower the volume setting, the greater the loss in resolution [18].

3.2.3.5. Comparison and Conclusion

By comparing the advantages of volume digital and volume analog amplifier and with the modern technology today, the volume analog amplifiers are more commonly used in today since they take more advantages than the volume digital amplifier. The volume analog amplifier is making the sound more enjoyable to the listeners since when noise in analog signal is less than digital signal when turn down the music. Also, the listeners can design what kind of music that they want to hear since they can design the gain in analog signal. The digital volume amplifier is useful but there is a problem in noise and the range of bits using.

3.2.4. Echo or Delay

An echo is a time effect. An echo can take a direct signal and storage it first for a set of small delay time, and process it to be played back later [9]. A delay can repeat a signal once or multiple times. It can be used to separate a vocal from the rest of the mix or for a special effect. When applying a large amount of delay time, it can make a sound off. The delay or echo can be analog or digital delay. In analog delay, there are some solutions that use: tape delay, oil can delay, and bucket brigade delay [9]. In the most use for analog, humans use the bucket brigade delay because it creates the most precise output signal, but the problem is that there is a signal is passed form this stage to another stage. Each stage takes a certain time until It comes to last stage for the last delay.

The bucket brigade is usually take about 300 milliseconds for delaying [9]. However, for the longer delay time, human uses digital delay with simple support circuit with a timing chip. A digital delay is made by less filters and produces clearer sound. There is also a hybrid delay, the combine between analog and digital delay. In this delay, first the signal goes to analog delay first for certain delay time. Second, it goes to a digital delay. At right now, the output can be analog delay, digital delay, or between these two delays [9].

3.2.4.1. Echo or Delay Amplifier

Echo and delay are created by copying the original signals in some way including analog or digital filters, then replaying it a desired time later [9]. Early echo units were the using analog filters with more stages that were based on tape loops, and the most using for analog delay filter is bucket brigade delay circuit that is design with more stages to let the signals delay [9]. The analog delay filter is not more flexible than the digital delay filter since the delay time in analog is very short. Today, most delay or echo filters are using digital signal processing, but they often include the controllers to help them emulate the characteristics of the early tape units and bucket brigade delay circuits. While delay digital filters produces perfect echoes, an analog filter can be more enjoyable in music. In physic, an echo is kind of the rebounded of the sound. In other words, an echo is a sound that is repeated because the sound waves are reflected. A sound waves is reflected with the same direction and original sound. For example, a human can hear someone's voice when he or she is walking in the cave [14]. In music, echo is very important because sound voice from a singer that record from the microphone has to delay in very short time, typically, about 35 milliseconds [9]. It should happen in millisecond to mix with the music sound combination. For the echo with the time delay is larger, there is may be no sound or the sound kind of repeated in series. For example, when put the delay in certain time larger than millisecond, when say the words "Hotel", and it will sound like "Hotel otel tel el l".

3.2.4.2. Analog Echo Amplifier

In analog, there are two type of ways that using analog signal to record the sound and make it echo. The first is using the tape. Most tape machines have three heads that read and record to the tape. The delay time is the distance of the read and playback on the tape. The tape will use the three head to make it delay. The first head is the erase head that clear the tape before it is going to the record head. After that, the tape will record by using the record head. In this record head, the record head is not only record, and it can play when recording at the same time. After the record head, the tape will use the playback head. That means the tape is already recorded and is ready to play back. The speed of the tape will determine how the time for delaying. Moreover, another way to delay is using bucket brigade delay circuit. A Bucket Brigade Device (BBD) is an analog circuit contained within a small chip that delays an incoming audio signal [9]. BBD is made from a lot of circuits with more stages to make the signals delay. That is why the form of BBD is very large circuit. It is consisted of a series of capacitance sections C_0 to C_n. The analog signals are stored and moved along the line of these capacitors, one step at each clock cycle [9]. The time for the signal goes through all of these capacitors is the delay time, typically it is in milliseconds for delay time.

3.2.4.3. Digital Echo Amplifier

The term digital delay is often used to describe the modern sounding technology and accurate delay. However, digital delay processors are also capable of producing convincing analog and tape delay in the quality of sound and time delay. Many audio effects are based on mixing the original signal with delayed. When the microphone records the voices as analog signal. The signal then goes into the analog digital converter (ADC) to convert the signal to digital. The digital signal then is sampled then pass through a series of digital signal processors. This recording or delaying signal is saved in memory, then it is play back the stored audio signal on a certain time that the listener wants. After this, the signal will go to a digital to analog converter (DAC) to convert back to analog signal again. Digital delay line is one way to design the digital delay in the audio. The digital delay line is adjusted discretely by adding or removing gates from the delay line. A delay line has two units. The first unit is unit delay (Td), and the second unit is the number of stages (N) [15]. the more stages and the more unit delay are making the delay time more. With the human frequency voices between 20 Hz to 20 KHZ, for the digital delay line, at least 16 bits is used to sample the signal. The delay time is controlled by a programable processor that allow the users to change or modulation the time delay.

3.2.4.4. Types of Echo or Delay Amplifier

There are two types of echo or delay circuits that humans use in audio system. The first is analog echo or delay circuit. In analog signal, using tape and bucket brigade delay are two ways to record and delay the signals. The analog delay circuits are only making a certain time, and the users cannot change the time delay. The circuits in BBD is large with more stages to design, and the tape delay is not flexible to design in electronic circuit. The analog delay circuits give the listeners the sound more warner and natural because the signal is not sampled through the ADC. In the digital delay circuit, the digital delay line is one way that most people like to make since the system can use a series of programable processors to program a delay time. Digital delay circuit can cut through the mix much better than the analog delay has a natural and warm sound. Also, digital delays make the repeated sound more like the original signal with better ADC and DAC today. The analog signal gets more distorted and darker with each repetition. Digital delay circuit make the sound clearer in quality of sound.

3.2.4.5. Comparison and Conclusion

By comparing a digital and analog delay amplifier that want to have the exact same quality of sound and design, the human would prefer to digital delay circuits since the digital emulation almost always matches up in terms of quality of sound. In other way, the digital delay amplifier can help the human to achieve all different styles of music and make the music more affectable to human by the flexible of changing time delay. All in all, the digital delay circuit is the much more practical route when it comes to the echo or delay in audio system.

3.3. Echo Design

The echo digital amplifier is the delay or echo voice signal at certain time that we can design to adjust the delay time. In this digital echo amplifier, there will be an echo or delay adjustment and a volume amplifier. Also, inside the chips, it has the ADC and DAC, so the digital delay processor can take an analog signals (voice) as input, and the output is an analog signal. In this process, we can design a circuit to change the delay time and the volume of our voice. The good processor must meet our desire input digital voltage, and make low noise, and low power consumption.

3.3.1. PT2399 processor

The PT2399 is a single chip echo processor IC utilizing CMOS technology which accepts analog audio input signal, a high sample rate ADC transfer the analog signal into a bit stream then storage to internal 44Kbit RAM, after processing the bit stream will de-modulate by DAC and lowpass filter. Overall delay time is determined by internal VCO clock frequency, and user can easy to change the VCO frequency by changing the external resistance [55].

3.3.2. ETK3699 Processor

ETK3699 is an echo audio processor IC utilizing CMOS Technology which is equipped with ADC and DAC, high sampling frequency and an internal memory of 44Kbit digital processing is used to generate the delay time, it also features an internal VCO circuit in the system clock, thereby, making the frequency easily adjustable [56].

3.3.3. M65850P/FP Processor

The M65850P/FP is a CMOS IC for generating echo to be added to the voice through a karaoke microphone. It is optimal to provide the echo effect function for karaoke player, such as radio cassette recorders, mini audio components and television sets. Increased master clock frequency assures high-performance short delay. The IC has the largest memory among the digital delay series [57].

3.3.4. BU9253 Processor

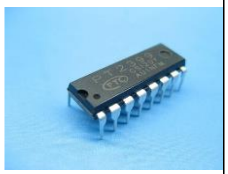
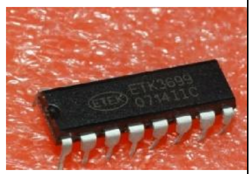
The BU9253AS, BU9253FS and BU9255FS are single-chip ICs that contain all the components needed to configure a KARAOKE echo system: an ADC and DAC converter, SRAM, LPF, and mixer for mixing source signals. With these ICs, an echo function can be configured easily and with minimum external components. Echo mixing ratio is adjustable with a DC voltage [58]. This chip supports Karaoke echo and surrounding system, all in a single chip.

3.3.5. Comparing These Processors

PT2399, ETK3699, M65850, and BU9253 are the chip processors that we will choose one of them to design for our project. It is good all the chips can help us to design an echo or delay circuit, and they all have the ADC and DAC inside the chip. Also, there is the memory inside these chip that can help us store signal in certain before playing back. As we can loop up some of their features and applications.

The table below is showing some very important limit values and characteristics that we need to do our project. By comparing these processor, the processor PT2399 has more advantages to use for our design than these others. The first reason is that the voltage supply and the bits range are met our requirement, also the output noise voltage is low. The good reason is that we can the overall delay time by internal VCO clock frequency because we can change the VCO frequency by changing the external resistance.

Table 3. Echo Processor Comparing

Processor Name	PT2399 	ETK3699 	M65850P/FP 	BU9253 
Application	Karaoke mixer and music instrument effect(echo/delay)	Karaoke mixer and electronic musical instrument and echo processor	Karaoke, echo for voice for karaoke microphone	Karaoke functions for portable stereo sets, mini components
Voltage Supply (V)	Min: 4.5V Max: 5.5V	Min: 4.5V Max: 5.5V	Min: 3.5V Mac: 5.5V	Min: 4 V Max: 5.5 V
Bits Range	16 bits	16 bits		
Output Noise Voltage	Min: -95 dBV Max: -80 dBV	-90 dBV	-85 dBV	-70 dBV
Echo Adjustment	VCO clock frequency	VCO clock frequency	Master clock frequency	External Adjustable

3.4. Tone Control Amplifier Design

The music speaker amplifier is designed to take analog input signal (music from DAC that is conned to Bluetooth), and the input signal can adjust by changing its magnitude (volume) and frequency (bass and treble) to make the best music into two speakers in the left and the right. In the process of this amplifier, the music is taken from any smart devices such as ipad, table, phone through Bluetooth. The signal output from Bluetooth is digital signal, so an DAC will convert digital to analog signal. This signal can be adjusted with frequency and magnitude to make music by changing external components such as capacitor and potentiometer. An analog chip is chosen to meet our requirement in voltage supply and output voltage with low noise, and low power consumption.

3.4.1. TDA1524A IC

TDA1524A is designed as an active stereo-tone and volume control, especially for car radios, TV receivers and mains-fed equipment. It includes functions for bass and treble control, volume control with built-in contour (can be switched off) and balance. All these functions can be controlled by DC voltages or by single linear potentiometers [60]. This preamplifier has the volume control (with integral power switch), Balance, Bass and Treble tone controls. It provides very pleasing performance and makes a useful addition to any of our audio power amplifier kits. RCA jacks for audio inputs [6].

3.4.2. LM1036 IC

LM1036 is a DC controlled tone (bass/treble), volume and balance circuit, especially for stereo applications in car radio, TV and audio systems. Four control inputs provide control of the bass, treble, balance and volume functions through application of DC voltages from a remote-control system or, alternatively, from four potentiometers which may be biased from a Zener regulated supply provided on the circuit. Each tone response is defined by a single capacitor chosen to give the desired characteristic [61].

3.4.3. LA4440 IC

LA4440 is a two channels power stereo amplifier that can deliver up to 6W output per channel of 19W in bridge connection. It requires minimum number of external parts, it has small pop noise at the time of power supply ON/OFF. The LA4440 audio amplifier circuit has good ripple rejection of 46dB, small residual noise, built in over voltage and surge voltage protection. ideal feature of the IC is its pin-to-pin protection. Also, it is low distortion over a wide range from low frequencies to high frequencies, built-in audio muting function, built-in protectors [62].

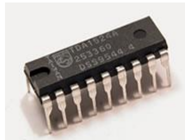
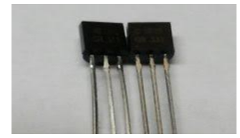
3.4.4. Audio Transistors (2SA1015 and 2SC1815)

2SA1015 is a Silicon PNP Epitaxial Type that is made from Toshiba transistors company. This is an audio frequency general purpose amplifier applications driver stage amplifier application. This audio transistor creates very low noise (1dB typically). 2SC1815 is a Silicon NPN Epitaxial Type that is also made from Toshiba. This is an audio frequency general purpose amplifier applications driver stage amplifier application. This transistor also creates very low noise (1dB typically). These audio transistors are working on very low power supply (typically 3.3V). Together, we can make a low voltage preamplifier circuit with tone control by using them.

3.4.5. Comparing these ICs

TDA1524A, LM1036, and LA4449 are stereo amplifiers that we will choose one of them to design for our project. all of these amplifiers are that can help us to design a tone control (bass, treble, balance) and volume amplifiers. They meet our requirements to design our tone control and volume. Also, they all produce low residual noise and the pop noise. As we can loop up some of their features and applications. The table below is showing some very important limit values and characteristics that we need to do our project. By comparing these preamplifiers, TDA1524A has more advantages to use for our design than these others.

Table 4: IC Comparing

IC Name	TDA1524A 	LM1036 	LA4440 	2SA1015 2SC1815 
Application	Stereo-tone/volume control for car radios, TV receivers and mains-fed equipment	Stereo applications in car radio, TV and audio systems	Stereo and bridge amplifier applications	Using in audio for headphone and portable electronic devices for low power supply
Supply Voltage (V)	12V	Min : 9V Max: 16V	13.2V	1.5 – 3 V
Volume Control Range (dB)	−80 dB to + 21,5 dB	70 dB to 80 dB	External component design	External resisters
Bass Range (dB)	Range at 40 Hz (−19 dB to +17 dB)	Range at 40 Hz (+/-15dB)	External component design	External resisters
Treble Range (dB)	Range at 16Khz (+/- 15 dB)	Range at 16 KHZ (+/-15 dB)	External component design	External resisters
Operating Temperature	-30^{o} C to 80^{o} C	0˚C to +70˚C	–20^{o} C to +75^{o} C	-55 to 125^{o} C
Output noise voltage	100 uV	16 uV	0.6 mV	10 uV

The first reason is that the voltage supply is met our requirement with working in small signal, also the output noise voltage at the minimum gain is low. Also, the range of the volume, bass, and treble are met our requirements by adjusting the potentiometers. The operating temperature Range of this chip is small (-30 to 80^{o}

C). This is good since when running the circuit, it will get hotter, so we need a large range of operating temperature

3.5. Voice to Visual Display (VTVD)

The voice to visual display will function as an individual unit to the system. The display is intended to be an enhanced feature to the Karaoke Portable System. The LED display is a collection of individual LED lights in which an LED driver can be accessed to toggle the individual addressable LED lights on and off to an array of colors. It will add an esthetically pleasing overall look to the KPS. Not only will the portable microphone allow the singer to have the ability to hear their favorite songs but perform to their favorite songs while an arrangement of lights display during the performance for an exciting and interesting show for all the enticed viewers watching.

The design workflow of the Voice to Visual display will be referenced in detail in the following pages. Below in Figure A1 is the connection diagram for the voice to visual display. Overall the display will take an input signal which will be used as a reference to determine the LED settings to be displayed. The display will allow the end-user to choose from a few options of how the LEDS will display. That is the ability to have a visual feedback of the specific notes that he or she is singing, set the brightness level, choose from a set LED flashing patterns or possibly display LEDs based on the tempo at will. These effects are being implemented to add a new dimension to Karaoke performances.

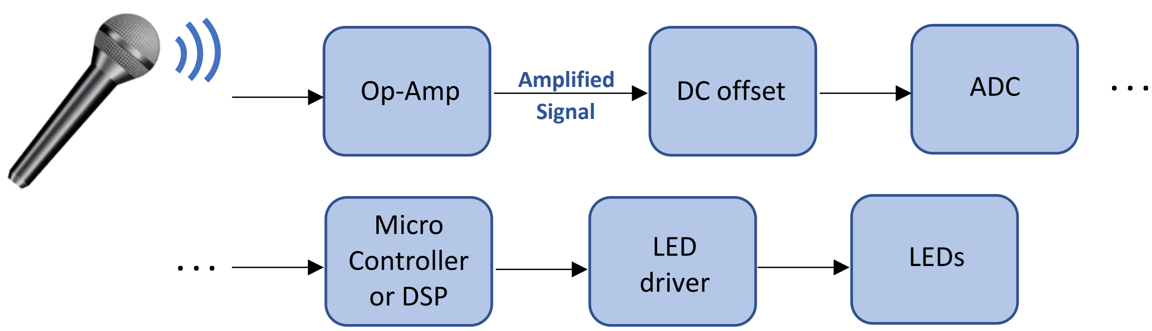


Figure 4. Voice to Visual Display Connection Diagram

The goal is to display the LEDs while the performer is singing at will. To accomplish this, an audio input signal from the microphone will be taken. An operational amplifier will then amplify the signal to be used with an open-source electronics platform. These signals need to be amplified to get them to the amplitude we want or need. Once the microphone signal has been amplified the signal will then need to be DC offset. The DC offsets the output of the amplifier and will cause the audio signal to stay within the acceptable range of the microcontroller that will be used. An analog to digital converter will convert the audio signal for use in the microcontroller to program the board and use the led driver to drive the LEDs and display the plethora of lights.

Some of the features for the visual LED display to be implemented in this prototype or most likely in future prototypes will be; LED intensity mapped to amplitude, potentiometer adjusting gain, adjustable frequency range, programmable output display with optional input tempo. The LED intensity mapped to amplitude will simply adjust the brightness of a LED configuration based on the intensity of the input, in this case being the vocalist performing a specific song. The louder the vocalist performs, the brighter the LED will display the given color corresponding to its frequency. A potentiometer to adjust gain serves a purpose to allow the output of the operational amplifier to be adjusted so that no clipping will occur. An optional LED can be attached with the potentiometer to give the performer an indication of clipping and be able to adjust the output accordingly. The adjustable frequency range would give the user an option to set the tolerance of desired frequency to correspond to an LED. This added feature would allow for a more dynamic high and low threshold for the incoming input to correspond to its color and specific LED. A future implementation would be to also have several configurations of lights displaying in different modes for enhanced performance and visual stimulation. With different modes, it could mimic a light show performance seen by typical shows nowadays. A tap tempo button could also be added to the different modes allowing a performer to tap the tempo of the song being sung, and have the LED patterns light up in sync with the tempo the of the song, giving a true feel of a dynamic light show in a concert hall.

3.5.1. Operational Amplifier

To select the appropriate op-amp for the Voice to Visual display the microphone level is to be considered. The microphone-level signal is the voltage level that comes out of a microphone when someone speaks into it. This will be the input signal that will be taken and modified to produce the output we need. The mic level is in the region of -60 dBV (0.001 volt) to -40 dBV (0.010 volt) [19]. The voltage will also vary in response to changes in the voice level and the distance from the singer-to-mic. But overall the signal is going to be quite small. [19] Therefore, we will need to prepare the signal for use in the microprocessor or for other functionalities. What the amplifier will do, is increase the amplitude of the signal as well as protect the audio source and produce a signal large enough to use in most processors. The outgoing amplified signal is going to source all its current from the amplifier, so any load that will be added later unto the circuit will not be felt by the microphone.

3.5.2. Traditional Split-supply Op Amps

The operational amplifier is a high-gain dc-coupled differential amplifier with a single-ended output. Operational amplifiers are available in thousands of types, offering various performance tradeoffs. One of the requirements is the power supplying the op-amp. A split-supply op amp has two supply rails with reference to ground to an op amp (i.e. +VCC and -VCC) rails. The applied voltage can swing between these two voltage levels. In an audio signal amplification, the input audio signal (voice) can swing between some (+Ve and -Ve) voltage levels (usually in mV) as stated above and cannot exceed these levels. Most traditional split-supply op-amps should be powered with +15 and -15V, but since the signal will never be amplified above + or – 2.5V it will be fine to run the op amp with something lower such as + or -9V power supply.

Taking into consideration this requirement for the split-supply, a few split-supply op amps were selected. Table 6 depicts various useful op-amps selected and their performance tradeoffs and price. For the selection of the op amp we need to consider which op amp has the highest input impedance and lowest output impedance. Table 5 shows a summary of what we are looking for in the selection of the dual power supply op-amp.

Table 5. Operational Amplifier Requirements

Device	Input Impedance	Output Impedance	Gain	Bandwidth
Op amp	High Output	Low output	High	Limited

Table 6 below is a summary of the op amps considered for the LED display. We want to achieve characteristics of an ideal op amp, so considering the requirements listed in Table 5 we chose a few op amps that could possibly be used.

Table 6. Operational Amplifier Comparison Table

Op Amp	TYPE	V_S(min)	V_S(max)	I_B Bias current	Bandwidth	Slew rate	Price
LF411	JFET – Input	-7V	36V	50pA	3MHz	13V/µs	$1.43
UA741C	CMOS	-18V	18V	80nA	1MHz	0.5V/µs	$0.45
LM 741A	Bipolar (BJT)	-22V	22V	80nA	1.5MHz	0.5V/µs	$0.73
LMC6482A	CMOS	- 3V	16V	20fA	1.5MHz	1.3V/µs	$1.92
TL081CN	JFET – Input	-18V	18V	30pA	3MHz	16V/µs	$0.52

3.5.3. Detailed look at Split-Supply Op Amp Table

One of the op-amps listed in Table B2 is the LF411. An inexpensive and easy to use operational amplifier. It serves as a good starting point for circuit designs. Most of the op amps listed above are very commonly used. Table B2 lists values for the best op amp you can get for each individual parameter. Let us note the input bias current I_B, which is half the sum of the input currents with the inputs tied together (the two inputs are approximately equal and are the base or currents of the input transistors) [20]

For the JFET-Input LF411 the bias current is 50pA, while a typical BJT-input like the LM741 has a bias current of 80nA. So roughly we can say that BJT-inputs have bias currents in the tens of nanoamps and JFET-Input op amps have currents in the tens of picoamps which is 1000 times lower. The significance of the bias current is, that it causes voltage drop across resistors in the feedback network. [20] Means can tolerate resistances up to some value before you must worry about it reaching the offset voltage max. From the table above, bipolar op amps have good wide supply voltage range, speed, and noise at the expense of the bias current; JEFT-input op amps like the LF411 and TL081CN are intermediate and CMOS op-amps like the LMC6482 displaying the lowest bias current.

3.5.4. Single power supply Op-amp

Single supply Op-amps have only one supply rail (+VCC) for which the applied signal will be amplified or swings only in between the +VCC and GND. Therefore, the output voltage has a swing in between +VCC and GND rails. Single-supply operation is commonly synonymous with low-voltage operation [21] One of the features we must think about is the fact that we cannot generate negative outputs. However, certain applications using high voltage and high current op amps can benefit from single supply operation.

3.5.5. Advantages & Disadvantages of single power supply Op-amp

Consider the basic op amp connection in Figure 5a. The circuit is connected as a voltage follower, which means it’s output voltage is equal to the input voltage. It is powered from a dual supply. There are limitations to the voltage swings, so as the input voltage swings positive the output at some point near the positive power supply will be unable to follow the input because of the limitation. A typical op amp has roughly a 2V difference from the actual max voltage supply. In this case it will be from -13V to +13V.

Figure 5b shows the same unity-gain follower operated from a single 30V power supply. The output of this op amp can follow the input if it doesn’t come no closer than 2V from either supply terminal of the op amp, its range is from +2V to +28V. Ideally any op amp is capable of this type of single-supply if the rails of the dual supply have different limits. The advantage of using a single supply op amp is biased toward the type of application that is targeted for single supply. For example, the limit on the “negative” power supply to the op amp is a significant limitation. If an application has the input signal referenced to ground, the input signals of less than 2V would not be handled accurately by the op amp. A single supply op amp would handle this application more effectively. The disadvantages of single-supply operation are that it aggravates the problems of noise, biasing, and distortion. Use of single supply op amps also consequently intensifies the common-mode input range, output voltage swing and CMRR. There are possible parameters that also need to be considered when operating a single-supply op amp.

High performance, single supply op amps are becoming more common, but to maximize performance sometimes a dual-supply amplifier is an ideal choice. The selection of dual-supply op amps is greater because dual-supply systems have been available longer and dual-supply op amps are not designed with the same restrictions as the single-supply. From the points listed in the disadvantages of single supply op amp, it was decided that ideally a split supply op amp is the best choice.

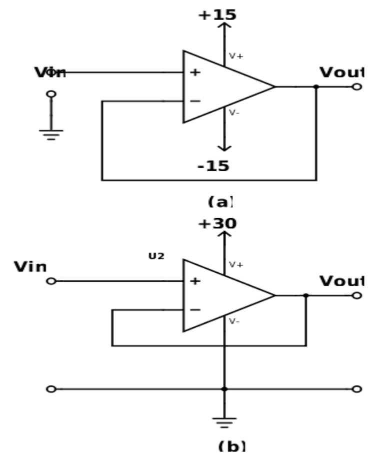


Figure 5. Dual vs Single Supply

3.5.6. Operational Amplifier Design (Non-Inverting Amplifier)

From the research for the op amps we chose to go with a split-supply op amp. The goal is to increase the amplitude of incoming signal from around + or – 200Mv to about + or – 2.5. The datasheet of the UA741 says that the op amp should be powered with nominal voltage of +15 and -15V, however since the signal won't go above the 2.5V we could run the op amp at a lower voltage value. The following equation describes the properties of the relative amplitudes of the signal after and before using the non-inverting amplifier.

$$Vout = Vin * \ 1 + {}^{R3}/_{R2} \tag{1}$$

$$or \qquad {}^{Vout}/_{Vin} = 1 + {}^{R3}/_{R2} \tag{2}$$

R3 is the feedback resistor and then we have the resistor to ground, R1. The output voltage (Vout) is the amplitude of the outgoing signal from the amplifier and Vin is the amplitude of the incoming signal, the input to the op amp. In the design of the amplifier there is also a capacitor-resistor combination not usually found in a typical non-inverting amplifier. The reason the resistor is inserted is to prevent the output of the op amp from driving into one of the voltage rails. Typically, the value for R1 may be 100 k ohms or more.

In the circuit R2 is a 100kΩ resistor and R2 is a 10kΩ variable resistor (potentiometer). By using this pot, we can change the resistance of R2 from 0 to 10kOhms. Using equations (1) and (2) we can find the ratio of Vout to Vin. If the pot is turned all the way to the left for a resistance value of 10kOhms the ratio will be 11, if the pot is turned the opposite way the value will be 2.2V, which is in the range that we are looking for, close to 2.5 V. Overall turning the potentiometer to the right decreases the resistance of variable resistor and increases the gain or the amplification of the signal to infinity (ideally). The use of having this design is to allow adjusting the variable resistor or the potentiometer to adjust the gain of the op amp and that way we can tune the microphone sensitivity and keep the ranges we want.

Figure 6 shows the non-inverting amplifier design with a capacitor-resistor combination.

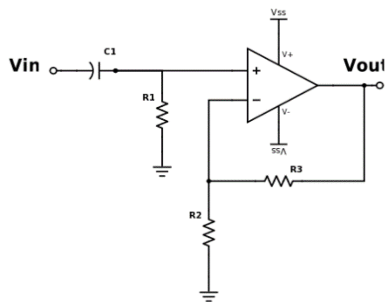


Figure 6. Non-Inverting Amplifier Design

3.6. ADC (Analog to Digital Converter)

An Analog to Digital Converter does just as it states. It takes an analog input (the VTVD), and converts it to a digital value in which the microprocessor can interpret. However, choosing an ADC isn't as straight forward as its functionality. Numerous different specifications are used to describe them. Namely Resolution, Accuracy, Sampling Speed, and Quantizing noise. Resolution is simply the number of output bits per conversion. Accuracy is how close the output is in representing the maximum resolution given. This is usually dictated by noise, and nonlinearities defined by the ADC itself. Sampling speed is the most conversions that an ADC can be made per second. And finally Quantizing noise is a specific type of noise (unwanted voltage) that is added to the input [22]. When researching different ADC's to use for the VTVD these parameter specifications will be taken into consideration for part selection and ADC type.

Along with the numerous specifications, also comes the various types of ADCs. The three most common ADCs are flash, successive approximation (SAR), and Sigma-Delta (ΣΔ). Flash ADCs are the fastest type of converter out on the market today. However, with speed comes drawbacks. The Flash ADCs are built with comparators. The number of comparators is based on the number of bit resolution. If it is an 8-bit ADC, the converter will have 256 comparators. This means that the size of these converters can be quite large, and because of the sheer number of comparators needed, it can consume a significant amount of power. Since size will ultimately be contributing factor to future prototyping of the V [23] TVD, the Flash ADC will not be considered for design of the system, leaving the two most popular types of converters left to choose from; the SAR or Sigma-Delta.

To calculate a conversion, the SAR ADC uses a comparator and counting logic. Interestingly, SAR takes the input voltage and compares it to half of the reference voltage given (in the VTVD case; reference voltage given by either the power supply, microchip, or DSP chip). If this is true, the most significant bit is set. The calculated value is then subtracted from the input once again and checked for one quarter the reference voltage.

This process is continually done until all bits are set. The huge advantage of SAR is that it is fast in computation and widely known as an industry standard. Sigma-Delta uses oversampling to achieve conversions, and is therefore very high in resolution. However, this is offset by lack of speed and the complexity of having a digital filter [23].

Table 7. ADC Comparison Table

Manufacturer	**Series**	**Architecture**	**Resolution (bits)**	**Sampling Rate(Hz)**	**Cost**
Maximum Integrated	MAX1243	SAR	10	73K	$5.51
Texas Instruments	TLV2541	SAR	12	200K	$4.55
	ADC121S101	SAR	12	1M	$3.93
Analog Devices	AD7478	SAR	8	1M	$3.90

Table 7 shows some of the ADCs for possible purchase. It is interesting to note that all of the architecture choices are SAR. The reason for this is chip selection and availability more than any other deciding factor. When filtering on any distributor with the following parameters; Analog Voltage ~2.5 – 5.3, number of bits (12-16), frequency ($\geq 42\ kHz$), it seemed that the only available option for single channel ADCs are SAR. No Sigma Delta chips had the frequency range with all the other options needed.

Once noting architecture, it became clear from the comparison of the chips, that the most reasonable option to go with is Texas Instruments' ADC121S101. In comparison, it has one of the highest resolutions, highest speeds, and is also close to the cheapest, only falling second by three cents to Analog Devices' AD7478.

Figure 7 demonstrates an ADC waveform. An analog signal and it the corresponding quantized signal. A sampled signal (right), the sample and hold function of the ADC, and the original analog signal. This model has added slight noise to create some real effects of what could occur while using a ADC chip. One issue that will be of great concern when designing and implementing the ADC chip is the reference Voltage, V_{ref}.

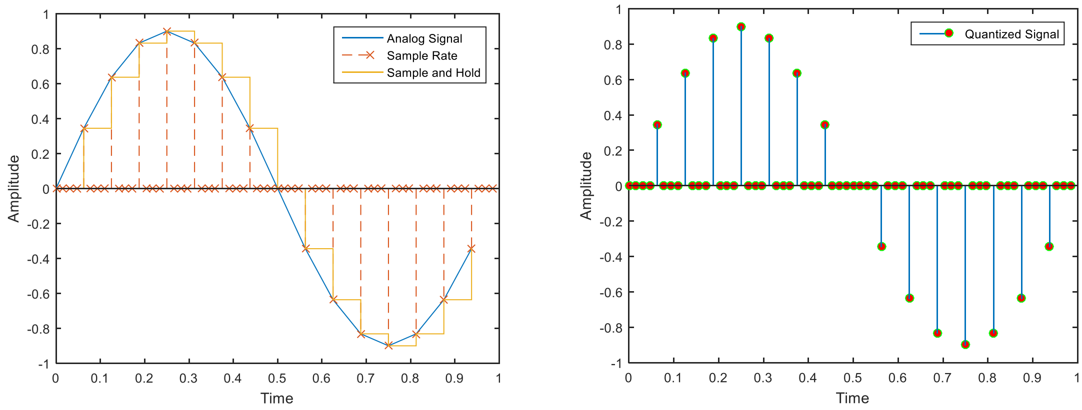


Figure 7. ADC waveform

Most microcontrollers allow for use of such an input pin. For the possible use of an Arduino, a 5-volt reference could be used. With the 12-bit resolution decided on the ADC, a value of $\frac{5}{2^{12}} = 1.22\ mV$ is the subdivision used for each precision value. However, this result is completely dependent on that 5V reference. Typically supplies can vary two to three percent, which can significantly affect the output of the ADC. If the power was off by the extremes $\pm 3\%$, it would be seen that the corresponding low value error would be 1.18 mV, and its corresponding high value error would be 1.25 mV. This may not seem to be a significant change from the ideal 1.22 mV. However, if there are 4,096 corresponding values accounted for in between five volts, then these values will become significant, and cause error in mistaken one voltage range for another.

3.7. Processor Adaptation: Calculating Audio Signals

One of the most important decisions for the VTVD is deciding how to calculate the input signal to get a corresponding frequency. That frequency will then be used to determine which color to display. The choices for processing a signal come down to two distinct options; Analog or Digital. This may seem a straightforward choice, being digital has dominated the audio processing world as of late. However, as will be outlined in the following sections, each choice brings advantages and drawbacks.

3.7.1. Digital Signal Processing (DSP) Chip

Digital Signal Processors have many advantages to use to process the input audio signal. The chips architecture and instruction set are built for real time processing using mathematical functions and algorithms. Along with its powerful ability comes multiple options when considering the correct chip for the given design for processing. When choosing a processor, peripherals (Serial ports, Connectivity, Sample rates), Memory (Flash), Size, Power consumption, ease of development and Price, play important roles. However, before considering these constraints, formatting and calculating data must be studied. The two categories for data manipulation are Fixed Point and Floating Point.

Fixed-point processors operate on integer values, with a minimum of 16 bits. This gives up to 65,636 possible bit patterns. In contrast, Floating-point processors operate on rational numbers with a minimum bit range of 32. This allows for significantly larger combinations of patterns; approximately four billion (2^{32}). Note that the gap for fixed-point processors always remains at a value of one. This allows for uniformity, yet does not allow for such a large dynamic range that floating point gives. When processing a large amount of data, floating point processors are an absolute necessity.

Precision must also be evaluated when distinguishing between the two types of processors. Rounding will occur no matter which type of processor will be used. This type of Rounding occurs due to memory limitations when storing data and from mathematical operations. The interval between integers can be much larger than floating point values for processing. Leading to a larger quantization error when using fixed-point processing. Floating-point processing does have several drawbacks. One which is of great consideration is the complexity in developing and implementing effective algorithms [24].

3.7.2. Digital Signal Processor Comparison and Benchmarks

When choosing a DSP, it is often difficult to understand what one may need given the typical data of clock speed and memory. Adding to the confusion further, Benchmarks given by manufacturers often highlight strengths of a chip while being elusive to any shortcomings it may have. The key to successful selection is to plan for what algorithm will be implemented, and how many clock cycles are required for the performance required by the system [25].

When it comes to benchmarks and execution time, the typical benchmarks used are FIR filters, Biquad Filters (second order IIR filter), and FFT computations. This is because FIR filters can be memory intensive. For an N-point FIR filter, $2N + 3$ accesses to memory occur, as well as N number of MACs (Multiply Accumulate Operations). Biquad Filters are also MAC intensive, having two memory accesses as well as five MACs. The FFT benchmark allows for a more complete test, having both memory accesses and mathematical operations. [26]. With looking at these benchmarks, the most fundamental question that needs to be answered is; how fast does the VTVD need to be? At the highest level of fidelity, voice or music in general, will contain the full spectrum of hearing; 20 Hz to 20 kHz. A popular sampling rate used is 44.1 kHz [25].

Most processors will be able to handle this requirement. Yet it will still be an important standard and therefore included in the comparison table. With the ever-increasing popularity and development of DSP chips, narrowing down a chip choice was a tedious task. Table 8 shows two main manufacturers under consideration; Texas Instruments and Analog Devices. All but one DSP had floating point arithmetic capability. The Clock speed all well exceed the VTVD needs and memory is sufficient for all anticipated mathematical computations. The I/O Voltages and operating voltages are all similar in design.

Table 8. DSP Comparison

Manufacturer	Series	Arithmetic	Clock Rate (MHz)	Memory (kB)	Voltage I/O (V)	Voltage Core (V)	Cost
Texas Instrument	C6720	Floating-Point	200	384	3.3	1.2	$13.10
	C553x	Fixed-Point	100	128	1.8, 2.5, 2.75, 3.3	1.3	$8.03
	C6748	Fix/Floating Point	375	448	1.8,3.3	1.2	$26.63
Analog Devices	BlackFin+ BF707	Floating-Point	400	512	1.8, 3.3	1.1	$21.01
	SHARC 21479	Floating-Point	266	1280	3.3	1.2	$20.76
	SHARC 21363	Floating-Point	333	384	3.3	1.2	$20.83

The leading factors for deciding which chip to lean toward, came down to price ease of use in development, and used in audio processing. The C6720 was chosen because it has the lowest price for Floating-Point Arithmetic capability, and it is used often in audio processing.

The benchmark that applies most to implementing with the VTVD is the Fast Fourier Transform (FFT). The FFT is simply a very efficient algorithm to implement the Discrete Fourier Transform (DFT). The DFT is a wonderful mathematical procedure for determining the frequency content of a time-domain sequence, but is very inefficient. The FFT changed this completely. Most theory of the FFT and DFT will be skipped due to its many citations in text books and applications throughout. However, it is worth observing the amount of multiplications and additions that would occur for the DSP chip to execute. Equation 3 shows the DFT. The FFT drastically reduces the number of complex multiplications. For a DFT, with N-points, N^2 complex multiplications occur. Yet with the FFT, for N-points, the number of complex multiplications reduces to approximately $\frac{N}{2}\log_2 N$. For large N values, the difference is quite large. Looking at the equation should also give indication of how memory and mathematical operations make this such a good benchmark to test DSP chips.

$$X(k) = \sum_{k=0}^{N-1} x(n) W_N^{-kn} \text{ , } where\ k = 0,1,\cdots,N-1 \text{ , } and\ W = e^{j\frac{2\pi}{N}} \tag{3}$$

Having FFT capabilities is required to implement the VTVD. Instead of analog, this will take real time signals (the voice), and transform them to extract the frequency of that signal, and be able to very quickly send communication to turn on the corresponding LEDs and color to display. Most DSP chips come with both ADC and DAC, and almost all have built in FFT algorithms with the software given. Yet implementing and prototyping can be complex and expensive. There doesn't seem to be much open sourced software for anything DSP related, and, as seen by Table 8, experimenting can be costly. With these concerns, it can be seen why choosing a microchip versus a DSP is not a simple choice concerning the VTVD.

3.8. LED Display and Color Design

The LED output is the most important aspect of the visual design of the VTVD. With this, a great deal of consideration is given for overall design, placement, enharmonic equivalent choices, and color palette. The design choice to give optimal visual representation of each note (i.e. A, B, C, D, E, F, G), was a $5x4$ matrix layout of the LED's. The left side of Figure 8 shows the basic matrix mapping that will be used to display each note. The forty-five-degree angle of each LED shall allow for more continuity when viewing any letter that may have curved edges, such as B, C, D, and G. Along with displaying the seven natural notes of the western music scale, the sharps and flats that exist between these notes must also be taken into consideration. These notes, that exist between their natural counterparts, add complexity in that they have two names corresponding to the same note. This occurrence is called enharmonic equivalence and exists for the remaining five notes of the western scale: $A^{\#} = B^{b}, C\# = D^{b}, D^{\#} = E^{b}, F^{\#} = G^{b}, G^{\#} = A^{b}$. It would be redundant to have a visual option for displaying either a sharp or a flat, and would be cumbersome to include coding to allow for options for both. The final decision when choosing a sharp or a flat is spacing for the LED matrix and attempting to use the smallest number of LEDs to still communicate well what note is being produced. The simplest solution was to choose the sharp sign (#). This can be seen at the top right of Figure 8 and utilizes a $2x2$ matrix. When one of the five sharps are sung, all the lights will illuminate giving differentiation between a natural note, and a sharp (raised) note.

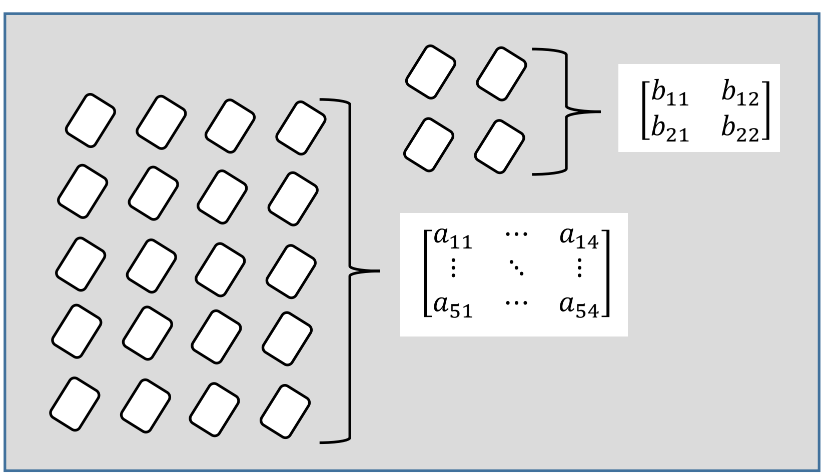


Figure 8. LED Board Configuration and Matrix Naming Convention

Not only will the VTVD present the notes as they are sung, but it will also display a color corresponding to each note. Table 9 lists all the twelve western notes, the mapped color to each note, and the matrix position to display the note visually. The choosing of colors for each note can be arbitrary, and is not limited to the twelve chosen here. They were selected from the traditional twelve position color wheel and have been mapped in sequential order, so that when singing ascending/descending melodic lines the colors will transition well from one to the other creating visual fluidity. In future prototypes of the VTVD, it would be beneficial to allow for the user to map the colors of choice to each note, and allow for steady solid colors to be displayed like that in any digital tuner today. Then the VTVD could also serve as a multipurpose unit, to be a visual display for performance, and at the same time be a valuable tool for users to tune to vocally.

Table 9. Color-Note Mapping

Note	***Corresponding Color***	***Matrix Position***
A	***Blue Green***	$a_{12}, a_{13},, a_{21}, a_{24}, a_{31}, a_{32}, a_{33}, a_{34}, a_{41}, a_{44}, a_{51}, a_{54}$
$A^{\#}$	***Green***	$a_{12}, a_{13}, a_{21}, a_{24}, a_{31}, a_{32}, a_{33}, a_{34}, a_{41}, a_{44}, a_{51}, a_{54}, b_{11}, b$
B	***Yellow Green***	$a_{11}, a_{12}, a_{13}, a_{21}, a_{24}, a_{31}, a_{32}, a_{33}, a_{41}a_{44}, a_{51}, a_{52}, a_{53}$
C	***Yellow***	$a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{41}, a_{52}, a_{53}, a_{54}$
$C^{\#}$	***Yellow-Orange***	$a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{41}, a_{52}, a_{53}, a_{54}, b_{11}, b_{12}, b_{21}, b_{22}$
D	***Orange***	$a_{11}, a_{12}, a_{13}, a_{21}, a_{24}, a_{31}, a_{34}, a_{41}, a_{44}, a_{51}, a_{52}, a_{53}$
$D^{\#}$	***Red-Orange***	$a_{11}, a_{12}, a_{13}, a_{21}, a_{24}, a_{31}, a_{34}, a_{41}, a_{44}, a_{51}, a_{52}, a_{53}, b_{11}, b$
E	***Red***	$a_{11}, a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{32}, a_{33}, a_{41}, a_{51}, a_{52}, a_{53}, a_{54}$
F	***Red-Violet***	$a_{11}, a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{32}, a_{33}, a_{41}, a_{51}$
$F^{\#}$	***Violet***	$a_{11}, a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{32}, a_{33}, a_{41}, a_{51}, b_{11}, b_{12}, b_{21}, b_2$
G	***Blue-Violet***	$a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{33}, a_{34}, a_{41}, a_{44}, a_{52}, a_{53}, a_{54}$
$G^{\#}$	***Blue***	$a_{12}, a_{13}, a_{14}, a_{21}, a_{31}, a_{33}, a_{34}, a_{41}, a_{44}, a_{52}, a_{53}, a_{54},$ $b_{11}, b_{12}, b_{21}, b_{22}$

Figure 9 shows three examples of how the VTVD will display notes and their mapped colors. The brightness of each color should correspond to how loud the performer is singing, while still being able to visually see the color and the note displayed. In future iterations and development of the VTVD, the display would also be able to have effects such as fading the note in, fading the note out, displaying the note sequentially from left to right, up and down, or even having different pulses and intensities. For instance, have the note B, pulse 3 times with each iteration getting brighter and brighter.

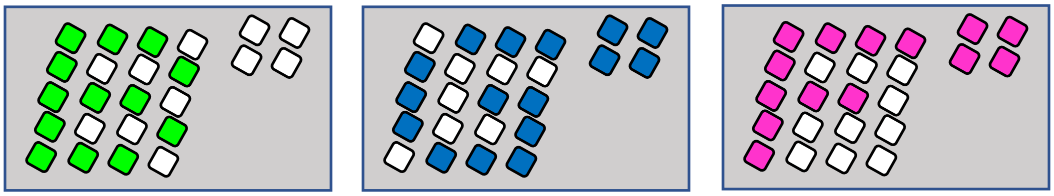


Figure 9. LED Examples, from left to right, B,G^#,F^#

3.8.1. LED NeoPixel Matrix

The Adafruit NeoPixel matrix is an 8 by 8 array matrix that contains 64 RGB LEDs. Each pixel is individually addressable and only one microcontroller pin is required to control all the LEDs. This matrix was the design that was ultimately used. Since programming the system would need to include controlling each of the LEDs. Wiring up the matrix requires two 3-pin connection port. The ports include the 5VDC to the +5V, the ground pin, and the DIN pin to the microcontroller to write the matrix. Each LED can draw 60mA if all LEDs are on bright white.

3.9. Bluetooth

Bluetooth is a wireless technology standard for exchanging data signal over short distances from fixed and mobile devices, and building personal area networks (PANs). Bluetooth is also a wire-replacement communications protocol designed for low-power consumption on low-cost transceiver in each device. Bluetooth was invented by telecom vendor Ericsson in 1994, it was originally conceived as a wireless alternative to RS-232 data cables [27].

Because the devices use a radio (broadcast) communications system, they do not have to be in visual line of sight of each other; however, a quasi-optical wireless path must be viable. Range is power-class-dependent, but effective ranges vary in practice [28].

There are four official classes of Bluetooth. The classes are determined by the types of distances and permitted power. Officially Class 3 radios have a range of up to 1 meter (3 ft.), Class 2, most commonly found in mobile devices, 10 meters (33 ft.), and Class 1, primarily for industrial use cases,100 meters (300 ft) [29]. Class 4 has the shortest range with 0.5 m; therefore, it is rarely use in the market. Bluetooth marketing qualifies that Class 1 range is in most cases 20–30 meters (66–98 ft), and Class 2 range 5–10 meters (16–33 ft) [30].

To satisfy the requirement of range and speed, there are different versions of Bluetooth Module. However, Version 3, 4, and 5 are commonly use. The table below shows the comparison between those three versions [28]:

Table 10. Comparison Between Bluetooth Versions

Bluetooth version	Maximum Speed	Maximum Range
3.0	25 Mbit/s	10 meters (33 ft.)
4.0	25 Mbit/s	60 meters (200ft.)
5	5 Mbit/s	240 meters (800ft.)

For our project, Bluetooth version 4.0 should be more than enough for wireless connection specification.

3.9.1. Application:

Bluetooth is one of the most convenient wireless device which have multiple applications in real life. Bluetooth has a huge potential in synchronizing information in localized area. Potential of Bluetooth application is tremendous. Nowadays, people are communicating within a closed area instead of communicating with those who are far away. The closed communication is commonly in business office or school. The following list shows some of the potential applications of Bluetooth; however, it will be soon more imaginative applications in the future [31].

When one considers how to do away with complex task of networking between computing devices in the office, installing a Bluetooth network will solve the issue. Each Bluetooth device could be connected to at least 200 computing devices. All the connections are wireless; hence, there will not have any tedious wires wiring throughout the office. Since Bluetooth connection support both point to point and point to multiple point, it will virtually make the maximum number of simultaneously linked devices unlimited [31].

The Bluetooth technology connects all the office devices wirelessly. For instance, connect your PC, iPad, iPhone to printers, scanners, or faxes without multiple cable attachments. All the office devices now come with Bluetooth module built in. In addition, mouse, keyboards, and headphones wirelessly connection to PC increase the flexibility of carrying them all around [31].

Nowadays, most of the updated digital cameras has Bluetooth function. One can stream or send the images or videos directly to the mobile phone. Therefore, he can take picture from distance with the camera wirelessly [31].

Most of the accident was caused by using the phone while driving. Bluetooth function reduces the rate tremendously by connect wireless connection from the phone to the car. One can make a phone call or changing the music without touching the phone. This will increase the caution of people while driving on the street [31].

Apple Watch becomes the best-selling watch from 2015 to 2017. What makes the smart watch becomes so popular? The reason is its convenient and flexibility by connecting Bluetooth to the iPhone. One can make a phone call, read a message, or make an alarm without using the phone.

3.9.2. Bluetooth 4.0 + LS:

By the rate and the range requirements to connect the electronics device to the karaoke microphone, Bluetooth version 4.0 should be the ideal choice.

Bluetooth version 4.0 of the Bluetooth Core Specifications was adopted by the Bluetooth SIG on 30 June 2010. It includes Classic Bluetooth, Bluetooth high speed and Bluetooth low energy protocols. Bluetooth high speed is based on Wi-Fi, and Classic Bluetooth consists of legacy Bluetooth protocols. It introduces support for collecting data from devices which generate data at a very low rate. The main intent of this feature, called Low Energy (LE), is to aggregate data from various sensors, like heart rate monitors, thermometers etc. Just like Bluetooth version 3.0, version 4.0 introduces for an alternate lower layer; for instance, all the applications that were available with Bluetooth radio earlier can be run over an alternate radio [32].

Capabilities - Basic rate + EDR (optional) + HS (optional) + LE (optional)

3.9.3. Possible Bluetooth Modules:

There are some common Bluetooth modules that already in the market. In this section, we will discuss the comparison between those modules to choose the best one that fit the project

3.9.3.1. Bluefruit LE nRF8001:

The Adafruit Bluefruit LE nRF8001 is a Smart Low Energy Bluetooth module version 4.0. It allows you to establish an easy to use wireless link between your Arduino and any compatible iOS or Android device. It works by simulating a UART device beneath the surface, sending ASCII data back and forth between the devices, letting you decide what data to send and what to do with it on either end of the connection [33].

The nRF8001 is a very good candidate because it handles all the BLE radio and low-level work. In addition, nRF8001 transmit all the signals over SPI which makes it easy to use with any kind of microcontroller. The input voltage can vary from 3V (standard) to 5V (maximum) which is perfect fit for out DC power battery source [33].

3.9.3.2. Silicon Labs BLE112-A-V1

Ideally, the Bluetooth module has low power consumption, short range, and small. BK8000L Bluetooth module is the best fit. It provides high quality sound and compatibility. Moreover, it has SBC audio decoding performance, auxiliary connection function as back up, and the size of a quarter. The digital signal of the any electronic device will go to the Bluetooth module. It will convert to analog sign then amplified by LM386 circuit before going to the speakers. [34].

BK8000L Bluetooth Module is 2.1 type with compliant. The Bluetooth module contains integrated stereo ADC and DAC, five bands hardware equalizer, digital equalizer for stereo line in, and integrated full duplex hands-free speakerphone.

The Bluetooth module also has the function for auxiliary connection in case the wireless function is not operated. However, the analog signal is still very low before it goes to the speaker. Therefore, a small external amplifier using IC chip LM386 will connect with BK8000L. Then the signal will go the left and right channels through 5 Watts speakers. [34].

3.9.3.3. HC-06 Bluetooth RF transceiver module:

This Bluetooth module with the input voltage from 3.3 – 6V can connect directly to any micro controller. With the 2.4GHz frequency digital wireless speed and data rate from 2Mbps - 3Mbps, HC-06 module is really good candidate for Bluetooth module choice. The size of HC-06 is very small (27mm×13mm×2mm) that could fit in the PCB board for the microphone. However, this module is very cheap compare with the other two, only $3.

3.9.3.4. *Comparison and Conclusion:*

The table below is the comparison of between nRF8001, BK8000L, HC-06 Bluetooth modules:

Table 11. Comparison between Bluetooth v4.0 nRF8001 and BLE112

	nRF8001 [33] 	BK8000L [34] 	HC-06 Bluetooth  [35]
Frequency	2.4 GHz – 2.48 GHz	2.4GHz -2.48 GHz	2.4GHz
Data Rate	2Mbps	2Mbps	2Mbps - 3Mbps
Serial Interface	SPI, UART	I2C, UART	SPI, UART
Voltage Supply	3V - 5V	2.8V – 5V	3.3V – 6V
Current Transmitting	100mA	36 mA	30～40mA
Dimension	29mm x 28mm x.8mm	13.5 mm x 25 mm x 1.5 mm	27mm×13mm×2 mm

3.10. Auxiliary Core

An auxiliary port (AUX) is the logical name for a standard communications port which commonly see in any electronic device. AUX is an asynchronous serial port with an interface that allows the auxiliary input of audio signals for: speakers headphones, portable music players. [35]

Our group has found that an auxiliary port is primarily used to permit a personal computer or any electronic device with the appropriate serial communication port to transmit and receive data in bits. Generally, the AUX port on a PC is computer port 1 (COM1), which is the first serial port with a preconfigured assignment for serial devices [35].

The AUX port is typically used for audio equipment that transmit the sound sources, such as digital music players or audio speakers. The peripheral sound device is connected to an AUX port or other medium such as a vehicle's audio jack [35]. An auxiliary port is also called as an auxiliary jack or auxiliary input.

3.10.1. Design Standards

From the research, there are no specific standards for auxiliary ports in the electronics devices. However, there is a research about the serial port standard in the document Recommended Standards 232 (RS-232). In the document, they define the minimum requirements in general electrical components standards, the layout of each pin connector found within the port, the physical size constraints of the pin, and the timing and signal configuration of the serial port. Overall, this document informs the most updated technological advancements which could provide the minimum requirements to reach the competitive consumer market. The technology has been improving every day; hence, the minimum designs specifications of pin design and data storage for high speed communication must meet the updated requirements.

3.10.2. Audio Jack

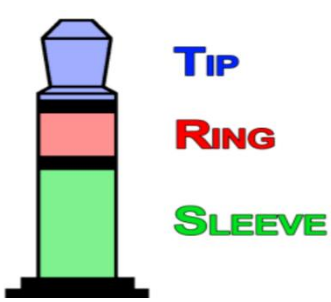
To interface audio in our circuit to the headphone, there are several types of audio connection. For one, we learned that one of the most flexible ways to implement the input audio into our circuit design was to use a stereo audio jack. The two most common types of audio jacks are 3-pole (TRS) and 4-pole (TRRS) plugs and sockets connections [36], where:

T stands for Tip - R stands for Ring - S stands for Sleeve.

3.10.3. TRS type audio jack:

TRS audio jacks are generally used for stereo headphones, mono headsets, stereo microphones, auxiliary cables, etc. which do not have the microphone. Since there will be a connection between the electronic device such as PC or smart phone to the portable microphone, TRS audio jack is a good choice. Because the acoustic vocal to the microphone is separated to the music from the electronic device to the speakers. Hence, the portable microphone will act as speakers when connect to PC or cellphone. In addition, the poles on these are usually straight forward, but it is possible they could vary [36].

For the TRS connection, the pinout usually goes as follows:



Tip = Left Channel

Ring = Right Channel

Sleeve = Ground / Common

Figure 10. TRS Audio Jack Configuration (Permission Requested)

3.10.4. TRRS audio jack

TRRS is commonly used for stereo headsets, such as cell phone headsets with an inline microphone, audio/video cables with stereo audio, etc. TRSS audio jack usually used in iPhone or Android device's headphones which contain the microphone. The catch with the TRRS is the poles can vary between CTIA (newer) and OMTP (older) standards. In the group discussion, the best option would be one set of headphones that we can use across multiple devices. Choosing to include a more versatile design could potentially help the design in consumer market popularity [36].

For the TRRS connection, they follow two standards:

CTIA (Newer and commonly found in iPhone's headphone) OMTP (older)

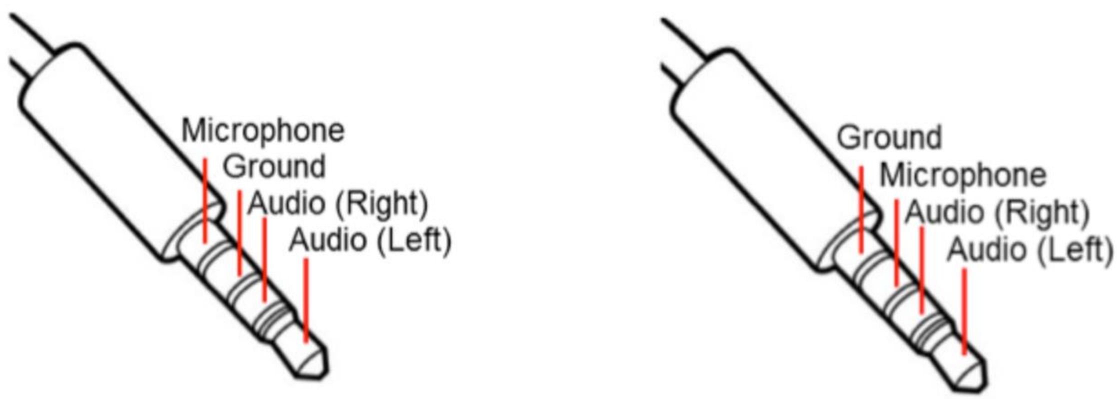


Figure 11. TRRS Audio Jack Configuration (Permission Requested)

This is very important of choosing the right types of TRRS connection; otherwise the device will not work. For example, there is a microphone that is wired according to the CTIA Standard, therefore the sleeve is the Microphone and the adjacent Ring is the Ground/Common. The microphone inside is an electret microphone, which it has a positive and negative polarity. Hence, if the polarity connection reverses, the electret microphone will not work. So, if you plug this microphone into an OMTP Standard device, the polarity will be reversed. In the case of a headphones with an inline microphone, such as iPhone or Android device headphone, the Left and Right channels will not have a ground connection and therefore will also not work. They will instead be connected to the microphone and that's useless.

3.10.5. Auxiliary Input Voltage Range

The voltage range for an auxiliary input should always be within 0 Volts to the reference voltage. Higher voltages would cause our converter and processor component to overheat, potentially damaging the overall circuit of the project. Further, the rise of temperature will also potentially cause damage to the user. To prevent the damage that might happen, this is necessary to research several types of AUX ports which can satisfy our design requirements.

3.10.6. Sound Quality

The main difference between a USB connection and an auxiliary input (aux) is that the USB sends unprocessed digital data to the head unit, and the aux input sends a processed, analog audio signal. In other word, the USB cable transferring data bit by bit like transferring to a computer, and the aux cable transferring an audio signal like one hear from the earbuds.

One advantage of aux connection is that will provide more utility due to wider range of device available. However, the sound quality of a USB connection is better than auxiliary port. The fact is that the head unit (speakers) is almost certainly better at turning digital files into analog audio. In addition, USB allows for more functionality such as control playback. On the other hand, auxiliary jacks are only transfer analog audio signal to the head unit [37].

Thus, if the choice of USB connection, the signal will go through digital to analog converter. The converter could also be used for signal from Bluetooth module. However, when an analog signal from auxiliary connection go through the ADC to digital amplifier and then to DAC, the sound quality is much worse than when a USB connector uses the digital to analog converter to process data due to the amount of noise susceptibility that exists [37].

In general, auxiliary inputs and USB are both good ways to connect a phone or MP3 player to speakers, example car stereo. However, there will be a huge difference in quality based on the DACs involved. This is because an aux connection utilizes the DAC in your phone or MP3 player, while a USB connection allows the DAC in your car stereo to process data located on your phone or MP3 player [37]. Hence, the sound quality from USB connection is much better than the auxiliary inputs.

3.11. Power:

For this project, the selection of size and capacity of the power source is very important. Rechargeable batteries are recommended. Even though rechargeable batteries initially cost more than disposable batteries, they have a much lower total cost of ownership and environmental impact, as they can be recharged inexpensively many times before they need replacing. The batteries will provide the power source to every single component and each of them will require voltage and size of the accessory. There are several types of rechargeable batteries in the market. The selection of batteries for this project will be small and light, 5V and 22mAh. The available power source options are solar power, lithium ion battery, nickel cadmium battery and USB B. To choose the right one, there are several parameters that need to be considered. The ADCs, DACs, and the processor needs 3.6 V to operate which will use 400mW of power. Moreover, most of the micro devices were in their low power mode will use around 200mW.

3.11.1. Solar Power

A 5V solar panel costs around 4 dollars to 60 dollars. The price of the solar panel is determined by the material and power delivery. The maximum power delivery has a large range depending on the price. It will need around 200mW to 400mW. On the other hand, material of the solar panel will also affect the price. For example, an amorphous solar panel would cost around 38 dollars, more than regular solar panel, because it is more environmental friendly and more resistance to be cracked [38]. For cheaper solar panel which cannot delivered enough power, it is possible to connect them. Therefore, it will be very cumbrous and hard to carry around. The tradeoff is it will make the product looks unattractive to the consumer. The solar panel has an additional feature. The solar panel can work light a light sensor. In future development, an LED will signal when the solar panel is charged or not. All the information present above such as numerical or properties of the solar panel are from the datasheet the company provides [39].

3.11.2. Lithium ion Battery

A typical 12V lithium ion battery can cost 20 to 25 dollars. They have different sizes, shapes, and capacity. The power delivery of the battery depends on the cost. For instant, 5 V battery will cost around 6 dollars which will deliver at least 400mW of power. On the other hand, the reliability and durability of the battery are important to be considered. The 1570 by Adafruit is ideal to be used in this project has around 500 recharge/ discharge cycles. To be said, a typical 5V lithium ion battery can be a good power source option for this project. For a microphone, the size of the battery is very important. First, the battery cannot be custom made. The smallest size of lithium battery is greater or equal than one inch in length which was the anticipated length of the project. The specifications of the microphone will be changed to accommodate to this type of battery. To recharge a battery, a port will be added on the device. The port will not affect much on the product. All the information present above such as numerical or properties of the solar panel are from the datasheet the company provides [28].

3.11.3. Nickel-cadmium battery

A typical Nickel-cadmium battery cost around 1 to 2 dollars. This is the cheapest option for the project. Moreover, this is a good option to lower the cost of the product compare with other power source. The power delivery from two batteries should be enough for the microphone. The size of nickel-cadmium battery is well fit in the device specifications. The handgrip of the microphone is around 4 inches in length which will fit perfectly 2 inches battery. However, this battery is usually non-rechargeable. For 5V batteries, they will last about 40 hours used. The battery would have to be replaced periodically every week for daily use. It should not be a problem since the batteries are very easy to replace and the cost was not very expensive. For a long run, there is an option to use nickel metal hydride which will be rechargeable. Therefore, it will be more cost for the port to recharge the battery. For this project, 5V nickel metal hydride batteries will be used since the rechargeable device had been very popular in the market. [40]

3.11.4. Micro-USB B

Micro-USB B is the power supply and supplies the audio. The cost associate would be the Micro-USB B port and cord which can range from 2 to 5 dollars. One great benefit is that it can supply power as well as send audio/video signals. The cable can deliver at most 100W of power using Thunderbolt 3 protocol. [41] The negative side of this implementation is the user's device battery will be drained very quickly. There are two common audio devices are the user's computer or phone. If it is a computer, there will be no issue because it will have a large capacity battery. The main issue to provide a user with the best possible experience is to limit battery drainage from the user's phone. The default voltage that Micro-USB B delivery is 5V and can supply up to 20 V and max of 5A. [41]

An additional benefit is the size of Micro - USB B is one of the smallest power supply with less than 1 inches in width and length. A typical port has a height of one fourth an inch or smaller. Further research will have to be conducted in later section to estimate the effect the minimum voltage and current to power the device would affect a typical phone. All the information present above such as numerical or properties of the chip was taken from the datasheet the company provides [41]

3.11.5. Conclusion & Comparison

Based on the criteria of reliability, efficiency, size, and cost the best power supply option is using Micro-USB B port and nickel metal hydride batteries. The reliability of using nickel metal hydride is that it does not require replacing new battery continuously. Moreover, Micro USB B will always be functional as long as the device is plug in the power source, although the batteries dies. It is most likely will become the new standard. The only other option that did not require a port to charge was the solar panel, but the solar panel is very expensive compare with other supply. Micro- USB B also supply the audio from electronic device such as computer and phone; it will replace the aux cord.

Table 12. Comparison between different types of power source

	SLMD121H04L 	N-700AAC 	ICR18650 
Voltage Rated	2V	1.2V	12V
Battery Chemistry	Solar Cell - Monocrystalline	Nickel Cadmium	Lithium Ion
Capacity	44.6mAh	700mAh	9.8Ah

Charge Time	N/A	16hrs	5hrs
Dimension	1.693" L x 0.551" W x 0.079" H	0.55" Dia x 1.96" H (14.0mm x 49.8mm)	1in x 3in x 7in
Weight	2.5g	23.13g	200g
Price	$6.8	$1.88	$20

The table above are some power sources comparison between solar cell, rechargeable lithium ion battery, and nickel cadmium battery. Therefore, Nickel metal hydride batteries and Micro USB B are the most efficient power delivery method as it can be configured to provide the most efficient voltage and current. Due to the batteries size and thinness of the USB port, it is the only option that will maintain the device specification and limit the necessary thickness of the product. Finally, the combination will be the lowest and most reasonable cost of any other combination costing at max 20 dollars.

As discussion above, Solar power source won't be selected. The consideration is only between rechargeable Nickel Cadmium and Lithium Ion battery chemistry type. Even though one Lithium Ion battery is more expensive and heavier than one Nickel Cadmium battery; however, it delivers enough power to the system. If Nickel Cadmium battery wants to deliver the same amount of power, there will be at least 10 batteries combines which will make the power source heavier and bulkier. Hence, Lithium Ion battery 12 V 9800mAh is the best choice.

3.11.6. Power inside a phone

Micro-USB B and nickel metal hydride batteries were decided to be the power supply of the device. There will be an examination of the power storage to determine a range of time the device can be used and how much power the device will consume from the phone. A typical smart phone uses a single cell of lithium ion as it offers the best power supply due to is energy density and ability to be thin. It has a typical life span of 400 to 500 charge/discharge cycles. Furthermore, a typical battery can offer the user 5.45-watt hours of power. It is then more important that the device does not exceed more than 1-watt hour of power in order not to shorten and drain the user's phones battery. Even then it is believed that the device will not consume near 1 watt every hour and will have around 300 to 400 mW consumption [42].

3.12. Microphone

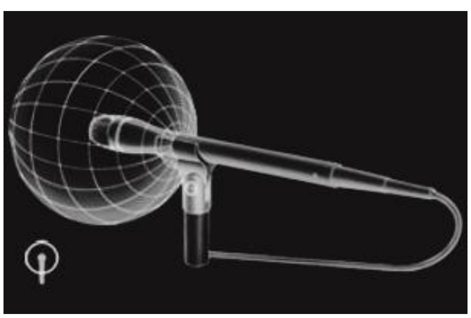
There are several types of microphones, which are determined by different methods of converting the air pressure variations of sound wave into electrical signal. The transformation is when sound waves hit the diaphragm of the microphone. [43] The vibrations of the diaphragm convert to current which will then be passed through an ADC. There three common types microphones are dynamic, condenser, and piezoelectric. [43] Dynamic microphones, which uses a coil of wire suspended in a magnetic field, have the advantage of being more resilient to the environment but lack the ability to pick up any rich sound. [43] Condenser microphones, which use the vibrating diaphragm as a capacitor plate, are more expensive compare to dynamic one and are prone to feedback. [43] Their only advantage is that they capture the sound more accurately than a dynamic microphone. Finally, the piezoelectric microphone, which uses a crystal of piezoelectric material, will most likely employ a dynamic microphone as sound quality can be sacrificed for cost and reliability. [43] The polar pattern of a microphone is the sensitivity to sound relative to the direction or angle from which the sound arrives. There are four most common types: Omnidirectional, Cardioid, Super cardioid, and bidirectional [44].

The omnidirectional microphone (Figure 12a) picks up sound equally from all directions. Therefore, the users do not have to aim in a direction. This makes it ideal for picking up sound from the surrounding environment. Lavalier microphones are the example for this polar pattern type. A disadvantage of omnidirectional pattern is that the microphone cannot be aimed away from undesired sources such as surround noise or sound from speakers which may cause feedback [44].

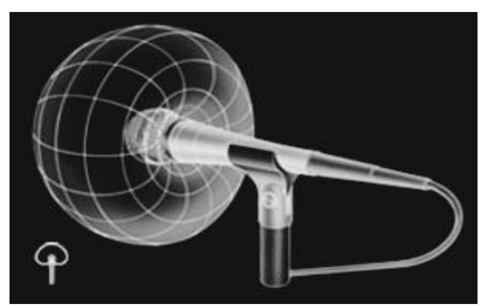
A cardioid microphone (Figure 12b) has the most sensitivity at the front and is least sensitive at the back. It isolates the unwanted noise and is more resistant to feedback than omnidirectional microphones. That makes a cardioid microphone ideally for a stage or in the auditorium where ambient noise is very loud [44].

Super-cardioid microphones (Figure 12c) have a narrower pickup than cardioids and a greater rejection of surrounding sound. However, they will also be sensitive at the rear. Hence speaker's placement will be important in this case. Super-cardioids are most suitable when single sound sources need to be picked up in loud environments just like cardioids. They are the most resistant to feedback [44].

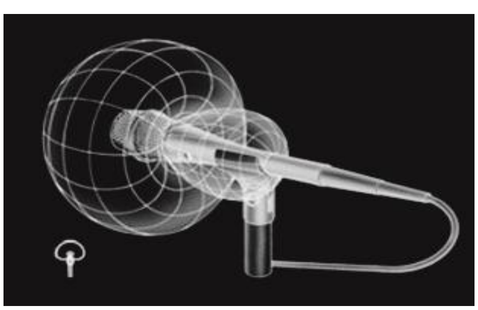
A microphone with a figure of eight polar pattern, or bidirectional pattern (Figure 12d), picks up the sound from the front and the back of the microphone and but not the side. Microphones with this figure of eight polar pattern are typically large diaphragm microphones [44].

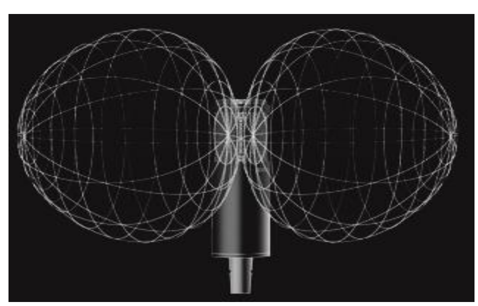


a. Omnidirectional



b. Cardioid





c. Super-Cardioid d. Bi-Directional

Figure 12. Four Common Types of Microphones (Permission Requested)

A major consideration of choosing the microphones is their frequency response. The frequency spectrum that it should pick up should fall within 20 Hz to 5kHz. From each microphones' datasheet, an EQ Vocals graph will provide the information to determine the sensitivity of the microphone for certain frequencies. Another consideration is the connectors used to connect the microphone to the device. There are different types of AUX connection.

A microphone produces tiny amounts of current. [43] For the processor to register the value of the current a pre-amplification process needs to be employed. To project the sound through speakers or to read the signal the microphone, the acoustic sound needs to go through the amplifier to increase the voltage. A final major consideration is the PCB design. There is a couple of microphones that already incorporate an ADC inside the chip. To create a more robust PCB, microphone should not have the ADC built in.

In summary, based on extensive research an omnidirectional condenser microphone should be the ideal microphone type for this project. The microphones should preferably have a frequency response of 20 Hz to 20 kHz. With a higher sensitivity to frequencies below 5 kHz. A pre-amplification stage should be incorporated. The microphones in the product should not have an ADC built in to create a more complex and extensive PCB for Senior Design.

3.12.1. Possible Microphones

The project will consist of one microphone. Depend on the size and application, the microphone of the project should be small and pick the vocal at every angle. The only option is electret omnidirectional condenser microphone. The reason of choosing that type of microphone is because it is small and fit to PCB board.

Below are three optional microphones that the group considered to be satisfactory to accomplish its role in karaoke microphone, and how it will work with the overall devices requirements.

3.12.1.1. CMA-6542PF

This microphone is electret condenser with omnidirectional device that produces an analog output. The device itself requires a supply voltage of 4.5 V (standard) to 10 V (max) [45]. The microphone has frequency range of 20 Hz to 20 kHz and output impedance of 2.2 kilo ohms. The device must be connected to at least two capacitors that act as bypass capacitors that are connected from ground to the voltage source. This electret 2pin microphone offers a low power consumption with a maximum current consumption of 0.5 mA. CMA-6542 PF has the dimension of

9.7 mm (diameter) x 6.5 mm (height) which is what the project specification is looking for to fit the device. The price of the device cost $1.01.

This device is an ideal candidate for the project, since it provides an analog output type that the Analog to Digital converter can be used to transmit the device into different digital amplifiers and to the processor. In addition, this microphone is easy to implement to the circuit with only 2 pins. As for the frequency range, we are looking for a range between 100- 5 kHz which makes this microphone ideal.

3.12.1.2. CME-1538-100LB

The second mems audio sensor is also analog microphone. This low power consumption microphone offers a 58 dB S/N Ratio which helps when already considered error of the component. The input voltage supply varies from 2V (standard) – 10 V (maximum) which helps with the project as there will be a constant 2-10 V voltage supply required by the system. With a size of 4mm (diameter) x 2.9mm (height), the device is small enough to fit within the project. This device costs roughly $ 3.16 per unit price and works within a 20 Hz – 20 kHz digital range which will fit perfectly for the ideal microphone. Lastly the microphone runs through external temperature range of -30 C to 80 C. One must be very careful when soldering the microphone to the circuit [44].

One big advantage of this microphone is waterproof, and dust protected. This project will be used mostly outside, wet environment is unavoidable. Therefore, this type of microphone is ideal candidate for this project. Moreover, the sensitivity is very good for a microphone (-38dB ±3dB @ 94dB SPL). This device is relatively new and meets common day standards [44].

3.12.1.3. CMC-2742WBL-25L

This microphone is also analog. It offers a 57 dB S/N Ratio which helps when already considered error of the component. The input voltage supply varies from 2V (standard) – 10 V (maximum) which helps with the project when the power supplies 3.7 V. With a size of 6mm (diameter) x 4.1mm (height), the device is slightly bigger than CME-1538-100LB; though it is still small enough to fit within the project. This device costs roughly $ 2.42 per unit price and works within a 100 Hz – 20 kHz digital range which is downside of this microphone. When human vocal frequency goes below 100Hz, the microphone will not take the voice [47].

One big advantage of this microphone is waterproof, and dust protected. Moreover, the sensitivity is very good for a microphone (-42dB ±3dB @ 94dB SPL). This device is relatively new and meets common day standards [47].

3.12.2. Comparison and Conclusion:

Overall three of the device are a good fit for the project. They have the wide range of frequency and high sensitivity. Noise cancelation might need to be studied and applied to the whole circuit. The price of the microphone is neglected compare with the whole project.

However, CMA-6542PF will be a better choice for this project. Most of the components such as microcontroller, converters, and processors will be operated at 2 - 10V and the power supply is 5V. This microphone will work perfectly with 2V – 10V voltage range. In addition, it is waterproof, and dust protected. CMA-6542PF fit perfectly to the project with the purpose of using under “outdoor” environment condition. Finally, with the size of diameter 9.7mm and 6.7mm height and have wire leads connectors, it is easy to solder to the PCB board.

Table 13 depicts the comparison between three microphones being considered for the system. Tabulated are the voltage ranges of each the frequency range, the sensitivity and the size of each microphones. Based on these parameters the choice was made for the device that would meet the requirements for microphone system.

Table 13. Comparison between microphone CMA-6542PF, CME-1538-100LB, and CME-1538-100LB

Electret omnidirectional condenser microphones	CMA-6542PF [45] 	CME-1538-100LB [46] 	CMC-2742WBL-25L [47] 
Output	Analog	Analog	Analog
Frequency Range	20Hz ~ 20kHz	20Hz ~ 20kHz	100Hz ~ 20kHz
Sensitivity	-42dB ±3dB @ 94dB SPL	-38dB ±3dB @ 94dB SPL	-42dB ±3dB @ 94dB SPL
Voltage Range	4.5V ~ 10V	2V ~ 10V	2V ~ 10V
Ratings	None	IP57 - Dust Protected, Waterproof	IP57 - Dust Protected, Waterproof
Termination	PC Pins	Wire Leads	Wire Leads
Size (Diameter x Height)	Dia 9.7mm x 6.7mm	Dia 4mm x 2.9 mm	Dia 6mm x 4.1 mm

3.13. Speaker

A loudspeaker is an electroacoustic transducer; which converts an electrical audio signal into a corresponding sound. Loudspeakers use both electric and mechanical principles to convert an electrical signal from a radio, television set or electric musical instrument into sound. For a loudspeaker to produce sound, the signal from the radio, television set, or electric musical instrument needs to be connected to an electronic amplifier.

Loudspeakers are usually built by using stiff paper cone, a coil of thin copper wire, and a circular magnet. The cone, copper wire, and magnet are usually mounted in a rectangle-shaped wood cabinet. The coil of copper wire moves back and forth when an electrical signal is passed through it. The coil of copper wire and the magnet cause the rigid paper cone to vibrate and reproduce sounds.

There are some speakers are considered for our project:

3.13.1. CLS0261MAE-L152

CUI's diverse line of speaker CLS0261MAE-L152 features diameters as small as 26 mm and depths as low as 4.6 mm. With sound pressure levels (SPL) is 88dBA and with a shape of cone types, the speakers are an ideal solution for engineers seeking durability and reliable performance [48].

The frequency range is from 540 Hz – 6kHz and the impedance is 8 ohms. The power rate of the speaker is 500 mW but the maximum is 1 W, that also be used for testing. For this project, there will be 3 speakers attached to 3 sides of a cube (1 side is for control board). Therefore, with the depth of 4.6mm, that would be no problem for this speaker. However, the price of this speaker is medium compare to other, which is $3.38 each piece [48].

3.13.2. ASE02008MR-LW150-R

ASE02008MR-LW150-R is the product of PUI Audio, INC. Instead of a shape of cone, this speaker has a square shape with the dimension of 0.787" L x 0.787" W (20.00mm x 20.00mm) and height at 5mm. With sound pressure levels (SPL) is 79dBA, the speaker is also an ideal solution for engineers seeking durability and reliable performance for the project [49].

Although it has very high range of frequency 800Hz ~ 20kHz but the efficiency is lower than L152 speaker, 79dBA. The power rate of the speaker is 500 mW but the maximum is 700 mW; the testing power rate is only 500mW. As mentioned, 3 speakers will be attached back to back, the square shape will create a problem when assembling. This will make the cube box of the microphone bigger to fit the speakers. The price for ASE02008MR-LW150-R is $4.05[49].

3.13.3. SP-1504

SP- 1504 is the product of Soberton, INC. Instead of a shape of cone, this speaker has a round shape with the dimension of 0.591" Dia (15.00mm) and height at 6.30mm. With sound pressure levels (SPL) is 93dBA, highest in all three options, the speaker is an ideal solution for engineers seeking durability and reliable performance for the project [50].

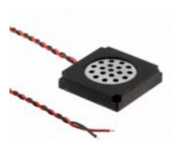
Although it has very high range of frequency 300Hz ~ 80kHz and the efficiency is 93dBA, the shape of the speakers makes it very hard to assemble in a cube box. The power rate of the speaker is 800 mW but the maximum is 800mW; the testing power rate is only 100mW. The price for SP-1504 is $2.31. The reason makes this speaker cheap is the material: Polyethylene Naphthalate (PEN), instead of Polyester, Polyethylene Terephthalate (PET) from the other two [50].

3.13.4. Compare and Conclusion

Table 14 shown below depicts the various speakers chosen for the system. Based on different identifiers the best speakers were chosen to meet the requirements needed to ensure good sound quality functionality for the output of the microphone sound. Hence the speakers chosen were based on the tabulated identifiers. The chosen CLS0261MAE-L152 device will meet the requirements.

In addition, with the long range of power rated 500mW and 1W for maximum, this speaker will avoid of power overrated. CLS0261MAE-L152 is a low-profile type speaker which is fit for the project.

Table 14. Comparison between speaker CLS0261MAE-L152, ASE02008MR-LW150-R, and SP-1504

	CLS0261MAE-L152 [48] 	ASE02008MR-LW150-R [49] 	SP-1504 [50] 
Type	Low-Profile	General Purpose	General Purpose
Efficiency	88dBA	79dBA	93dBA
Frequency Range	540Hz ~ 6kHz	800Hz ~ 20kHz	300Hz ~ 8kHz
Power Rated	500mW-1W	500mW-700mW	5W
Shape	Cone	Square	Round
Material	Polyester, Polyethylene Terephthalate (PET)	Polyester, Polyethylene Terephthalate (PET)	Polyethylene Naphthalate (PEN)
Dimension	1.024"Dia (26.00mm) x 0.181" (4.60mm)	20.00mm x 20.00mm x 5.00mm	0.591" Dia (15.00mm) x 0.248" (6.30mm)

Price	$3.38	$4.05	$2.31

Even though the price of CLS0261MAE-L152 is higher than SP-1504 and efficiency is lower, the material of Polyester, Polyethylene Terephthalate (PET) will make CLS0261MAE-L152 more reliable. Overall, CLS0261MAE-L152 is the best choice among those three chosen speakers for the project.

3.14. Digital to Analog Converter

One of the key components for the portable karaoke microphone project to work is a proper Digital to Analog Converter (DAC). DAC can make the system properly function without being too powerful that it consumes a lot of energy or takes up space. The digital to analog converter is required for processor to transfer the data that was received digital data from the analog to digital converter to the speakers for the user to hear the filtered and amplifier sound.

When looking for a required DAC to use, it is imperative that the D/A device is compatible with the requirements and constraints of the system. In addition, the requirement of this project required that the converter must be 16 bits dues to market predictions making it a standard. While researching the project, we know that the product is designed to be an open loop system, there will be no feedback; therefore, DAC need to be assigned very carefully. A proper representation of accuracy to use is the Integral nonlinearity (INL).

To prevent any significant noise within the system, the output impedance must be minimized. Hence, the DAC system that has an R/2R architecture will produce a lower noise compare with other DAC systems.

Another thing is to be considered is jitter. Although it is minimal and fast, jitter can affect buffering portions of the system. Since jitter is typically fast it is mostly not considered however for audio purposes it can cause an error and diminish user listening experience.

3.14.1. Audio DAC

Due to the requirements for the product, Audio DAC will be used converting the digital signal for analog sound purposes. This means that typically the DAC will already have the items listed above, such as an integrated circuit that is minimally distorted output and input, and can prevent minimal jitter. The higher resolution the higher the quality of the sound is. For this project, 16-bit DAC will be used to transfer the digital data to analog.

Sampling rates are typically a minimum of 48 kHz; however, due to the [51] advancement of these devices they are within a range of 96kHz and 192kHz basic standards. Typical SNR ratios for high quality audio devices are seen around 90 dB – 110 dB as a standard, where anything over 120 dB is exceedingly high end [51]. Since the input power source is 3.7V, the DAC power rated should be around less than 5V. In addition, the size of the DAC should be small to fit in the PCB board.

3.14.1.1. PCM 5242

One possible choice for a DAC is the 32-bit TI PCM5242 that accepts inputs of 16, 24, and 32 bits. PCM is a pulse code modulator which basically estimates what the analog signal is based on a range from the digital component. The accuracy depends on both noise and resolution. The PCM5242 Commonly used in DVD players and headset systems it requires a supply voltage from 3V to 3.46V. The device has can fend off jittering like most ideal devices. Although not low power device for TI's standards it has a max sampling rate of 384 kHz and an SNR of 114 dB [52].

PCM5242 has a size of 5 mm x 5mm and a price of $3.50 per unit. This device can be controlled using either I2C or SPI as their protocol. Overall the device meets the standards required for the product, and would be an ideal candidate for the device had it not been powered by another small device. Although commonly used in audio devices due to the nature of the supply voltage and the over qualification of the sampling rate it might be unqualified to produce and efficient device component. [52]

3.14.1.2. PCM 5100

The PCM 5100 has a dimension of 10.2 mm x 5.3mm 20 pin DAC that requires from 1.65 V ~ 3.46 V supply voltage. The device meets consumer standards with a sampling frequency of 384kHz along with an SNR of 100 dB. This device is very flexible when it accepts inputs of 16, 24, and 32 bits which was one of the requirements that the ADC had to be either 16 or 24 bits. The cost for the device is around $2.90 per unit, and can also be controlled using I2C or SPI [53].

One thing to note about this device is that it is commonly used as video players, along with Car audio systems. This device is more reasonable with the requirements that the system would like to meet, however it is also not low power even though it meets the desired specifications without over compensating too much. [53]

3.14.1.3. PCM1774

The PCM1773 DAC has a 16-bit resolution along with a low power feature that requires around 6.4 mW power when using a source of 2.4V which also makes this device quite efficient for the project requirements. The sampling frequency for this device can have a max of 50 kHz which is close to the minimum requirement that an audio DAC needs [54].

It is controlled by the I2C protocol or SPI and input receives 16 or 24 bits which as mentioned earlier is one of the demands the requirement would like from the ADC. The $3.19 device has applications that are typically used for cellphones and portable audio players due to its low power mode, and it meets the sampling frequency. In addition, the device has a SNR of 93 dB which can be classified as higher standards [54]

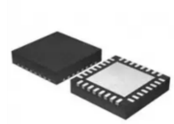
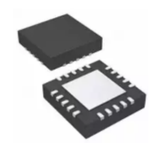
3.14.2. Compare and Conclusion

As discussed, we need to transfer the digital data to analog signal in at least 16 bits. Even though three devices satisfied the minimum requirement, the size is also matter. PCM 5100 is bigger than twice of the others and of course, it will not be the best option to choose. In addition, we also need to choose a DAC that will consume least power. PCM1774 has the lowest sampling rate 50 kHz, which is very closed to the minimum specification; however, it will save tremendous amount of power compare with other two devices. With the data interface of DSP and I2C, PCM 1774 is the best choice for Audio Digital to Analog Converter.

Table 15 shown below depicts the various possible DACs chosen for the system. Based on different identifiers the best DAC was chosen to meet the requirements needed to ensure the digital signal received from the Bluetooth device is converted to an analog signal for the addition of effects to the audio signal to be made. Hence the digital to analog converter chosen was based on the tabulated identifiers. The chosen PCM 1774 device will meet the requirements.

The PCM 1174 is fit for the project. Ensuring the conversion of the binary or digital code into an analog signal.

Table 15. Comparison between DACs PCM 5242, PCM 5100, and PCM 1774

	PCM 5242 [52] 	PCM 5100 [53] 	PCM1774 [54] 
Resolution (Bits)	32 bits	16/24/32 bits	16 bits
Sampling Rate (Per Second)	384kHz	384kHz	50kHz
Data Interface	I^2S, PCM	PCM	DSP, I^2S
Voltage - Supply	1.65 V ~ 1.95 V, 3 V ~ 3.46 V	1.65 V ~ 3.46 V	1.7 V ~ 3.6 V
Dimension	5mm x 5mm	10.2 mm x 5.3mm	4mm x 4mm
Power Dissipation (Maximum)	155 mW (at 3.3V)	148.5 mW (at 3.3V)	6.4 mW (at 1.8 V/2.4 V)

3.15. I2C Communication

The Inter-Integrated Circuit (I^2C) Protocol is a protocol intended to allow multiple “slave” digital integrated circuits (“chips”) to communicate with one or more “master” chips [W]. Like the Serial Peripheral Interface (SPI), it is only intended for short distance communications within a single device. Like Asynchronous Serial Interfaces (such as RS-232 or UARTs), it only requires two signal wires to exchange information [66]. The I 2C bus is a very popular and powerful bus used for communication between a master (or multiple masters) and a device [67]. There are two wires that the data can share. For example, in embedded system, the I2C communication is presented as the master (microcontroller) and the its slaves (I/O, LEDs.). The picture above from Texas Instrument is the example how the I2C work. In this picture, the microcontroller represents the I 2C master, and controls the IO expanders, various sensors, EEPROM, ADCs/DACs, and much more. All of which are controlled with only 2 pins from the master [TX]. There are two wires that use to control all slaves from the master. The two controller signals from two wires are: Serial Data Line (SDA) and Serial Clock Line (SCL). Data is placed on SDA when SCL goes low, and the data is read when SCL line goes high.

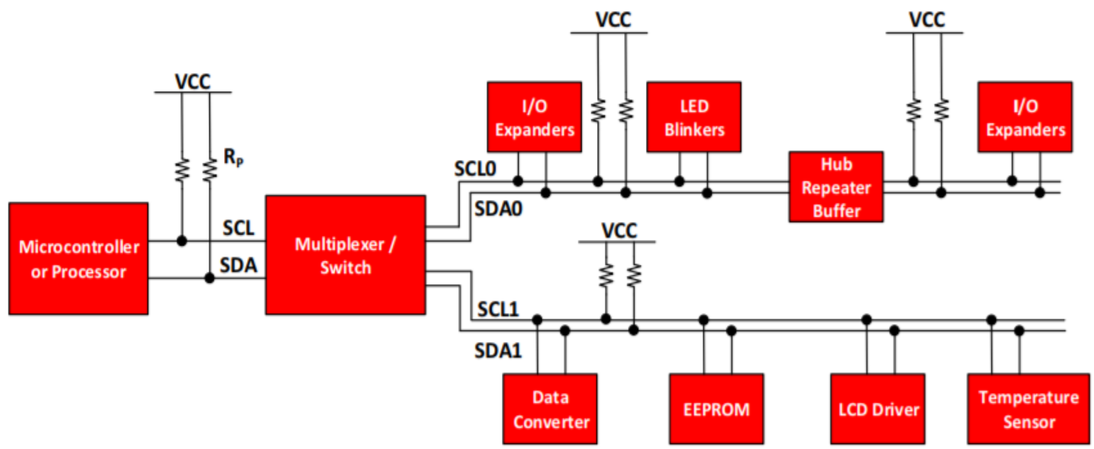


Figure 13: I2C Bus Example (Permission Requested)

3.16. Microprocessor (Analog)

Using a traditional microcontroller to process the audio signal has its own set of advantages and disadvantages. The two main advantages are familiarity and implementation. Already having familiarity with different types of microcontrollers such as TI's Code Composer studio environment as well as using multiple iterations of TI's microcontrollers gives a great advantage when trying to implement design. Arduino and several of its microcontrollers provide an endless resource because of the open source nature. With Arduino comes multiple venues and suppliers that provide parts to purchase specifically made for Arduino. Purchasing ATmega chips is uncomplicated and burning a bootloader isn't always necessary, depending on which chip the design calls for. At this point, both platforms, Texas Instruments (TI) or Arduino, will be considered if a microcontroller is chosen to use for the brains of processing. The other large advantage is implementation. Most of the DSP chips are expensive, require proprietary software and seemingly are complex in design and implementation. Both IT and Arduino are very straightforward. Software is free to use and relatively simple to run without much configuration. However, the large disadvantage comes with calculations itself. A DSP chip will be much faster in computations, and although audio processing isn't some of the fastest implementation (41kHz max), it still will require some configuring with a microcontroller as demonstrated in the following sections.

Figure 14 displays a simple decaying sinusoidal wave that anticipates the input audio from the user of the VTVD. Notice the simulation includes the ideal, pure tone sinusoid as well as its realistic counterpart; a sinusoid with noise. The issue with capturing an audio signal with a microcontroller is remaining strictly in the time domain for analysis. This poses several problems. The first and foremost is that the signal is not an ideal sinusoid. Not only is the sinusoid not ideal because of noise, but also it is not ideal because of amplitude decay. How shall the microcontroller deal with the decaying audio signal? One approach will be to base the validity of the signal on Thresholds. In Figure 14, two dashed lines represent potential threshold values that could be implemented for an incoming signal. Notice that if the threshold is too high, at 0.4, the signal will not be completely captured, and some information could be lost. At 0.2 as a threshold, a complete cycle would be captured, and possible analysis could be completed. However, even with this figure, the signal isn’t completely realistic to an audio signal. As it decays it starts to look less and less like a sinusoid in general. Care must be taken if choosing the microcontroller, for it must have the ability to quickly capture the incoming signal and apply thresholds that will dismiss undesirable ranges.

If not capturing and analyzing in the frequency domain, what would be the approach? The idea would be to extract or find one full period of the signal, and ensure that at least more than one period exists to exclude any noise that could be mistaken as an incoming signal. Once a period of the signal is confirmed, simply taking its reciprocal will yield frequency. When this is done, a general mapping can be written and tolerances for ranges of the detected signal will have to be set. Lower frequencies pose a problem with having a universal set of ranges because the frequency proximity are much closer and therefore require a tighter range than those in the higher frequency range.

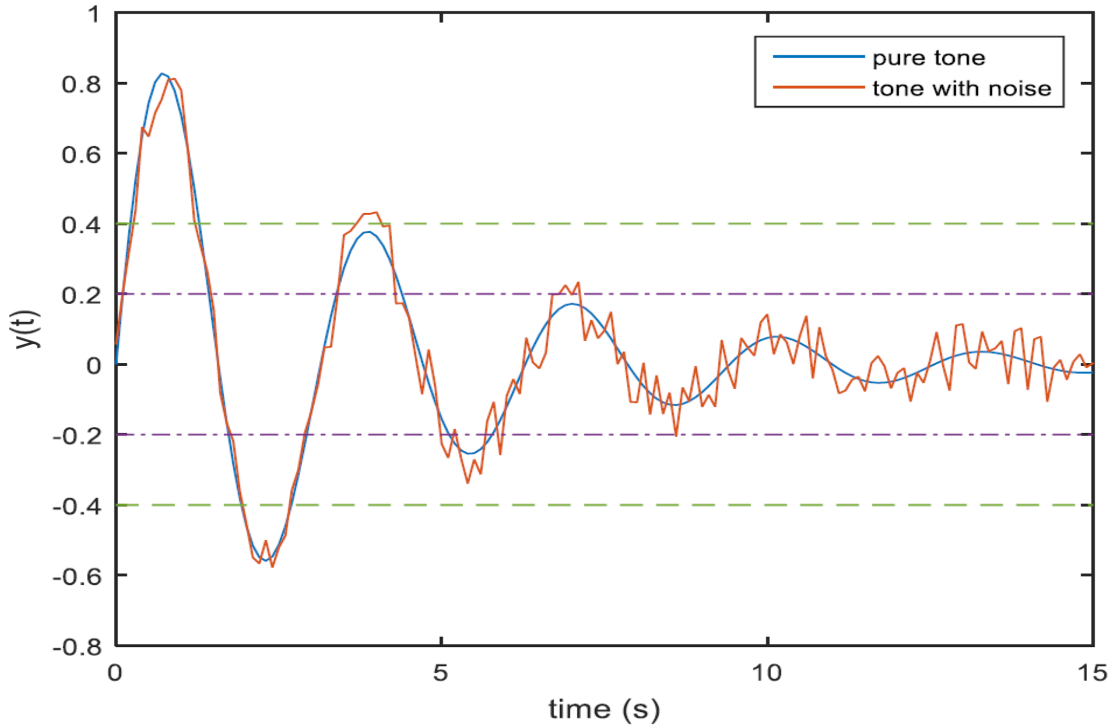


Figure 14. Simulation of decaying audio signal with and without noise

3.17. (VTVD) Power Distribution

When thinking about power a few questions arise, will the same power be supplied to all the devices? How will the power be regulated and what kind of power is being supplied to power the components that make up the VTVD, a few power specifications such as the voltage and current for each component had to be considered. Considering these specs two design options are being considered to power the components of the VTVD. The first design that is being considered is the dual power supply design using a dual rail supply op-amp. The second design being considered is the single rail power design using a single rail op-amp. In this section both power designs options and their advantages and disadvantages along with their different power supplies, will be discussed.

3.17.1. Device Operating Conditions

TableP1 tabulates all the essential different components that will be powered. Listed is the device along with typical or min and max values of current and voltage to calculate the maximum power intake to be used to power all the devices being used in the VTVD.

According to Table 18 the total current to consider is approximately 550Amps. This current amount will be taken into consideration for the design of the system. Depending on the operating voltage and current, the power may need to be distributed using different techniques to satisfy the operating conditions. Therefore, different aspects of the design are being looked at in more detail.

Table 16. Power Operating Conditions

	PER UNIT			TOTAL
	Min Voltage	Max Voltage	Max Amperage	
UA741C	±5V	±15V	2.8mA	
TLC5955	-3V	+5.5V	29mA	60mA
ADC121S101	+2.7V	5.25V	3.2mA	
MIC 1555	+2.7V	+18V	300µA	
ATMEGA 2560	4.5V	5.5V	2.0mA	
OPA376 (alternative)	2.2V	5.5V	1mA	
LEDs			20mA	480mA

Decisions will be made for which components will be best to meet the requirements of the VTVD.

UA741C

The UA741C is a general-purpose dual rail power supply operational amplifier with an offset-voltage null capability. The typical open loop gain is 106 dB when driving a 2000-ohm load. Has short circuit tolerance, offset voltage trimming, and unity-gain stability makes it useful for many applications. The voltage range for the UA741C is ±5V to ±15V to operate.

TLC5955

The TLC5955 is a 46 channel, 16-bit LED driver. Each channel has an individually-adjustable pulse width modulation (PWM), grayscale brightness control with 65,536 steps. The chip also offers constant brightness control for the LEDs. The chip needs around 3.0V to 5.5V to operate.

ADC121S101

ADC121S101 is a low-power single channel CMOS 12-bit analog to digital converter with a high-speed serial interface. The ADC121S101 is fully specified over a sample rate range of 500 ksps to 1 Msps. The single power supply with 2.7 V to 5.25 V range to operate.

MIC 1555

The MIC 1555 timer is designed to provide rail-to-rail pulses for precise time delay or frequency generation. The MIC 1555 can be used as an astable (oscillator) or monostable (One-shot). The MIC 1555 is powered from a +2.7V to +18V supply voltage.

ATmega 2560

The ATmega2560 is an 8-Bit Atmel chip with high performance. Has high endurance non-volatile memory segments, an advanced RISC Architecture with 135 instructions and up to 16 MIPS throughput at 16MHz. The chip takes roughly 4.5V to -5.5V to operate.

OPA376

The OPA376 is the alternative op amp. It is a single rail Texas Instrument op amp which will have the positive rail powered and the negative rail will be set to ground. The OPA376 is powered from 2.2V to 5.5V

3.17.2. Dual Rail Power Distribution Architecture

The first power design option is the Dual Rail. The block diagram shown below in figure 15 represents the design options being considered for the VTVD. Both design options are show how the power will be regulated and dispersed though out the system and to the separate devices. The VTVD will have a power source from a wall outlet of about 9 volts. Table 18 shown above helped to find the average max voltage needed to power all the separate components that make up the VTVD. As depicted in table 18 the average max voltage to power the components (op-amp, LED driver, ADC, timer, microcontroller, LEDs) of the VTVD is going to be 5volts. The figure below shows the dual rail power design and its workflow of how all the devices will be powered up.

According to Figure P1 the VTVD is powered by a wall wart. The incoming voltage will be 9volts. From here the VTVD will have two design options to operate the aforementioned parts in table 18 that will make up the VTVD. The dual rail design will need an inverter to supply negative voltage. Hence voltage regulators will be needed to satisfy the requirements needed to operate all the parts of the system. The VTVD will need two regulators, one for the DC inverter along with its configuration of components and the second, a step-down voltage regulator to drop 9volts to 5volts. This voltage was chosen based on the voltage ranges of each of the parts and finding the max voltage to operate all of them. This first design will mean having a bigger footprint and therefore more cost. The sections that follow go into detail of the regulators that may be used for the system.

The Dual Rail design option illustrated in Figure 15 shown in blue on the left side of the figure, depicts the workflow of the system along with its components. Noted for the Dual Rail design, 9 volts positive will be taken from a wall outlet as the main power source to the system. As illustrated in Figure 15 the voltage is inverted to obtained 9 volts negative, these will power the rails of the op amp. The voltage will then be stepped down with a regulator to 5V. The 5 volts will be used to power the rest of the devices of the VTVD.

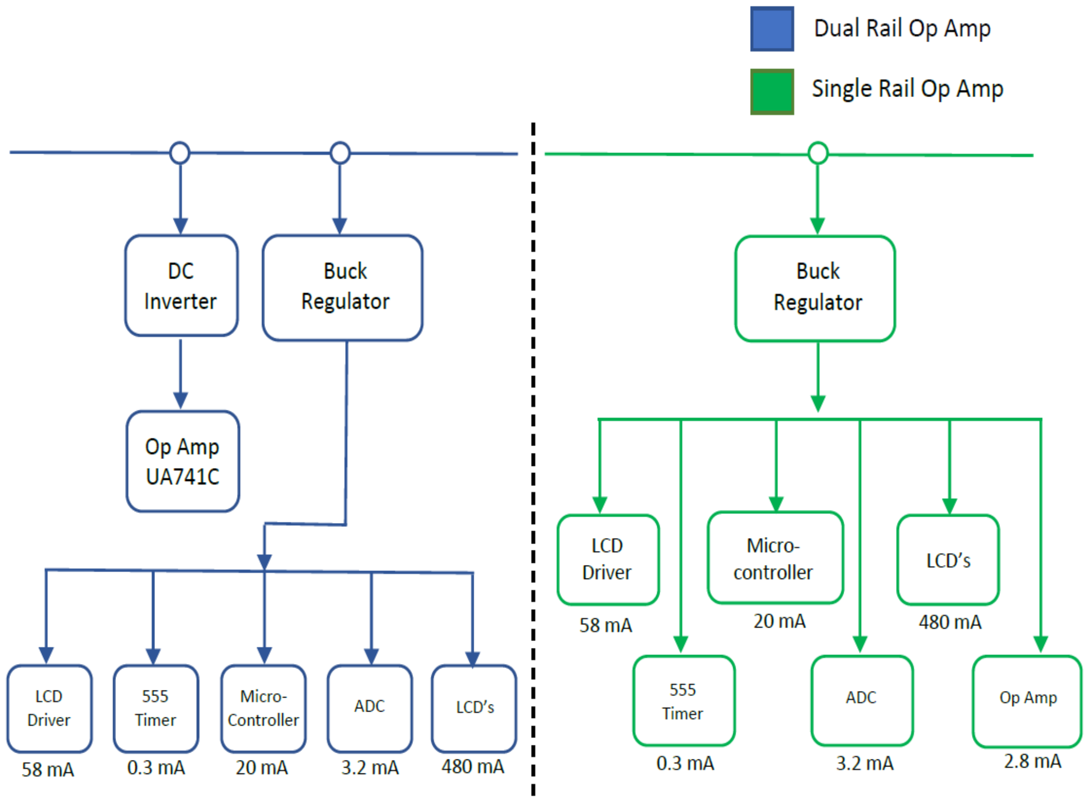


Figure 15. Power Distribution

The section that follows describes the regulators considered to step down the voltage to obtain the amount needed. As well as looking at the design in more detail.

3.18. DC Inverter

For the VTVD we will require a negative voltage for operation of the amplifier. Creating a dedicated negative supply rail would add to the cost and space of the PCB. Thus, it makes sense to generate the negative voltage from existing positive supply rails in the system. The way to do this would be to use a step-down regulator.

3.18.1. LM43601PWP

The LM43601PWP is a step-down DC-DC converter with high efficiency at light load. The part will be configured to operate as an inverting power supply. The regulator will generate negative voltage from a positive input voltage. The LM43601PWP is ideal for this design and will be used as the step-down DC-DC converter to supply the negative voltage to the dual rail op amps used in the dual rail design.

3.19. Step down Voltage Regulators

Voltage regulators take an input voltage and create a regulated output voltage at a fixed voltage level or adjustable voltage level. Voltage regulators come in three types. Linear Regulators are often used in low voltage, low power system.

Switching regulators are more efficient than linear regulators, but are more expensive and harder to deal with.

3.19.1. Linear Regulator

To stabilize and regulate voltage a standard 3-pin linear regulator such as the popular LM7805 could be used. Linear regulators work by adjusting the effective series resistance of the regulator based on feedback voltage, becoming a voltage divider circuit. [67] With this, the regulator can output a constant voltage without regard of the current load put on it, up to the current capacity it can hold.

One of the disadvantages to a linear regulator is the minimum voltage drop across the voltage regulator. The amount is usually very large, and on the standard LM7805 that is 2.0 volts. [67] The voltage drop is important because it is crucial in the power dissipated by the linear regulator. To get a 5volt stable output in the LM7805 for example, it would need roughly a 7volt input. Which

would mean it would have to dissipate 2 watts if it was delivering a 1amp load. [67] The larger the difference between the input and output voltage, the worse the power dissipation.

3.19.1.1. LM137

The LM317 is a commonly used three-terminal adjustable linear regulator. This linear regulator was considered because of its range for the output voltage. According to the datasheet, the LM317 has overload protection and is current limiting.

3.19.1.2. MIC5265

The second linear regulator being considered for this design is the MIC5265. Used for portable devices which made this regulator an interesting pick. It has a low drop out voltage and the supply voltage ranges from -5.5V to 2.7V. The chip offers low output noise which is ideal when filtering out noise.

3.19.2. Switching Regulator

Switching voltage regulators have higher power efficiencies or better compared to linear voltage regulators efficiencies that are often below 50 percent. [68] Switching regulators require extra components and the values have more impact on the performance of the regulator. One of the switching converter types being considered is the commonly used Buck switching converter. The buck is used to reduce a DC voltage to a lower DC voltage. This is important for the dual rail design and the implement of the design.

There are two topologies discussed in this report to understand further the implementation of the best voltage regulator to use for the system. Switching regulators can step-up (boost), step-down (buck) and invert voltages with ease.

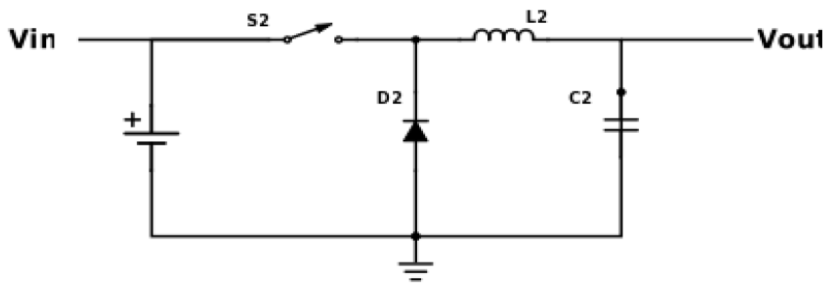


Figure 16. Basic buck-switching regulator circuit

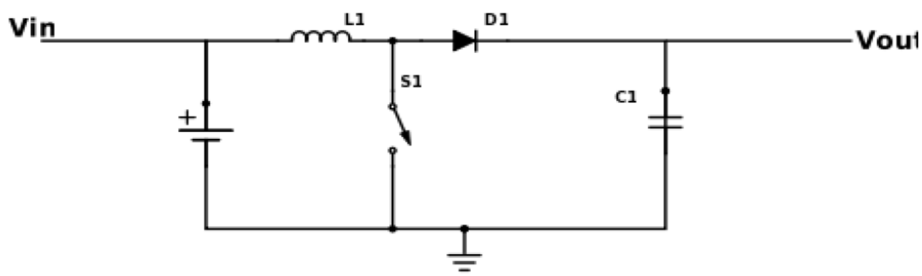


Figure 17. Basic boost-switching regulator circuit

The Buck converter uses a transistor as a switch that alternately connects and disconnects the input voltage to an inductor. [68] The current through the inductor goes up when the switch is on and ramps down when the switch is off. The DC load current form the regulated output is the average of the inductor current. [68] Table 19 tabulates the regulators being considered for this design option.

3.19.2.1. LM2596

The LM2596 is a voltage step-down (buck) switching regulator that will be used to step down the voltage from 9 to 5volts as shown in Figure 15. Capable of driving a 3-A load with line and load regulation. This device is available in fixed output voltages of 3.3V, 5V, 12V and an adjustable output version. Requires a minimal number of external components and is a simple to use regulator and therefore chosen as the step-down regulator to be used in this design.

3.19.2.2. MIC34063

The MIC34063 is one of the switching regulators that is being considered for this project. It is a Step-Up/Down/Inverting Switching Regulator. Per the datasheet the regulator has a low standby current, operates at a range of 3.0V to 40V, has an output switch current of 1.5 Amps and is current limiting. The operational frequency is 100kHz with an adjustable output voltage.

Table 17. Regulator Comparison Table

Model	Manufacturer	Type	Voltage Range	Price
LM317	Texas Instruments	Linear Regulator	1.25V – 37V	$0.15
MIC5265	Microchip Technology	Linear Regulator	2.7V – 5.5V	$0.24
LM2596	Texas Instruments	Switching Regulator	4.5V – 40V	$2.43
MIC34063	Texas Instruments	Switching Regulator	3V – 40V	$0.60

After researching both linear and switching voltage regulator chips the adjustable linear regulator would be the ideal choice, since it has a wider range of voltage and can be dropped at a bigger value. Some advantages of using a linear regulator is that it is low in complexity and cost. However even if linear regulators are great solutions for low power, low cost applications the biggest downside is that they are inefficient, which is where switching regulators can come into play. Hence when high efficiency is needed, or a wide range of input voltage is expected, including input voltages below the desired output voltage, the switching regulator becomes the best option.

For this project, the LM2596 is the voltage step-down regulator chip that will be used because it best fits the design with a supply voltage from 4.5V to 40V, the price however is the highest from among all the chips but is a great Buck switching regulator that could be used as through hole for prototyping testing, this is one of the few regulators that comes as a through hole component which it made very ideal. Even though the regulator might lack in efficiency to other regulators such as the MIC34063 we might adjust this later. The reason for choosing the LM2596 was mainly based on testing. Being able to use the through hole type chip and test the design and then transcribe it to surface mount for Single Rail Power Distribution Design The single rail design shown in Figure P1 on the right side of the figure indicated by green is the second power distribution design option being considered for the VTVD. It uses a single rail op amp which previously discussed in the report, the viable single rail op amp is the OPA376.

The reason this design is being considered is that there will be no need to implement a second regulator that will invert the positive voltage to negative. How could this be done? Referencing the TI precision designs with a single-supply amplifier the solution became available. What this second design does is, amplifies the ac signal and shifts the output signal so that it is centered at one half the power supply voltage. [69]

3.19.3. Single Rail Power Distribution Benefits

It is also noted in the TI reference that the input signal has zero dc offset and, so it swings above and below ground. [69] The key takes away and benefit of implementing such a design is because of this there is less components being used and therefore a saving in costs. The design also accepts signals which swing below ground even though the amplifier does not have a negative signal. This was ideal for the system.

3.19.3.1. Dual Rail / Single Rail Power Distribution Advantages

The overall conclusion made for the Power distribution model for the VTVD is that both the dual rail and single rail op amp designs will be implemented. Both options show promise and therefore both will be tested and designed. The reason this is being done is to take into consideration any failure of power being distributed correctly or any power failure all together. A switch may be put in place to toggle between both designs and see the accuracy of each. For the single rail op-amp the UA741 will be used. After careful consideration and research done for the operational amplifiers being used in the LED display the UA741C is the best option for the system, the op amp is powered in the range of voltages needed and hence we will try this op amp. For the single-rail option, the OPA376 is the op amp that will be used. This power distribution design allows for less components and less configuration in the system and thus less cost. By shifting the output signal so that it is centered at one half the power supply voltage, the DC offset is no longer a concern for the system. These modifications help to meet the design goals. the PCB. This way we guarantee the design will work.

3.20. Major Components

Picture below is all the major components that we purchased:

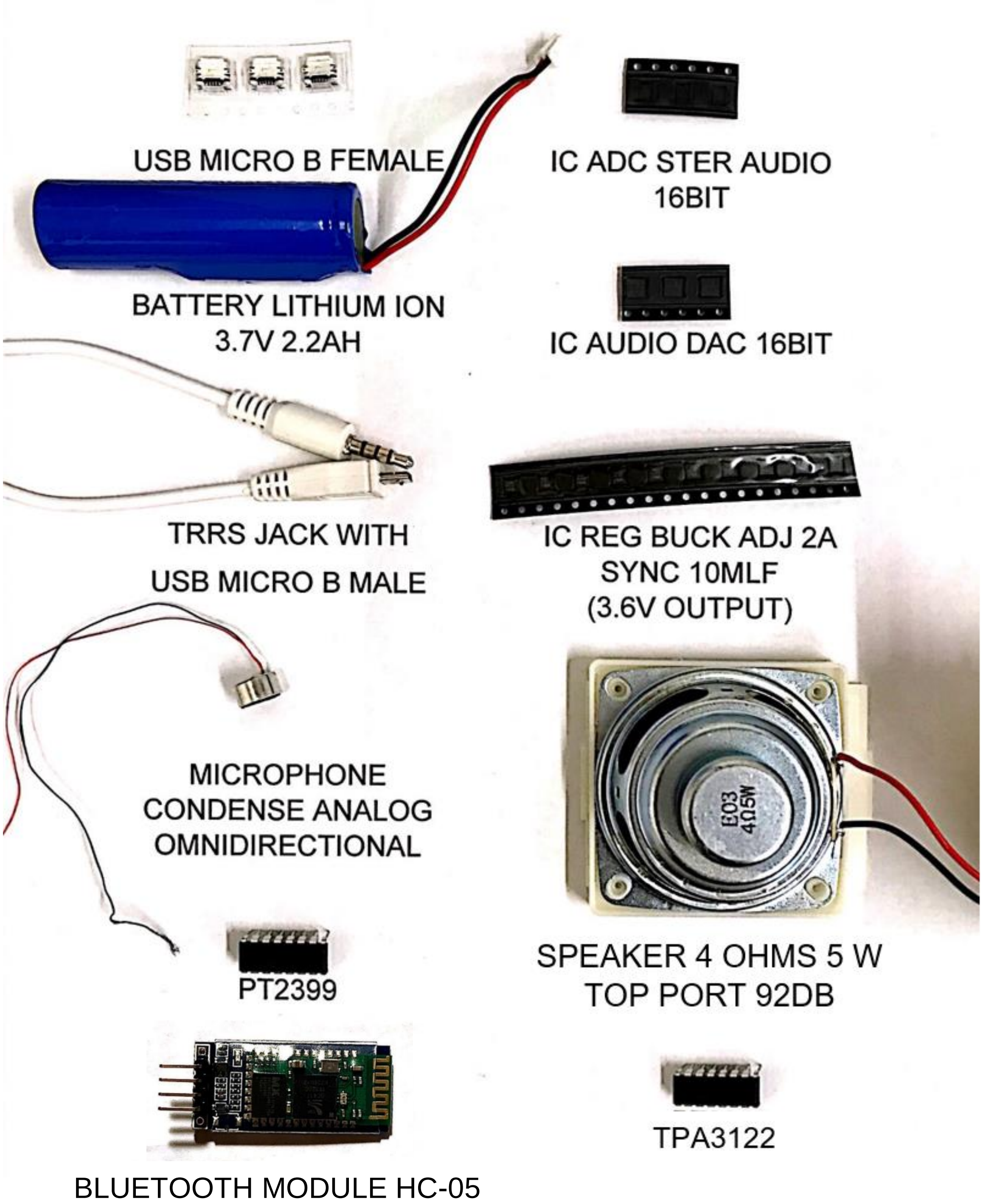


Figure 18. Major Components

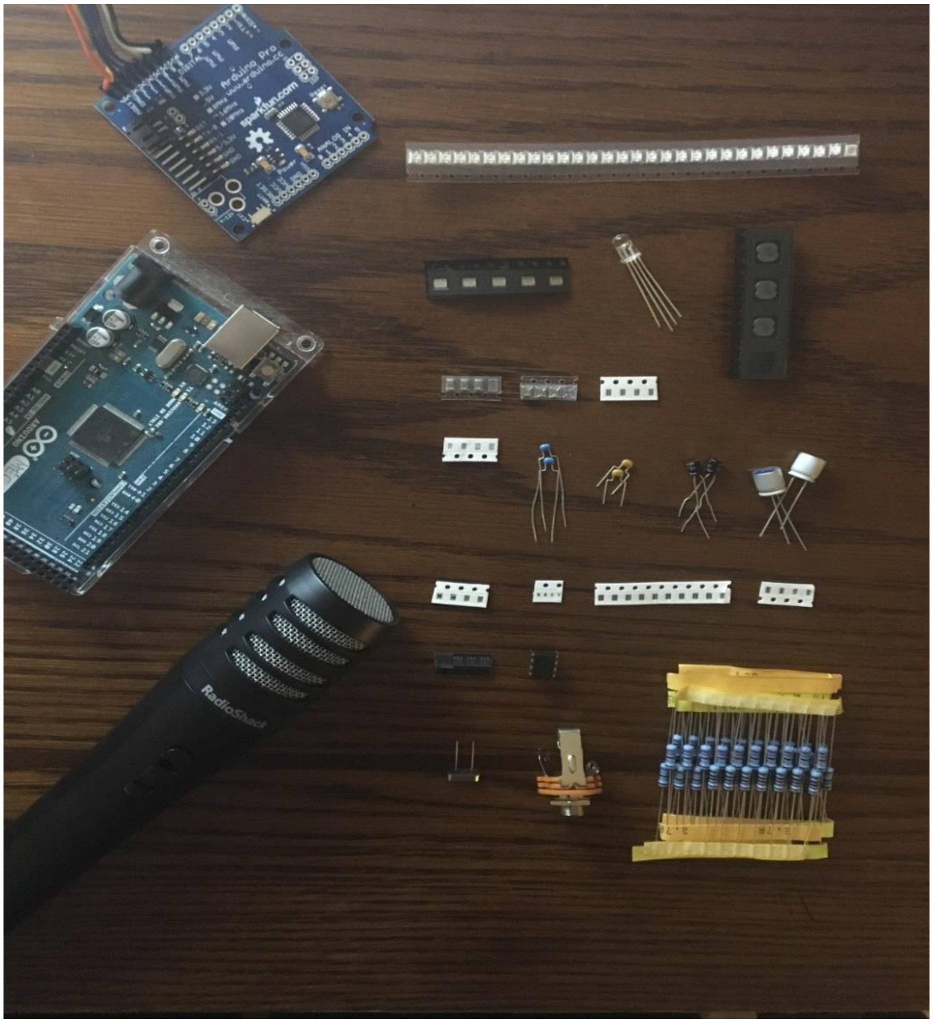


Figure 19: Partial Components for VTVD

The parts shown are basic to the design and capture an idea of where prototyping has led for design. The top left, is a breakout board that was initially thought of to use, and had been configured to do basic functional tests, such as LED RGB output. However, using the ATMega2560 deemed more plausible and using a functional Arduino for prototyping helped speed up modular testing with components, while keeping simplicity in using the microcontroller. From the top, moving downward, the LEDs are shown, both surface mount components and LEDs used for prototyping. Both LEDs have been purchased from Kingbright, and will continue to be used for the final design. Beside the LEDs are a row of SMD Inductors to be used with the Buck Converter. Several capacitors and resistors are shown in both through hole and surface mount packages. The Buck Converter, Op Amp, Crystal Oscillator, input jack and microphone are all displayed at the bottom of the figure. The only item missing is the LED drivers which were purchased recently.

4. Standards and Constraints

In this section, our group will discuss several standards and constraints. All the standards we discuss that have strictly meet the set of regulations that our government agencies want for consumers. Before that our group successfully produce our Karaoke Portable System (KPS) to consumer electronic product, we should understand some of the standards and constraints that we need. Some of these standards primarily relate to the well-being of the consumers or the government agencies, so we should aware of that to make our product successfully. We will also discuss the variety of constraints and challenges that our team expects to face or faced some of them already in our time in senior I and II courses. These will be our personal constraint and challenges that we should face now and in the future.

4.1. Realistic Constraints

Team seventeen is a group of three students including: two electrical engineers and one computer engineer. We all are taking some classes this semester of our last final years of engineering school. With our limit time between working and schooling, we have estimated that our consuming time to work on our project smartly and hardly, we would do our work mostly on weekends and at least one day in weekday. Also, because of the inconvenience work places for all of us, we consider meeting at either member’s houses that closes to all of us and mainly in school if possible. Also, with our busy schedule for the final year in school, we would have to divide some work that the members can work at home if they live far or do not have a car to meet us, so they can work at home and facetime with us through the internet. Again, our group also consists of two electrical engineers and one computer engineer, so all of software programs and skills that is needed for our project are push to the computer engineer, and so she is getting more stressed and hard working.

The rest of our members could be considered skilled programmers in embedded courses and C programing, but we do not have the coursework and skills to successfully fully assist our only computer engineer in this project. Also, in our hardware, we consider designing our PCB board by using digital circuits, but the two electrical engineers do not have enough knowledges and experiences to design the digital amplifiers in our project, so we would go ask our professors around the school for the helps. Furthermore, with the limit time we only have for making a successful project. Our group must meet the requirements and the deadlines to successfully implement our design at the end of the two-semester course. Our group chose to begin the senior design course in the fall semester, so we have more time to do, but we say that “we will work hard and smart to meet the deadlines and requirements on time. To release the amount of stress that the time limit problems may cause our group members more overwhelming, our group has created a group sites that we can put all the works on and the work schedule for each member.

Also, we encourage each member to write a schedule and desire a time to finish, and closely focus on the schedule. The tasks for each member in our group is very strictly whether it is research, asking people for helps or writing pages for our paper, or working with the components of our project. We are making sure that will follow the deadlines strictly. If one of our member ask the other members for helps, he and she can ask, but she or he need to explain how and understand the problems, so we can sit together and work together. After two months working on our project, we feel very strongly and confident that this is a team work, and we can finish it successfully. At right now, we all start to work on components and designing the PCB board, so we can begin prototyping our design as soon as possible, or at least have a very good idea of what we need to improve. There are several websites that we are looking at such as Digi Key or Texas Instruments and other similar website that helps us to but a good component for our project.

4.2. Financial Constraints

The design cost of the KPS is $300. To meet this amount extensive research will be done before ordering any components. If the correct or suitable components are not chosen correctly this could lead to fatal design flaws. Since there will be trail runs while designing the microphone more than one PCB board will be ordered to ensure good quality. The potentiometers, wireless device, LED lights and other expenses will also be taken into careful consideration for budgeting the KPS project.

Time will be leading factor in producing a good quality microphone. Milestone deadlines must be kept and checked through the course of the semester by each member of the group. There are 25 weeks to design the device, so to meet each goal all members of the group will work diligently and keep a timely schedule as planned.

4.3. Safety Standards

Safety has been the most concern for any electronic device. Therefore, there will be some safety standard that we should research to encompass our electronic product. The system should be tested strictly in unwanted levels of emissions and hazardous material. The portable karaoke microphone should be safe as much as possible because it will be directly used by the consumer.

4.3.1. Consumption

Most of the electronic product has the warning labels. They are usually on the box. Their job is to warn the user should consume or use the product in proper way; otherwise, there will be adverse effects that may have on the human body. The warning label could prevent some serious problem can be caused by the product and protect the marketing company in question. Mostly, the warning labels are primarily aimed for young children to prevent the adverse effect of eating components that might happen.

For instant, in the case of chewing the USB cables that our product uses for wiring, there are hazardous effects that may occur. For one, the consumer might have potential electric shock if the protective rubber covering is pierced and direct contact with the wiring is made. Full consumption of the wiring may cause death through suffocation, as the wiring may become trapped inside the trachea. Further, the material in question is very questionably dissolvable within the standard human gastrointestinal tract.

4.3.2. Electromagnetic Field Emissions

Over the past century, with the increase in electronic devices in our society, exposure to human-made electromagnetic fields (EMF) has substantially increased. It has become the focal point of many studies and debates into the potentially hazardous effects they may have. The World Health Organization has established a division to assess the whether the EMF of varying frequencies have any adverse effects or potentially good effects on the general populace. To conduct this study, a range of 0 to 350 GHz were chosen to as this is a typical range for radio frequency emitting devices.

Our team has chosen to follow what the FDA designates if our project is safe. We found out that there is not a concrete answer on how dangerous electromagnetic field radiation from every-day consumer electronics. The FDA organization has been a long partnership with the World Health Organization. They has conducted substantial research of EMF effect through a program called the International Electromagnetic Field Project. It is designed to review all scientific claims and documentation produced concerning the effects that electromagnetic fields can have on an organism. Through this research, the International Electromagnetic Field Project hopes to adamantly have a final and conclusive answer to the aged debate on whether electromagnetic fields are indeed as harmful as some claim them to be. According to the project, there is no substantial evidence that prolonged exposure to radiofrequency emissions can lead to serious, or terminal, health complications. The administration cannot rule out the possibility of risk, but claim that they are quite adamant on the fact that if such a risk existed, it would be minimal. [43]

The International Electromagnetic Field Project primarily uses information provided by organizations such as the international commission on non-ionizing radiation protection and the institute of electrical and electronics engineers (IEEE). Through this substantial research, several guidelines have been created for consumer electronics radio frequency emissions that are also followed by the EPA, OSHA, FDA, and NIOSH. Each of these organizations work at different levels of government to ensure that the general populace remains at safe exposure levels [44].

4.4. Testing Constraints

Our group is limited to using the University of Central Florida’s electrical engineering laboratories and our own limited equipment. In senior deign lab, there are limit in resistors and capacitors available. There are also no breadboard or probes for oscilloscope, function generator, and power supply machine. In order to test, we have to rent from a professor which his office hour is limited. We plan to make use of several oscilloscopes and specification documents to accurately test each individual component for defects and tolerances. Therefore, it will prevent us to any issue when making the project.

In testing our design, we must ensure that we produce very accurate data to potentially narrow down what is causing our design failures. Most of our testing will be performed at room ENG1-257, open lab for electrical engineering student. It equipped enough components and machines provided by the University of Central Florida. Our group plans to keep several spreadsheets where we consider changing the component to fit the project better.

Resistors used in our project must be adequately tested for their tolerance and resistance to ensure accurate data when prototyping our design. To accomplish this, our group plans to use the very accurate oscilloscopes and multi-meter found in room ENG1-257 laboratories to test the resistors used in our initial design process.

To test the analog, digital chip or microphones, speakers, ADC, DAC, we have to build a particular circuit to see if the components are working. For instant, echo digital chip will be built with other resistors and capacitors. Oscilloscope will read the input waveform as well as the output waveform. The delay of output can be determined by the phase of two waveforms. In addition, there will be a potentiometer to adjust the amount of delay that the output could be to the input signal.

4.5. IEEE Standards Association (IEEE-SA) Copyright Policy

In our project, there is one thing that is very important is the copyright. Our group rather take a lot of time to research and make our own work than copy or use someone’s work. In case, we have to the work or the reference from the work of someone else, we should ask them first and cited the link that we use. In IEEE standards Association, there are some policies about the copyright that we should aware of. First, enhancing the accessibility, distribution and use of information is a major objective of IEEE and the IEEE-SA, limited only by the requirements of viability and professional propriety [63]. Second, in exercising its rights under copyright, the IEEE-SA recognizes that it is acting in part to serve and protect the interests of its standards participants and participating entities [63]. Third, fees for the reuse of IEEE standards material are appropriate for contributing to the cost of standards development oversight and original publication, especially where the reuse involves a license to copy, or allows resale, or is of a magnitude that would tend to reduce subscription or another sales income [63].

4.6. IEEE standards

The Institute of Electrical and Electronics Engineers Standards Association (IEEE-SA) is an organization within IEEE that develops global standards in a broad range of industries, including: power and energy, biomedical and health care, information technology and robotics, telecommunication and home automation, transportation, nanotechnology, information assurance [63]. In this section, as an electrical, compute, sound, and electronic engineering, following IEEE standards is very important. We must follow the rules and the standards of IEEE organization before the products are made and produced. The table below are the titles of IEEE standards and descriptions of some IEEE standards that we should aware of when performing this project:

Table 18: IEEE Standard Code

Standard Titles	Description
IEEE 1-1969	IEEE General Principles for Temperature Limits in the Rating of Electric Equipment
IEEE 15.S1-1961	IRE Standards on Radio Transmitters: Definitions of Terms, 1961
IEEE 1888.3-2013	Cyber-security
IEEE Std 1686-2007	IEEE Standards for Intelligent Electronic Devices

4.7. Federal Communications Commission Regulations on Digital Devices

The Federal Communications Commission (FCC) is an independent government agency responsible for regulating the radio, television, satellite, cable, wire and phone industries. The FCC regulates all interstate communications, such as wire, satellite and cable, and international communications originating or terminating in the United States [64]. Leading the FCC are five commissioners appointed by the President and confirmed by the Senate for five-year terms [64]. The President designates one of the commissioners to serve as chairperson. Only three commissioners may be members of the same political party, and none of them can have a financial interest in any commission-related business [64]. The FCC works towards six goals in the areas of broadband, competition, the spectrum, the media, public safety and homeland security, and modernizing itself [65].

Today, with the development of the scientific science and electronic digital devices, the most of the life and change the change of all the face of life. The more modern life improves, the more the presence of electrical and electronic equipment. These devices are available everywhere and serve all human interests, from living to production. Therefore, the electrical engineering and electronics industry is always an important field in the field of engineering. Electrical and electronic engineering is a discipline that studies and applies electrical, electronic and electromagnetic issues to a wide range of disciplines such as energy, electronics, control systems, signal processing, telecommunication. For example, at home, we can find digital technology in coffee pots, microwaves, toasters, televisions, computer, kitchen cooking tools, and even your refrigerator door and your cars.

The developing modern technology help human save a lot of time and do the work more accurate. While digital technology provides a lot of usefulness to our society, it can also have several negative effects on several important components of our lives. For example, when making a stereo music speaker, it must be follow the FCC rules, if so the product is not recognized in our society because the product is not met FCC requirements. Therefore, the product will be useless and would make some negative effects on society. As of the result, to make sure for making a good and safe product for our society, the producers must strictly follow the important of FCC regulations for the overall design. Before making a product, the producer must test and submit to FCC, and let them know the product meets the requirements. As a result, the producers can be happy to manufacture the products to our society. In our project, we will make a Karaoke Portable System (KPS) that can serve the human needed. The output of the signal is from 20 Hz to 20 KHz. This range of frequency is meet the FCC regulations. Furthermore, all our components such as digital chip, analog preamplifier IC, ADC, DAC that we are using are fit to FCC criteria of digital devices.

4.8. Programming Standards

Programming will be an essential role in the completion of the system. The use of coding conventions is important when a project involves more than one programmer. To ensure the system runs as accurately designed, both hardware and software design must go hand and hand. The key points include the life cycle, maintain a structure and plan to control the cycle of the software portion of the KPS system. Making sure the goal of what the software is meant to do is met. Have a detailed design of individual components that will need programming such as the microcontrollers, ADC and DAC. Below are a few guidelines or practices and prerequisites that must be met to ensure and guarantee a good running programmed system.

4.8.1. Commenting

It is good practice to comment throughout a program by using comment blocks. Commenting is important for any software project to ensure all programmers can understand what has been coded. If complicated logic is being used, it is good practice to leave a comment block near that part so that another programmer knows what is going on. Suggested below is a brief commenting description to have as good practice when starting a coding process.

1. Name of the module
2. Purpose of the module
3. Brief Description
4. Programming Author

4.8.2. Naming Conventions

When dealing with a complicated or simple system, naming variables throughout a program in a descriptive manner or using proper names helps each programmer working on the system to understand and follow code easier. The naming convention should be consistent to prevent wasting time trying to decipher code. Use of proper naming conventions is also considered good practice.

4.8.3. Routines

Having a good naming convention is one of the most influential aids to understand the logic flow of an application. Recommended naming techniques for routines are discussed below.

1. Avoid elusive names that are open to interpretation such as Analyze() or X1 as a variable.

2. Use verb-noun method for naming routines, functions or methods or objects such as CalculateInvoiceTotal()

4.8.4. Variables

Variables are used through any programming application and therefore to ensure understanding of the application and system a few guidelines include appending computation qualifiers to the end of a variable name where appropriate. [60] Constants should be uppercase with underscores between words.

4.8.5. Portability

“Hard coded” should be kept at a minimum if necessary or not included at all. Such as absolute file paths, file names, user names, and addresses or shared variables should be avoided otherwise the application will not run as the design anticipated. A programmer should parameterize such variables for an outside hosting environment. This way the code can be portable and updated as needed and moved and used in any system.

4.8.6. Formatting

Formatting makes the logical organization of the code easy to read. Establish a standard size for an indent and use it consistently. Align sections of code using indentation.

Taking the time to ensure source code is formatted is helpful to everyone working on the application. Establish a maximum line length for comments to avoid having to scroll to read the comment. Use good white space to aid the reader in comprehending the logical segmenting of the application. Break large complex sections of code to smaller comprehensible portions.

The overall formatting of the code for the system ensures all members or outside readers looking at the code can follow thoroughly without any complications. As mentioned above, line length, good white space, standard size fonts and alignment will be keep the code formatted well. The overall programming design must be clean and readable, if any justifications must be made the code will be well organized in order to make the needed justifications to the code.

5. Project Hardware and Software Design Details

In this section, we will have possible design details of hardware and software. It contains the schematic circuits from Eagle of power supply distribution, amplifiers, and LED flashing circuit. There are some software procedures to design the flashing function of the KPS. Finally, we will test all the chips and components on the breadboard to make sure everything we purchased are functional correctly.

5.1. PCB Design

Printed Circuit Boards as known as PCB is used to simplify the complex electronic circuit by removing the wires and creating cleaner and less prone for error circuit. The copper wires used to connect the components in the circuits now are replaced with the copper tracks that connect components together. The PCBs are usually small and compact. Therefore, Surface-mount components are often referred to as surface-mount devices (SMDs) and are replacing through-hole components in most commercial products. This is so because SMD components are smaller and cheaper than their through-hole counterparts, and the boards that use them are also easier to make. For this specific project, surface mounted components were chosen to limit the number of assembly steps. These components will be mounted on the surface of the PCB by the manufacture. In addition, here can be up to 4 layers that can be created to have their own traces. This is important because every trace should not overlap with each other. [71].

5.1.1. PCB design principles

There are several PCB's principles from David Marrakchi and Ian Poole. By following those principles, we could design a compact and successful PCBs. The following design principles could be used as a checklist for any PCB designs: Preparation, component replacement, traces, separation of Analog and Digital signals, heat, and verifying the design.

5.1.2. Preparation

The first important step is to create a schematic of the circuit and to test the schematic on a bread board. Once the test is successful, the schematic now is created the layout of the PCB on Eagle software [72]. Hence, the ports should be placed on the desired position and how the consumer will be affect by the orientation of the ports. An estimate of the size of the PCB should be specified and each component should be able to fit on the PCB. [73] How the board will be mounted on the final product should be considered. [73] For this project, there will be three square PCBs boards that can link together with one voltage source to create a rectangular shape box.

5.1.3. Component Placement

The components should be placed in such a way as to make the routing of the traces short and not overlap [74]. The components should be laid out in a way to minimize the PCB's size. Smaller components should be oriented the same direction to avoid soldering errors. [74] All surface mounted and through hole components should be placed on the same side of the board to limit the assembly steps [50]. For this project, surface mounted devices (SMDs) will be use since there are three separated PCBs boards, which will take a lot of spaces. Additionally, the larger components should be placed first in the direction of the trace followed by smaller components. Usually, the capacitors with higher value do not have in SMD form, so there is possibility to make holes on PCB board to fit those capacitors. [74]

5.1.4. Traces

To avoid bending the board and misaligning the components it is important to use a common rail for the power supply and making the common rail wide. [74] All traces should be designed to be as short as possible which could avoid overlap. The components should be placed as close to the processor as possible. The net widths or width of the traces is proportional to the current in each component. [74] It is recommended to use 0.010'' width for low currents and wider nets for currents greater than 0.3 A. [73]

5.1.5. Separation of Analog and Digital signals

To reduce distortion and pollution of the signals there should be a separation between the analog and digital signals. [74] The PCB should have a digital side where all digital components will go and likewise for the analog signals. This is due to the differences in voltage between the two types of signals. [74] An important consideration is to avoid running tracks in parallel to avoid crosstalk with the signals. [74] When the tracks must be cross they should cross at right angles to each other. [73]

5.1.6. Heat

Each component produce heat which will cause deterioration in the traces, reduce the life span of the PCB, and create a dangerous device for the user. Thermal reliefs should always be used for any hole or via that is connect to ground or power plane. [74] Teardrops can also be implemented where traces join pads to provide support for the materials used in manufacturing. [74] Additionally, heat sink such as thermal resistors can be used for components that produce large heat dissipation. [74] Finally, ventilation such as fans or holes for the case can be employed for the PCB temperature to be lowered. Separating the components out can also help with heating issues. [73]

5.1.7. Verifying the design

Due to the cost of producing a PCB it is important to verify that the design will work. An electric rule check (ERC) and a design rules check (DRC) should be done. [74] The ERC will conduct an overall check for each component and notify the user of any major flaws such as unused pins or unassigned values to components. [74] The DRC is user made and checks the design for specific project specification such as, trace width. [74] The next thing to check is the routing of each signal and that the PCB matches the schematic. Additionally, the measurement of each component should be verified as well as the width of each net and trace.

5.2. Power System Design

The following section contains the methodology behind the power system chosen. This section contains reasons between choosing certain specifications for the design, testing procedures as well as the design itself and schematic of the power system of the device.

The objective of the device is to produce a long lasting small power system that is powered by a battery. Since the project is portable karaoke system, the primary focus for the power system is both power efficient and rechargeable. Overall what should be produced is a long lasting rechargeable battery that can power the device for long periods of time. The purpose of this section is to go over how the power system design will be implemented, it will contain both schematic references as well as a PCB layout of the proposed system, along with the testing implementation.

This implementation should contain two sections. One for the powering of the device, this can be achieved by using a switch that transmits power throughout the device and possibly turns on a LED light, so the user knows the device is on. The other is the recharging station that the battery will be undergoing along with what it will intake and what it should produce. This power system should be able to produce a minimum of 3.7 V to fully power all the power sources at a stable rate, however all of the devices can still run under lower voltage sources, but it would be beneficial if the power source produces that specific voltage source.

5.2.1. Voltage Regulator

A Voltage regulator will be used in order to regulate the voltage source to component input voltage. What a voltage regulator does is intake a voltage source produce a constant voltage source. This source is typically smaller which will be more efficient; however, there are possible to go higher with less efficiency. There are two types voltage regulators types are switching and linear. Switching regulators use a switch to power the voltage source, by taking the average of the voltage resulting in a constant voltage. The linear regulator on the other is just a device that compares itself with the reference voltage and adjusts. The advantages and disadvantages of one another affect the component design. These differences vary for uses, such as the linear which can only produce step down voltage, meaning that the voltage regulator will lower the input voltage.

5.2.2. Power Supply Distribution

The battery will be the primary component of the power system. In order to know what type of battery to use it is imperative that we take into account the majority of components that will be powered by the system. Since we have already considered powering options in the previous sections and have decided that lithium ion batteries seem to be the common choice, when dealing with long lasting batteries that meet today’s standards. However, what was not covered was what the battery size capacity should be such as the mAh needed to properly power the device for hours at a time. In addition, what the battery voltage should be minimum to power all of the devices properly, since the group wants to produce a fast acting and efficient device.

Table 19. Power Consumption of Device

Qty.	Devices	Working Current (mA)	Total Consumption (mA)
2	Pre-Amplifier (LM386)	4	8
2	Amplifier (LM386)	4	8
1	Tone Control (TDA1524)	50	50
1	Speakers 4 Ohms 5W	312	624
1	Speakers 4 Ohms 10W	625	625
1	Echo (PT2399)	100	100
1	Bluetooth Module	45	45
TOTAL			1460
	Battery = 9800 mAh	**≈ 7 hours**	

Theoretically this is a rough estimate expecting that the processor is at 100 MHz and that the mA/MHz is at its worst condition which is .21. However now we can calculate what the battery size should be in its expected case in order to meet the 10 hours condition a 2200 mAh lithium ion battery with a voltage of 3.7V would suffice. This would provide at least 8 hours and even more however, this will be implemented and tested in the future.

This is a rough estimate assuming that the power is 100% efficient. Based on under the assumptions that the product is at least 70% efficient we can assume that the battery will power itself for 6 hours which is a good starting point, however what would be better is if the device had a power efficiency that was well implemented which the group will be trying to achieve.

In order to properly test the battery there much be some precautions to take note of. First the battery should not all times the power is turned on we are going to assume that the battery fits current industry standards in terms of battery life. Due to not knowing how exactly efficient the product is there will be some testing with battery, however in this case we are going to assume that the battery and headset is something that the group did properly.

5.2.3. Powering the battery

When recharging a device, it is important to know what type of power consumption you are working with. In this case there needs to be a power source for the recharging station. This source will be coming from a USB adapter, so that it can be powered by either an outlet or another device. This will require a female USB port. In addition to that there will be LED lights to determine the battery charging stage, at least 3 LED so that the user can distinguish between fully charged and minimally charged and the user can have an estimated life expectancy of the device. The main two ports that will be used from the USB power source are the VCC ports which the voltage is being sent to the battery.

The first to receive the voltage from the USB is a voltage regulator which will adjust the voltage to produce a safe voltage that can power the battery. Things to watch out for when implementing the voltage regulator are the load current which will be implemented by the battery, which can cause a shift in the Q Point. However, since the project will mainly be focusing on a DC to DC regulator it should be easily implementable since there are no sinusoidal waves to worry about. In addition to that using a switching regulator to read the voltage battery can be used to implement led lights. When certain threshold voltages are met lights could turn on and produce an indicator for the user to know how charged or uncharged the battery is. This could be implemented into the circuit.

Most importantly the circuit built should properly know when to stop charging the battery. This process can be done through the implementation of diodes, which can be activated. When the battery voltage is larger than a certain threshold, the diode should turn on and prevent any more current to flow through the battery. It is imperative the circuit is properly working and will be the first concern of the power system as lithium ion batteries are very dangerous due to their common occurrences of batteries blowing up and catching on fire due to the charge build up.

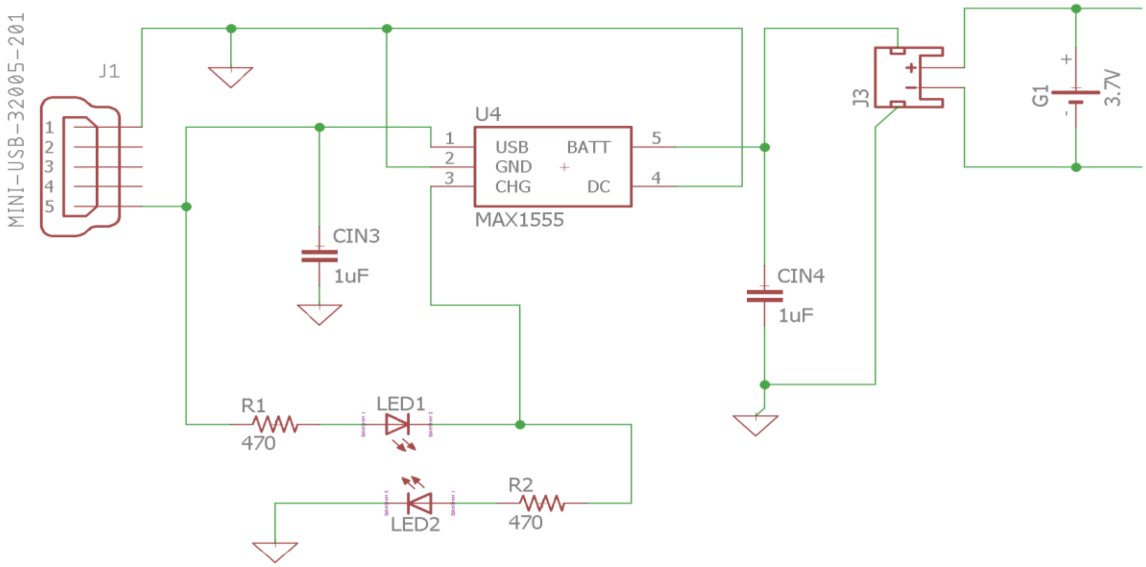


Figure 20. Battery Charging Schematic

This schematic used the MAX1555 as a charging unit. The only thing the component emphasizes on is a steady input current and feedback prevention of circuit. A simple explanation of this is as follows. This circuit was created using eagle and used a schematic reference for the MAX1555 had, however that schematic did not use the implementation of led lights to display to the user the charging state. The 5-volt VCC goes from the mini-USB to the USB pin of the IC. The 5V is also connected to a resistor which pulls down some of the voltage to turn on an LED light. This led light is used as a display to show that the battery is being charged. The LED is connected to the CHG pin which has a high impedance when the battery is fully charged. Once the battery is fully charged the high impedance will turn on and the voltage will continue on to the LED2 and resistor, this would be a visual representation that the battery is fully charged. On the battery pin is a simple jumper connection that connects from the battery pin to the actual lithium ion battery cell. This circuit needs to take into account over charging which can be done with a simple switch. Once the battery hits a maximum of 4.2. Voltage it will stop charging and should cool down similar to other mobile devices.

5.2.4. Powering the Device

The objective of powering the device is to produce a steady and stable power source that can power all of the components with their respected power source requirement. The second portion of the power system is the device itself. Powering all the components requires different voltage sources for each one. However most of them lie within the 3.0-5 voltage range, this can be used to simplify the circuit by using another voltage regulator to produce a constant voltage source. In addition to this implementation for the system to not be using power at all times and not wasting battery a switch will be implemented to connect the battery towards the device. In addition to that bypass capacitors would have to be added from the VCC line which is currently the voltage regulator output to the components that had been selected by the group to reduce the noise caused by the regulator.

In this case a regulator that would be ideal would be a linear regulator due to its low noise output compared to the switching one as stated earlier. However due to the nature of the battery size which is currently 3.7 and regulator would be too much which it might be better to use an alternative route such as voltage division path or even a reference voltage by attaching a diode towards the device, which could then be used to power the LED light indicating the circuit is on.

Finally, the switch that is to implement the powering of the device in no way will switch from charging more to power on mode. Instead it will turn on power mode, the device should still be charging if connected to the power supply. This means that if the user is working in an office or at home, they can connect themselves to a power source such as an outlet.

There are four majority voltage limits for all components: 3V, 3.3V, 3.7 V, and 5V. The schematic circuit below This circuit contains two voltage regulators from the 3.7V input. This is just in case the microphone input requires a different voltage from the actual 3.3 volts required from the other main components. Should there be no need the bottom half of the circuit can be taken out. I use the software called Webench from Texas Instrument website to design the power supply distribution system. There are 3 Integrated Circuits chips that will be used: TPS61240, LP5900, and LD1117.

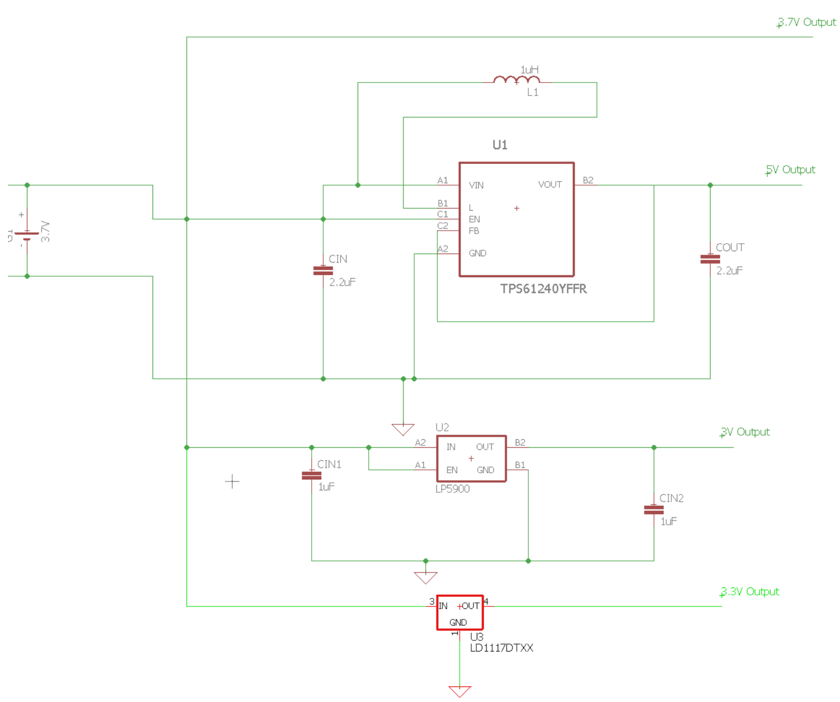


Figure 21. Power Distribution System Schematic

5.3. DAC Design

A digital to analog converter (DAC) is an electronic device that converts digital audio information that comprise bit o's and bit 1's into an analog audio signal that connect to the speakers. In our project, the input of DAC is connected to a Bluetooth module device. The input is a digital signal from the Bluetooth device, and the output is an analog signal that comes from DAC. PCM1774 is a DAC that we choose since it has low voltage supply that mostly below 5V. The PCM1774 is a low-power stereo DAC designed for portable digital audio applications. The device integrates headphone amplifier, line amplifier, line input, boost amplifier, programmable gain control, analog mixing, and sound effects [75]. It has 16 pins and use a single voltage supply from 1.71 V to 3.6V.

It is available in a small-footprint, 4-mm × 4-mm QFN package [75]. The PCM1774 devices accept the audio data formats with using the range of input from 16- to 24-bit data. It uses left-justified, I^2S communication (MODE high) and SPI (MODE low). The benefit of using I2S for the system connection allows for the audio to be able to support stereo audio. The Sampling rates of this DAC is up to 50 kHz. It has a full set of user-programmable functions is accessible through a 3-wire serial control port, which supports register write functions [75]. By researching at the datasheet of PCM1774 from Texas Instrument website, the connection diagram is shown below:

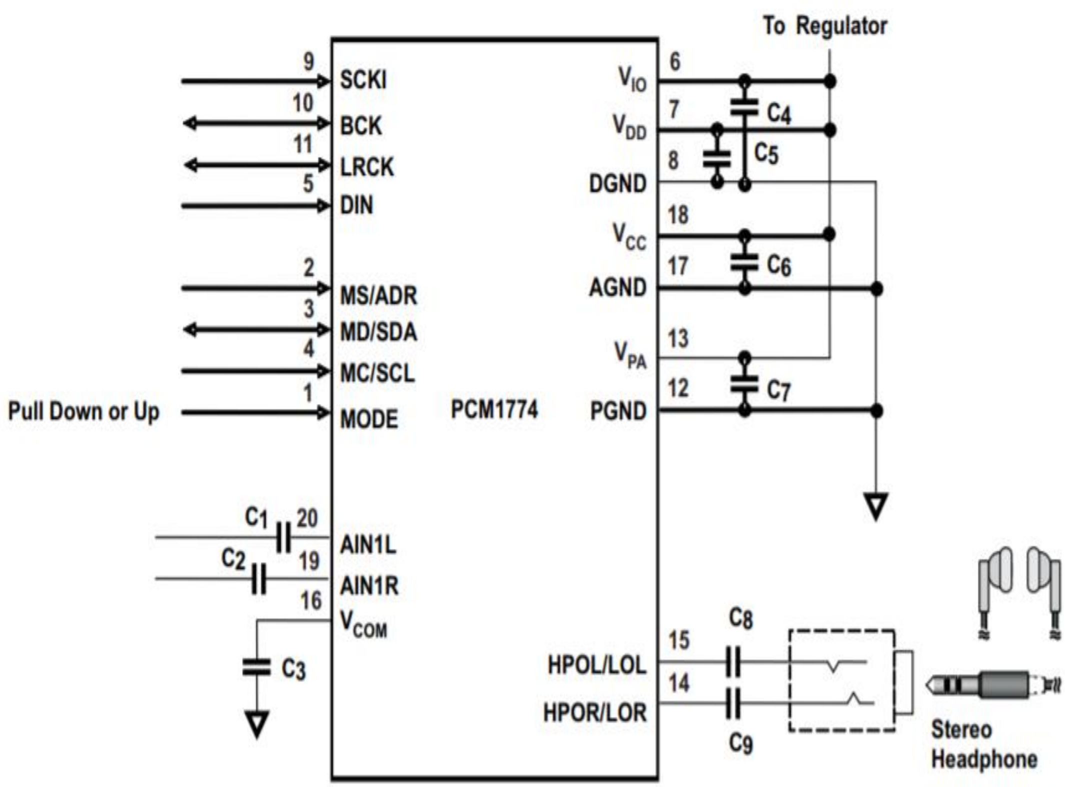


Figure 22: Connection Diagram (Permission Requested)

In this diagram, PCM1774 connects to the power supply and its components. In pin 1 is MODE that can be put up (I2C) and down (SPI). Pin 2 is mode control select three or two wire interfaces. Pin 3 is mode control data for 2 or 3 wires. Pin 4 is mode control for clock. Pin 5 is input of data audio. Pin 10 is serial bit clock, and pin 9 is system clock. Pin 11 is left and right channel clock. Pin 20 and 19 are analog input for left and right channels. Pin 15 and 14 are headphone/lineout for right and left channels. Pin 12 and 13 are ground for speaker power amplifier and power supply for power amplifier. Pin 6, 7, and 8 are power supply for digital I/O, digital core, and digital ground. Pin 17 is the ground for analog, and pin 18 is analog power supply.

5.3.1. Schematic of DAC

The schematic of this DAC is designed base on the connection diagram found in the data sheet. The capacitor values were used based on the connection diagram. The capacitor values are make sure to choose. The unconnected pins are left to connection to the output of Bluetooth device, microcontroller, and input of tone control chip. The design of this DAC is to test the connection between the DAC and Bluetooth module device, and the output of DAC is to connect to input of tone control chip to adjust the effects of bass, treble, and bass. Eagle-Cad is used to design this schematic, and the schematic design is below:

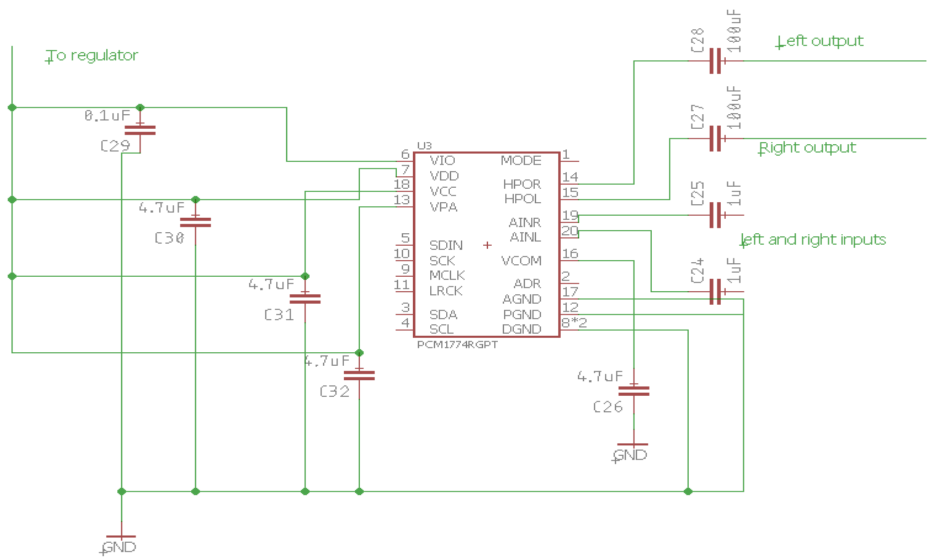


Figure 23: DAC Schematic

This is only the first version of this schematic and most likely will not work when implemented in the laboratory. The problems that could occur when testing are capacitor values and the connection for the analog outputs, voltage that using the same ground for the digital and analog causes any serious issues. An additionally another main area may occur are the muting pins. These main problems could block the analog output properly.

5.4. Volume Design

Speaker are devices that translate signals from an electronic device, such as a receiver or CD player or sound waves. Sometimes, the listeners want to get a sound louder or smaller, then speakers may need a bit of a power boost to produce louder and cut the power to get sound lower. Running an amplifier between the audio device and the speakers will raise the wattage going into the speaker. This amplifier is called volume amplifier. The connection of the speaker and preamplifier is important, and without one of them, the listeners cannot hear any sound. The best combination of these relationship is to get a pre-amplifier that has twice wattage of the speaker [4]. For example, speaker can produce 50 watts, so the amplifier should be 100 watts. There are two types of volume amplifier. They are analog and digital volume amplifiers. Finally, LM386 chip is decided to choose as a volume amplifier from Texas Instrument (TI). LM386 is a low voltage audio amplifier and frequently used in battery powered music devices like radios, guitars, toys etc. The gain range is 20 to 200, gain is internally set to 20 (without using external component) but can be increased to 200 by using resistor and capacitor between PIN 1 and 8, or just with a capacitor. Voltage gain simply means that Voltage out is 200 times the Voltage IN. LM386 has a wide supply voltage range 412v. In a special case, the voltage supply can be between 5V-18V if using another LM386 package. There are five packages for LM386 audio amplifiers. To meet the battery requirement, the package (LM386N-4) is decided to choose in this project. In this package, the power output is enhancing a lot of better than a normal one. Its power output is between 0.7W-1.3W. The voltage gain (Av) in typical case is 26 dB (Av=20), and in special case when putting a polar capacitor 10uf, the voltage gain is 46 dB (Av=200).

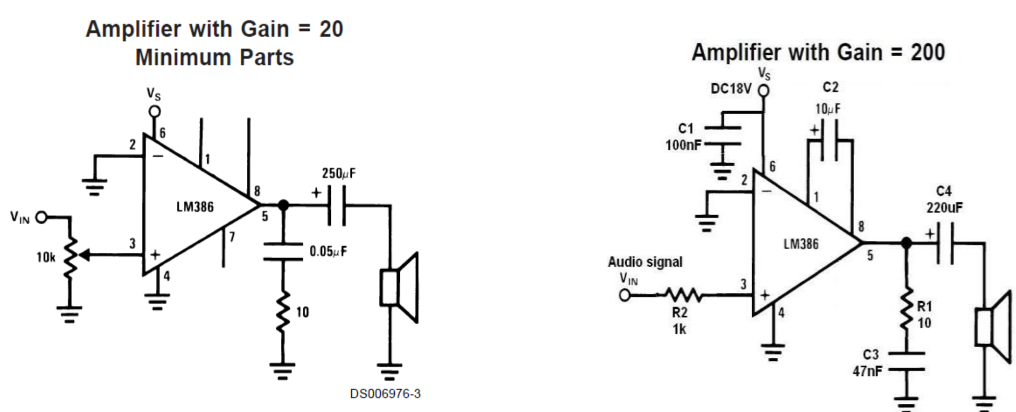


Figure 24: Volume Control Diagram with Voltage Gain 20 and 200

In the circuit above, there are two circuits for volume amplifier with voltage gain 20 and 200, the designs will have two potentiometers that allow us to change the gain or the volume. The circuit with voltage gain 200 is chose with a potentiometer to adjust the gain. The output power of the speakers is depending on the voltage supply since it can work on 5 – 18V voltage supply.

5.4.1. Preamplifier for Microphone and Volume Design

Volume Amplifier requires power supply between 5V-18V for its minimum operated limit. This meets the requirements of the battery 12V. The design of the volume amplifier is complicated. It has two LM386 chips. The first one is pre-amplifier for the microphone. At this one, the voltage gain is only designed with the voltage gain 20. At this stage, the current is so high, that makes the LM386 getting hotter, so putting another LM386 as second stage with the voltage gain 200 is a solution for this problem. At this second stage, using a potentiometer is designed to adjust the current supply. This potentiometer is very sensitive since it controls the voltage gain. To design the whole system for the microphone, the order of the connection of each amplifier is important, so putting echo amplifier and stereo amplifier between these two stages is a solution. For the schematic of the volume amplifier design, there are two LM386 chips. The first one is used for preamplifier for the microphone, and the second one is used to amplifier the output of the whole karaoke system. The IC LM386 is a power amplifier used for amplifying small audio signals with low supply voltages. Though the gain of this IC is set at 20 internally, it can be raised almost 10 times higher - that is up to 200, just by introducing a resistor and a capacitor across its pin 1 and 8. The IC is available with four versions: LM386 N-1, N-2, N-3 which typically show very low distortion characteristics and function well with voltages ranging from 4 to 12 volts DC. The fourth type, the LM386 N-4, is specified with working voltages from 5 to 18 VDC, these being the final safe thresholds beyond which either the devices stop working or become too hot and get damaged. In this design, there will be two extra 12V voltage supply for the echo amplifier and stereo amplifier

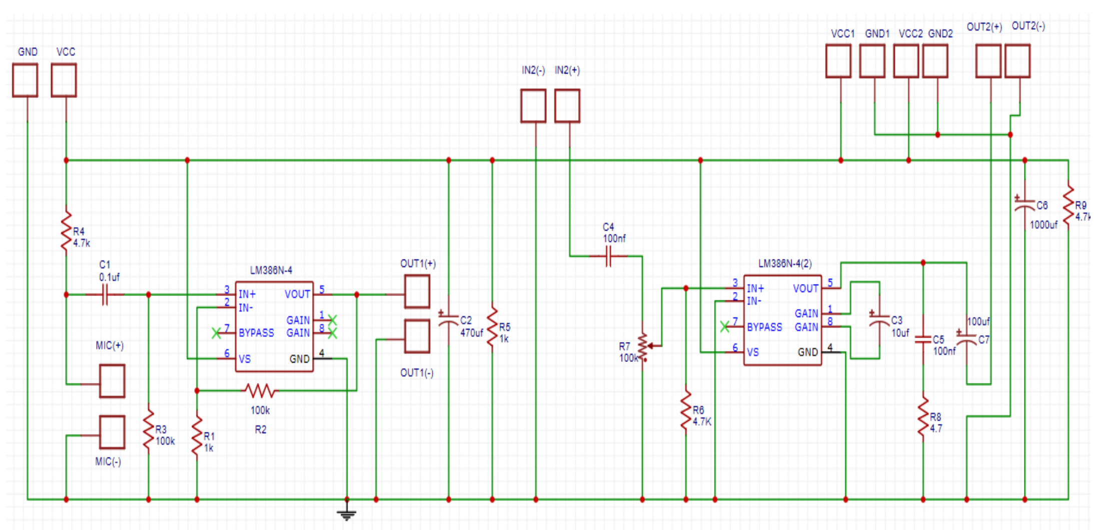


Figure 25 : LM386 Volume Amplifier Circuit Diagram

The design for preamplifier for the microphone and the volume amplifier PCB board was designed very carefully. The dimension was to make sure to fit the KPS case. The dimension is about 2x1 inches. There is a potentiometer to adjust the voltage gain of the volume amplifier. The design can work between the voltage gain 5V-18V. The picture below is the PCB design:

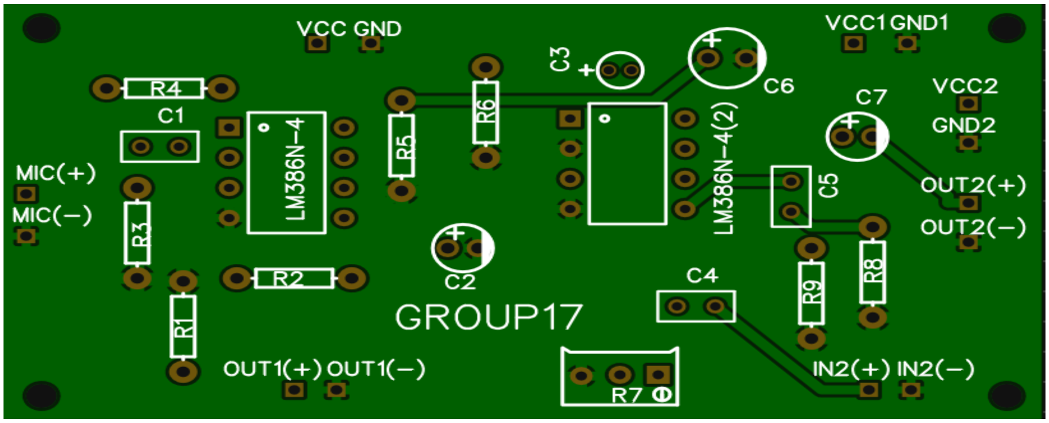


Figure 26: Preamplifier for Microphone and Volume Amplifier PCB Design

5.5. Treble and Base Circuit Design

The music speaker amplifier is designed to take analog input signal (music from DAC that is conned to Bluetooth), and the input signal can adjust by changing its magnitude (volume) and frequency (bass and treble) to make the best music into two speakers in the left and the right. In the process of this amplifier, the music is taken from any smart devices such as iPad, table, phone through Bluetooth. The signal output from Bluetooth is digital signal, so an DAC will convert digital to analog signal. This signal can be adjusted with frequency and magnitude to make music by changing external components such as capacitor and potentiometer. An analog chip is chosen to meet our requirement in voltage supply and output voltage with low noise, and low power consumption.

5.5.1. Schematic Treble and Base Circuit Design

Stereo Amplifier requires 12 V power supply for its minimum operated limit. This is a challenge for the project to find the battery that could meet the requirement. The design of the stereo amplifier is complicated. It has four potentiometers to adjust the magnitude of each functions: Volume, Bass, Treble, and Balance. Moreover, the input signal will decrease tremendously after being filtered by this amplifier. Therefore, the output signal need to gain at least 10 dB by another amplifier to have the sound be hearable. TDA1524A chip is manufactured by PHILIPS. It is designed as an active stereo-tone and volume control, especially for car radios, TV receivers and mains-fed equipment. It includes functions for bass and treble control, volume control with built-in contour (can be switched off) and balance. All these functions can be controlled by DC voltages or by single linear potentiometers. It provides very pleasing performance and makes a useful addition to any of our audio power amplifier kits. RCA jacks is optional for audio inputs. Usually TDA1524A supports for left and right channel; however, there is only one output to the front speaker, one channel will be grounded.

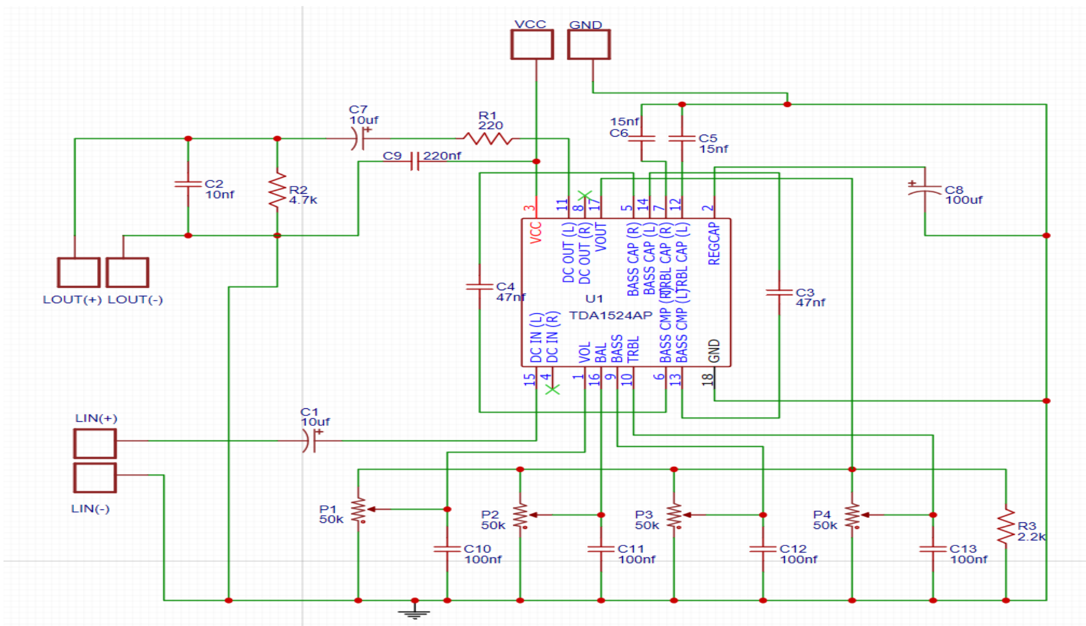


Figure 27 : Tone Control Amplifier Circuit Diagram

For the PCB board design of tone control amplifier, the dimension is about 2.5x1.5 inches. The board is working on 12V from the battery supply. There are four potentiometers to adjust the volume, the bass, the balance, and the treble. The picture below is the PCB design for the tone control amplifier:

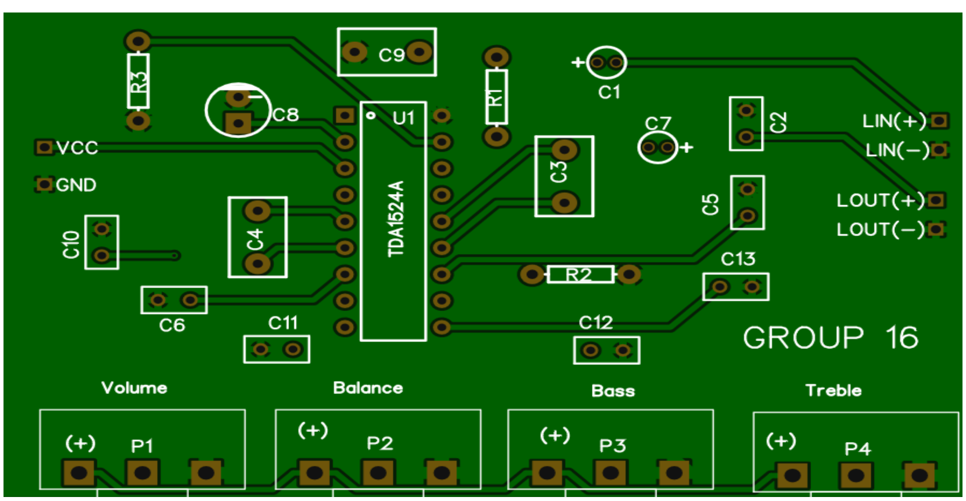


Figure 28: Tone Control Amplifier PCB Board Design

5.6. Echo Circuit Design

An echo is a time effect. An echo can take a direct signal and storage it first for a set of small delay time and process it to be played back later. A delay can repeat a signal once or multiple times. It can be used to separate a vocal from the rest of the mix or for a special effect. When applying a large amount of delay time, it can make a sound off. There are two types of echo or delay circuits that humans use in audio system. They are analog and digital echo or delay circuit. PT2399 is chosen in this project because it has both ADC and DAC, also a memory 44 Kbit. The PT2399 is a single chip echo processor IC utilizing CMOS technology which accepts analog audio input signal, a high sample rate ADC transfer the analog signal into a bit stream then storage to internal 44Kbit RAM, after processing the bit stream will de-modulate by DAC and low-pass filter. Overall delay time is determined by internal VCO clock frequency, and user can easy to change the VCO frequency by changing the external resistance.

5.6.1. Schematic Echo Circuit Design

PT2399 echo chip are working in the low power supply, and they are easy to build than using other IC echo chips because it has an ADC and DAC that are built inside the chip. PT2399 is a good chip since the echo and feedback effects is adjusted by the potentiometers. When testing this chip PT2399, the designed circuit is testing multiple times when using this PT2399 echo chip. The first thing is to make sure the power supply must be low. The PT2399 is working on the range of supply voltage (4.5 – 5.5 V). This is met the requirement. Echo Amplifier requires power supply between 4.5V5.5V for its minimum operated limit. This is a challenge for the project to find the battery that could meet the requirement. The voltage regulator is used to step down voltage 5V from the battery 12V. The design of the echo amplifier is complicated. It has two potentiometers to adjust the magnitude of each functions: the delay time, and the feedback. All in all, by using this PT2399 echo chip, the sound that is recorded from the microphone with the echo and feedback adjusting by potentiometers, but the output is still low and small noises, so the volume amplifier is decided to use for boosting the output. Moreover, the input signal will decrease tremendously after being filtered by this amplifier. Therefore, the output signal need to gain at least 10 dB by another amplifier to have the sound be hearable. Also, because the output signal of this echo amplifier is small for human to hear, this echo amplifier is designed like an affect to change the signal for delay time, not to change the volume, so this echo amplifier is put in the middle of two volume amplifiers. For the schematic of the echo effect design, the designed circuit is used with the PT2399 digital chip, and this chip is a single chip echo processor IC utilizing CMOS technology which accepts analog audio input signal, a high sample rate ADC transfer the analog signal into a bit stream then storage to internal 44Kbit RAM, after processing the bit stream will de-modulate by DAC and low pass filter.

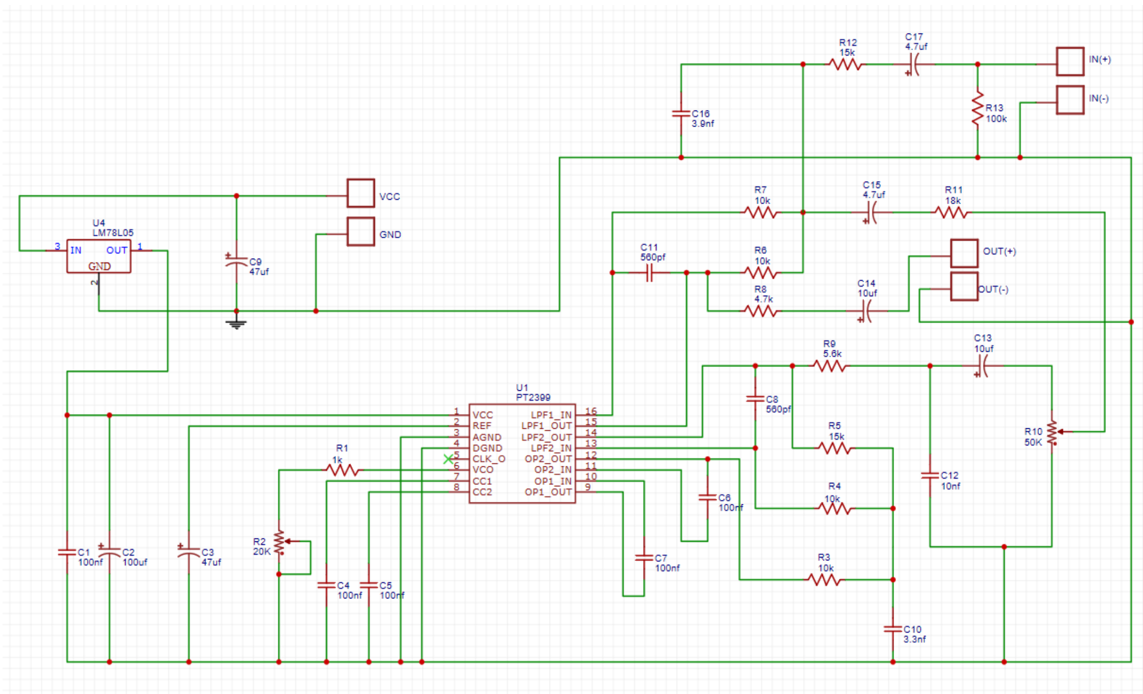


Figure 29: Echo Amplifier Circuit Diagram

The input signal is connected to the potentiometer 100K. This potentiometer is to adjust the echo effect of the signal from the microphone. At pin 1, there is a 5V supply voltage typically. For the output, the output signal is on pin 15, and it connects to the speaker. These circuit is used to test this echo chip PT2399. The PCB design is below with two potentiometers:

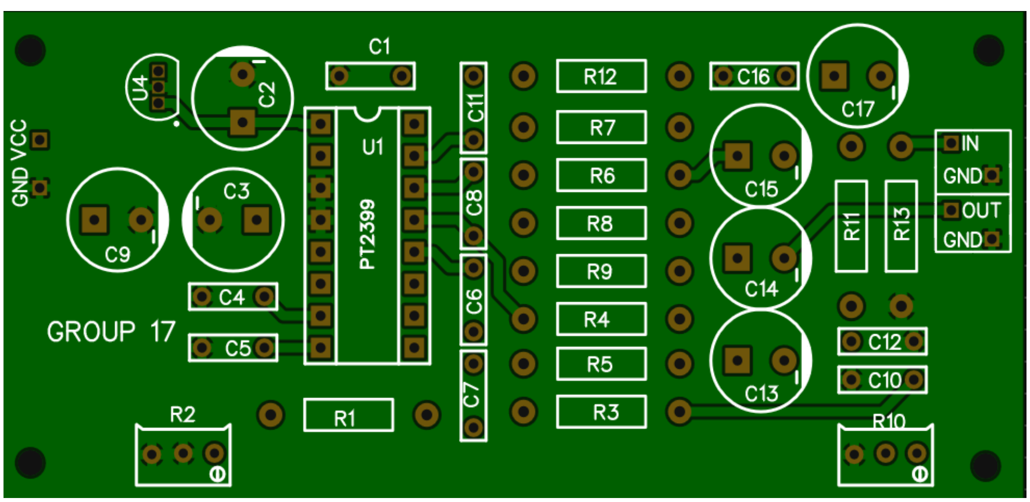


Figure 30 : Echo Amplifier PCB Board Design

In this PCB design, the dimension is about 2x1 inches that fits to the KPS case. The board can take 12 v power supply from the battery, then step down to 5V by voltage regulator. There are two potentiometers. Potentiometer R2 is to adjust the echo effect, and potentiometer R10 is to adjust the feedback.

5.7. VTVD Eagle Design

The schematic designs for the LED module system are described in this section. This includes the schematics for the power, voltage inverter, op-amp, external clock, timer, ADC, and the microcontroller.

5.7.1. Power Schematic design

Figure 27 displays the design for power for the VTVD. From the left, the decision on how and when to get power was made by allowing for the VTVD to be modular. The three pins represent a common barrel or DC jack used for many applications. The 9 volts in value is chosen for many reasons. First and foremost, for Buck Converters to function, a value greater than the desired output is ideal. However, most, or close to all, IC circuits require 5 Volts for operation. A switching regulator (Buck) was chosen to be able to drop down the voltage to 5 while still maintaining high efficiency (approximately 90%). The second reason for choosing the 9-volt input is that many adapters that plug directly into the wall are available at this range. The TPS562208 Buck Converter has a small footprint. It has been decided for prototyping that all components such as resistors and capacitors will have a minimum size of 0805. Yet, the switching regulator's footprint will still be relatively large due to recommended PCB layout guidelines dictated by the Datasheet as well as the inductor and larger value capacitor. This provides one drawback to using a switching regulator versus a linear regulator. More components are required, and it can be costlier to implement.

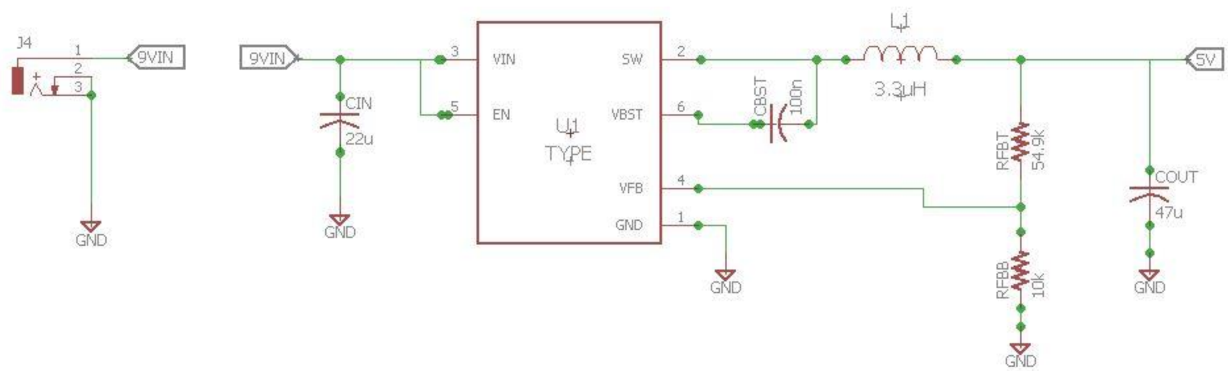


Figure 31: TPS562208 Buck Converter

5.7.2. Voltage Inverter Schematic design

Figure 28 demonstrates what is essential for providing an operating rail voltage for the Dual power supply op amp. The ICL7660S chip has a multitude of functionality. However, one of its most common uses in application in functioning as an inverter. The drawback to using this component is by its very nature, adding more IC's to the PCB. The option to use a single rail op amp is still not completely out of the equation. However, for the first prototyping VTVD, the dual rail was chosen. This inverter takes the 9 volts from the DC barrel and inverts the signal into a negative 9-volt value. The capacitor values for the ICL7660S were determined by the Datasheet. It is worth noting that this is a rather simple footprint for the inverter versus using a 555 timer to invert voltage values.

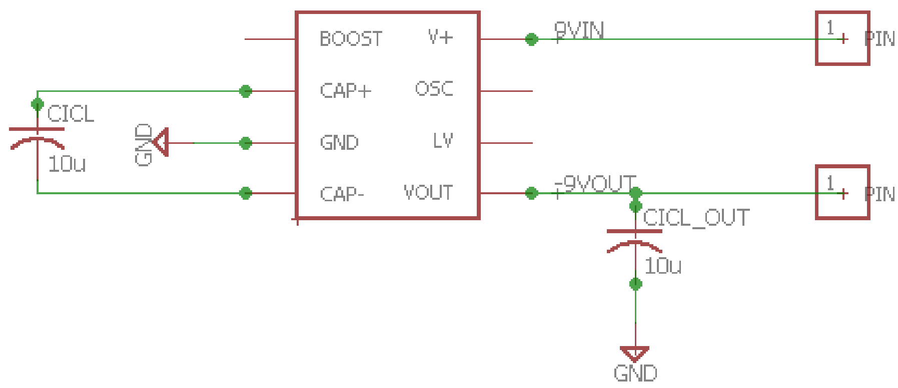


Figure 32: ICL7660S Voltage Invertor

5.7.3. Op-amp Schematic design

The Op-Amp chosen for enhancing the small input signal of the voice is the UA741P shown in Figure 29.

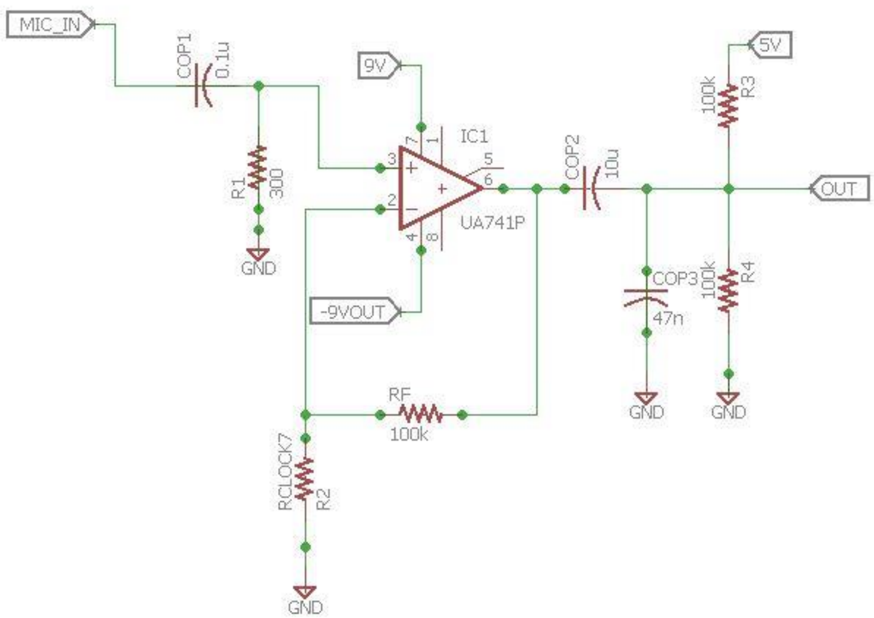


Figure 33: Dual Rail Op-Amp

It was chosen for its General-Purpose Op Amp class given by Texas Instruments. Taking a closer look, one can see that pins 1,8, and 5 are not being utilized for the VTVD. Pins 1 and 5 are used for offset-voltage null capability. This will not be required for the VTVD. Pin 8 is designated for "No Internal Connection."

5.7.4. ADC schematic design

The third and final schematic (pictured at the bottom of the figure) is the Analog to Digital (ADC) converter. The IC ADC121S101 is configured with bus pull-up resistors on CS, SDATA, and SCLK. It is imperative that the resistors are pulled by the same voltage potential as the reference V_A. This allows for all logic levels of all devices on the bus being compatible. Choosing resistor values for the bus was dictated by the datasheet.

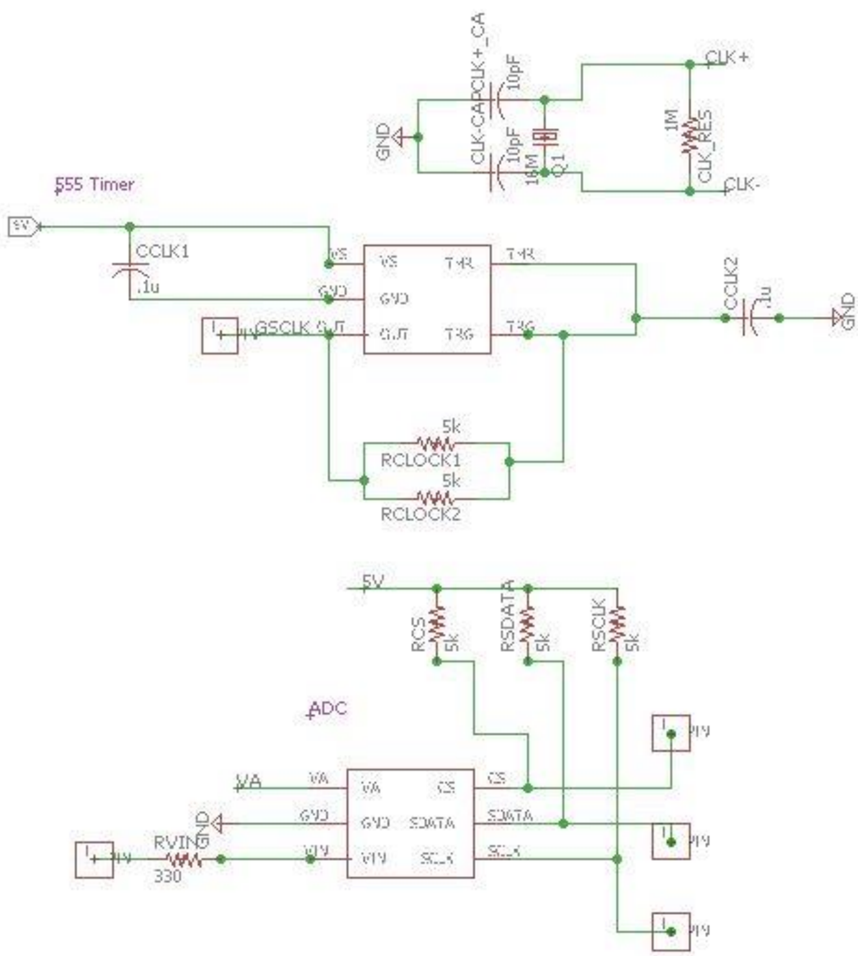


Figure 34: External Crystal(Top), Timer (Middle), ADC (Bottom)

For Standard or Fast Mode bus configurations the recommendation is to use $5k\Omega$ resistors. A recommendation for successful use of an ADC is having a dedicated line for the reference voltage V_A. The reason for such a requirement is because noise in the reference can change the actual interpretation by the ADC allowing for more errors to occur, and causing much less efficiency.

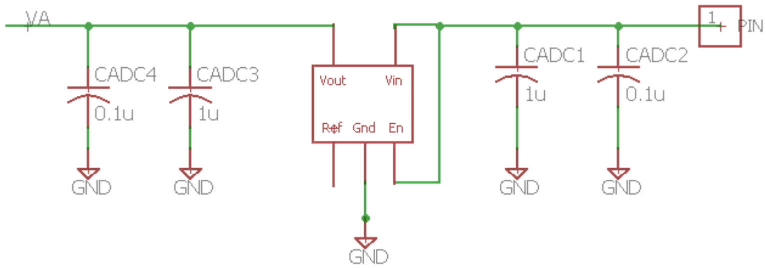


Figure 35: LM4120 Precision Voltage Reference

The Datasheet for the ADC suggests using a separate regulator or, for the VTVD, a precision reference Voltage supply was used. Figure 31 depicts the LM4120 that will be used to buffer the unregulated 5 volts with 0.05% accuracy over temperature. This should allow for a much more efficient reference.

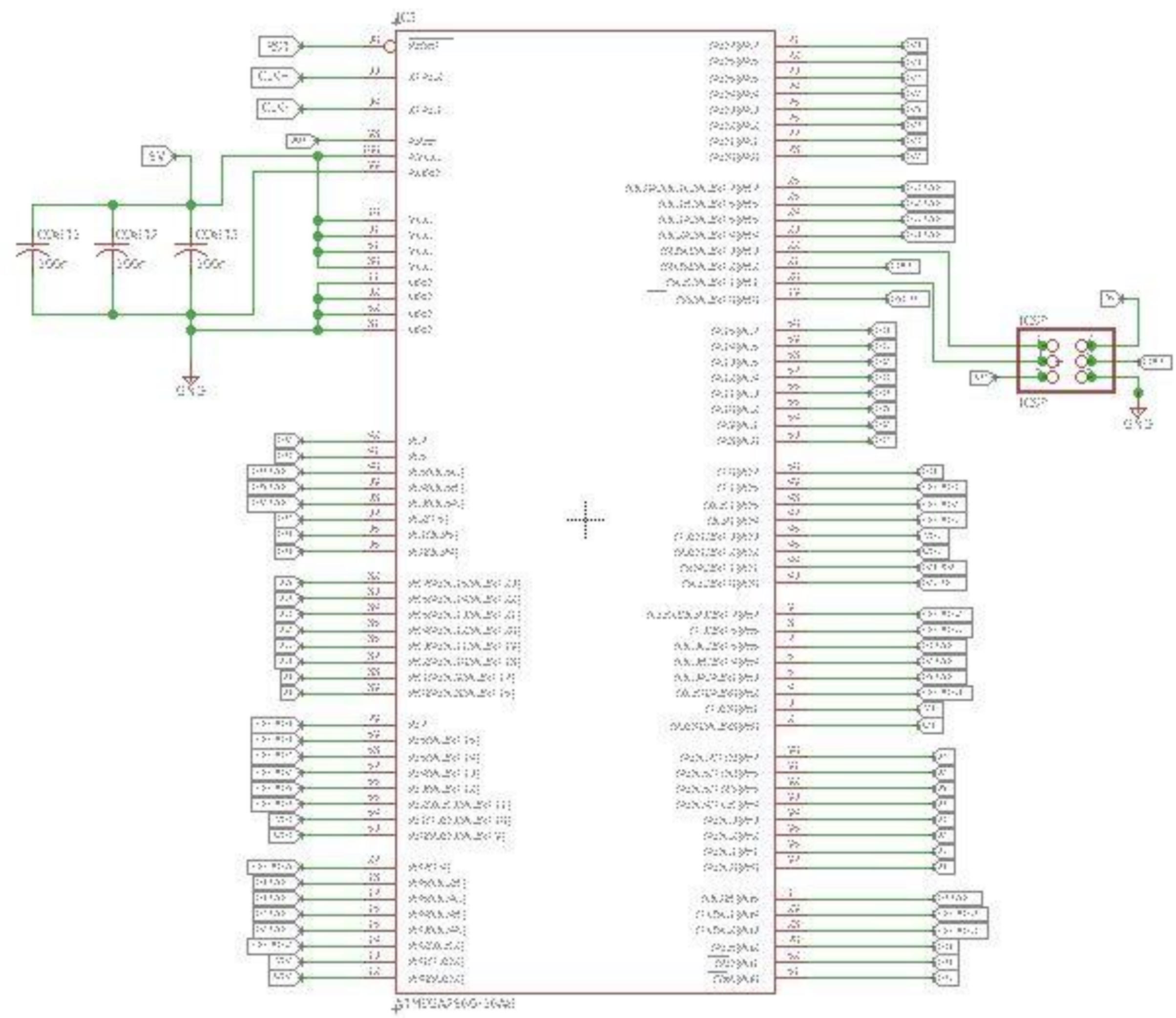


Figure 36: ATMega2560 Microcontroller

The capacitors on both input and output also provide more noise immunity. The drawback is much the same as introducing the inverting voltage IC. Using the LM4120 adds more components to the PCB (a bigger footprint), adds cost to the system, and adds to the overall power consumption of the VTVD.

5.7.5. Microcontroller schematic design

The ATMega 2560s schematic is depicted in Figure 32. The chip was chosen more for its onboard memory than its number of Input/Output (I/O) pins. It is interesting to note, that even Arduino developers have chosen not to set 15 pins for the ATMega itself. These pins are simply not mapped and cannot be accessed by any current development board. If these pins were chosen to be utilized, another environment would have to be chosen. The few small schematics pictured connected to the ATMega is for basic operational functionality. The capacitors on the top left are chosen to once again clean the voltage supplied by the buck converter. A simple button will also be configured to reset the microcontroller on pin 30 of the ATMega chip. The 6-pin configuration to the right of the chip is the ICSP (in-circuit serial programming) header. This allows for the ATMega to be programmed while already soldered to the PCB, versus the alternative of programming it prior to putting it on the board itself. The header consists of using three SPI (Serial Peripheral Interface) pins (MISO, MOSI, SCK) and the pins used for power (5V), ground, and reset. These six pins will be used to re-flash the firmware.

5.8. VTVD Software Design

The LED display is responsible for using the incoming input signal picked up by a microphone or device (connected to the display) and sampling it to obtain the frequency and amplitude of the signal to output the proper display. The LED display will map a specific RGB color to the 12 notes of the Western scale, read from the ADC. This feature will allow the performer to receive visual feedback of the specific notes he or she is singing.

5.8.1. Frequency Detection Algorithm

Accurate frequency detection is important for identifying the frequency of a signal. Open source software was written and embedded into the system to display the array of LEDs. This section explains the code (frequency detection algorithm) used in the system. To make this algorithm possible a 12 bit-ADC detected the incoming signal and converted the input analog signal into a digital signal. To properly capture the analog signal and make the conversion to a digital signal, port manipulation was used and coded in the Arduino IDE for the ATmega2560.

The ports that were available for the ATmega were PORTB and PORTH. The ADC required a 12bit resolution, however since the data being read was of a byte data type, the data needed to be shifted from 16-bits being read in to 12 bits. The ADC has three controls to make the proper conversion of signals, according to the datasheet. These are the chip select, clock, and data pins. The first port (PORTH) was used to enable the ADC by set the chip select high. PORTB was used to enable the CLOCK while in a for loop that ran at a size of 16. The data read was then shifted over to only capture 12 bits of data necessary for the ADC used in the system. Once this function completed the data read from the ADC was sent back to the main program.

Once a digital signal was received, manipulation of the incoming data being read was mapped to the LED display. The frequency detection algorithm first set a threshold from the signal. Refer to figure 15, two dashed lines represent the potential threshold values that could be implemented for an incoming signal. Notice that if the threshold is too high, at 0.4, the signal will not be completely captured, and some information could be lost. At 0.2 as a threshold, a complete cycle would be captured, and possible analysis could be completed. The approach was capturing and analyzing in the frequency domain. The microcontroller (ATmega2560) quickly captured the incoming signal and applied the thresholds that would dismiss undesirable ranges. The algorithm extracted the data to find one full period of the signal and ensured that at least more than one period existed to exclude any noise that could be mistaken as an incoming signal.

Once the period of the signal was confirmed, taking its reciprocal yielded the frequency. The simple equation below is

$$T(period) = 1/f(frequency) \tag{1}$$

When this was completed, a general mapping was written and tolerances for ranges of the detected signal were set. Lower frequencies posed a problem with having a universal set of ranges because the frequency proximity is much closer and therefore required a tighter range than those in the higher-frequency range. By using a table of note frequencies that gave the notation for pitches on standard piano key frequencies, each RGB LED was accurately mapped. This was accomplished by use of a struct data type composed of various RGB colors that could be called from the main program. The American Standard Pitch Notation (ASPN) also made it possible to accurately map the LEDs to the correct musical note. The American Standard Pitch Notation is a method of specifying musical pitch by combining a musical note name and a number identifying the pitches' octave.

Table 20. Summary of Note Frequency

	C	C#	D	EB	E	F
0	16.35	17.32	18.35	19.45	20.6	21.83
1	32.7	34.65	36.71	38.89	41.2	43.65
2	65.41	69.3	73.42	77.78	82.41	87.31
3	130.8	138.6	146.8	155.6	164.8	174.6
4	261.6	277.2	293.7	311.1	329.6	349.2
5	523.3	554.4	587.3	622.3	659.3	698.5
6	1047	1109	1175	1245	1319	1397
7	2093	2217	2349	2489	2637	2794
8	4186	4435	4699	4978	5274	5588

5.9. Combined Hardware Schematic Design

In this section, our group will discuss the final schematic for our hardware design including the echo effect chip from microphone and the tone control music effect from the Bluetooth module. After going through of testing of each individual components, our group feel that this is a great first iteration of our design when moving forward towards senior design II. This hardware schematic combines all our previously tested simulated schematics that adds the microphones for the input of the echo chip and the Bluetooth module for the input of music effect. Although this schematic tested well through simulations from Eagle cad, so now we can imagine how the whole system of how our project goes. Next semester (Spring 2018), we plan to order our printed circuit board during the small Christmas and New Year break between the end of senior design I (Fall 2017) and the beginning of senior design II (Spring 2018).

To test the overall design, we plan to first put our design through a power system test. To do that, we have to research all the power supply of our components to know the range the power supply. After that, we design the power system and make sure all of components in this project are working. In this test, we will measure all of the values of our resisters and capacitors to make sure the input and output of our project are shown as expected with the simulations. When we feel confident for our power system design, we turn the power on in the circuit and determine whether our printed circuit board is operating or not. This is a first version of testing this combine hardware design, so we must make sure to put several tests again to make sure the circuit can work properly.

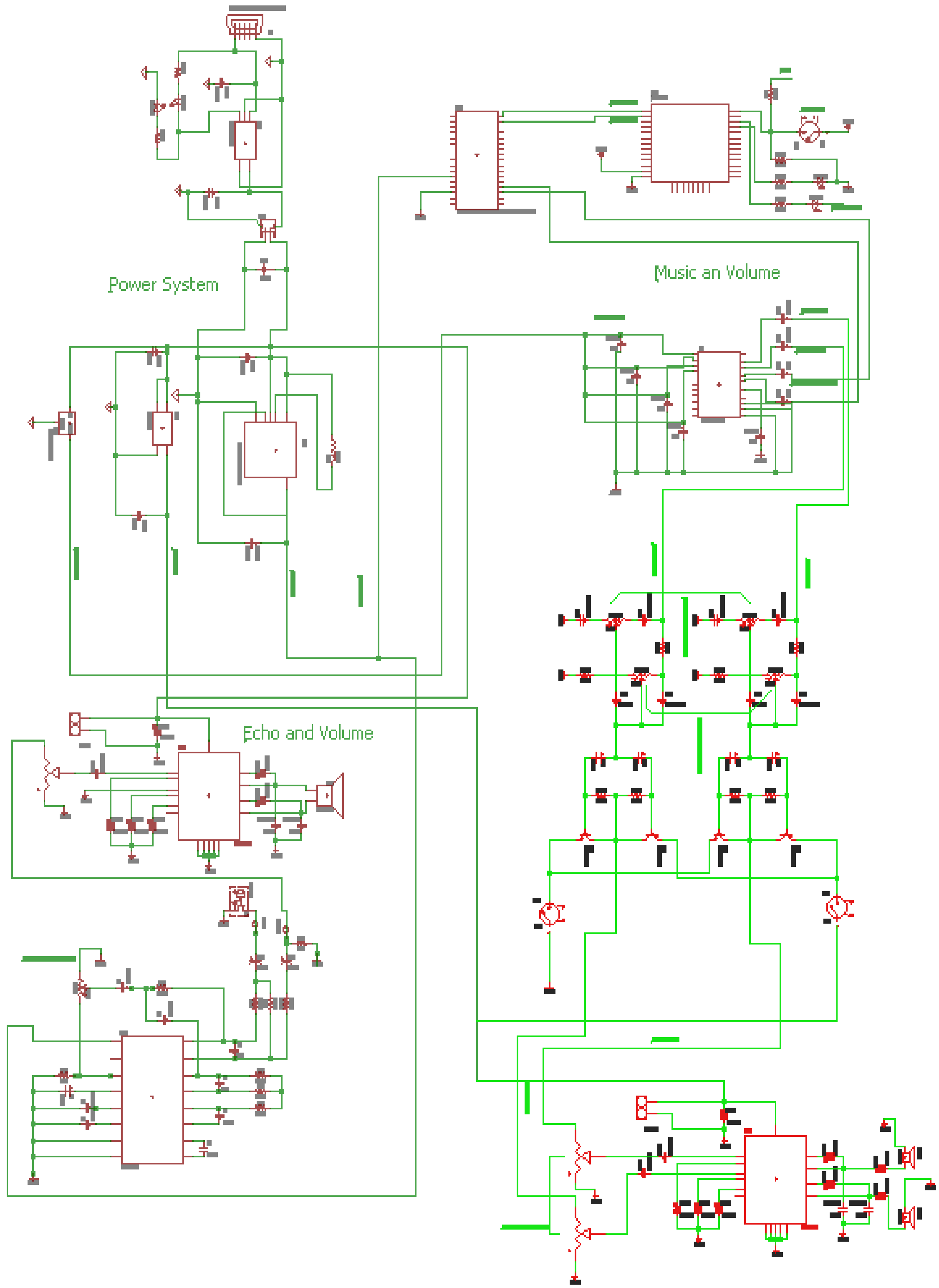
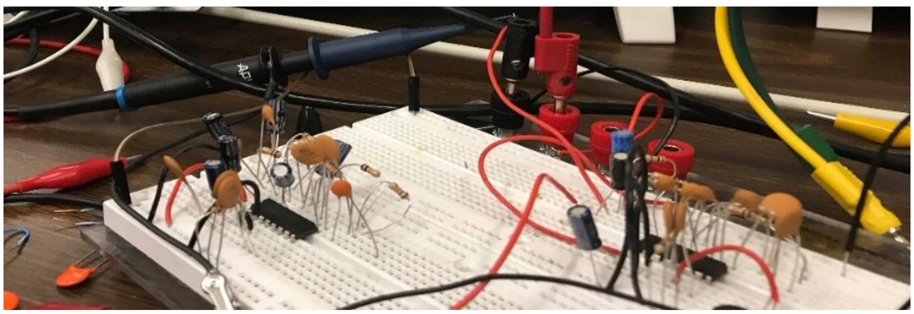


Figure 37: Combination Hardware Schematic Design

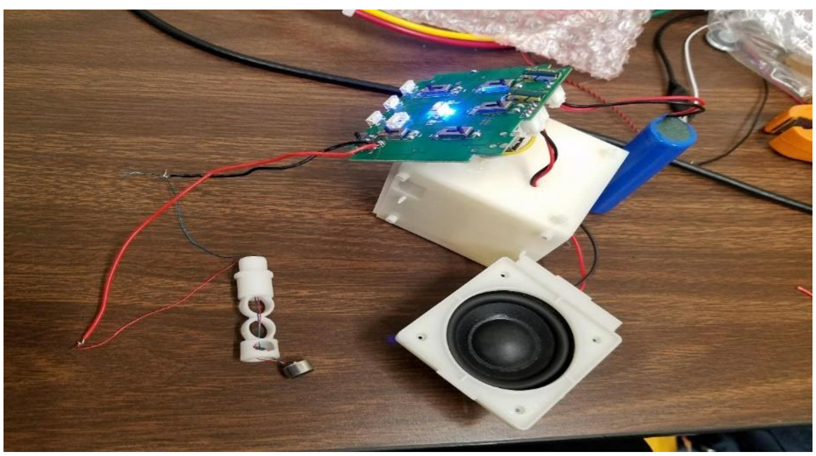
Next semester (spring 2018), when in senior design 2, we like to test in again and order the PCB design as soon as possible, so we make sure that we do not run this issue in the last minute. The connection diagram below is the combined hardware schematic design. When testing the schematic below, the currents and voltages are changing, and not as expected when simulating. We thought that the problem is the component percent errors, and it may be a short somewhere, so we should make sure get the right values of the components. In senior design 2, designing PCBs board and soldering the components into the PCBs board are very important. To accomplish this, our group has been researching potential companies who can make our PCBs correctly. Because when they design a bad printed circuit board for our project, it spent a lot of our time consumption, and our budget is very important. Moreover, our group has researched the safety of our device and has gone through some tests to ensure that no short circuits that may cause a critical failure within our device in our senior design 2.

Therefore, in senior design 2 next semester, our group is planning to make sure that choosing a good quality component, test the short circuits in the entire of the whole system design, and order a good PCBs board, so we can finish it on time and avoid running in the last minutes.

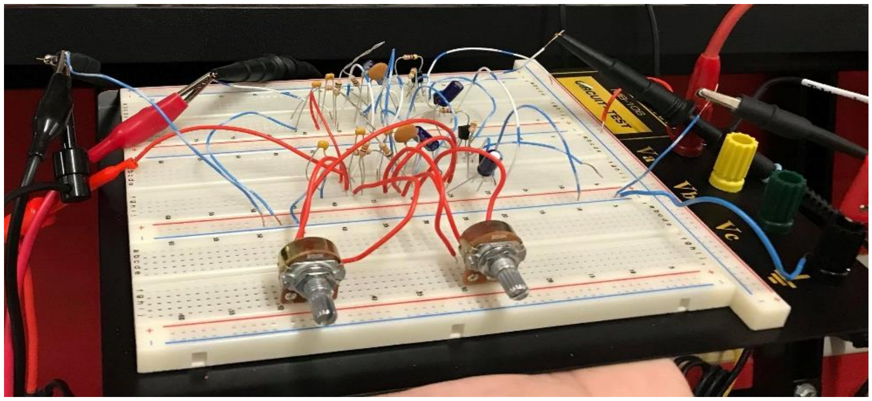
5.10. Breadboard Testing Components



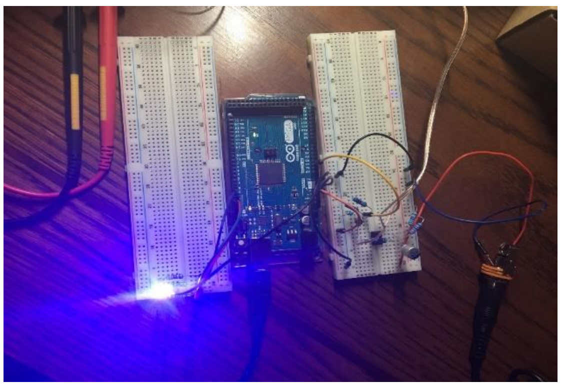
(a)



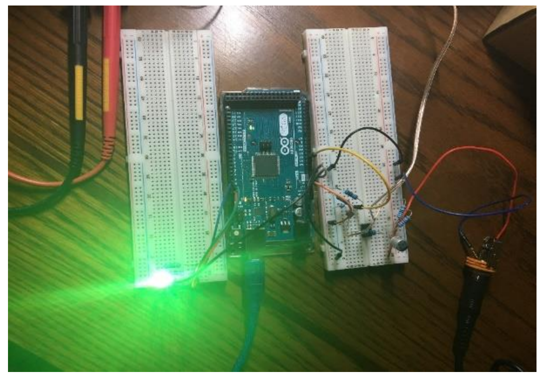
(b)



(c)



(d)



(e)

Figure 38. (a) Echo and Volume Amplifier Test (b) Battery, Speaker, and Microphone Test (c) Bass and Treble Amplifier Test (d) Keynote “F” Test (e) Keynote “B” Test

6. Project Prototype Testing Plan

Testing will be conducted in the final weeks of the semester. Each schematic for each system block will be built and joined together. The hardware portion will be tested, and each pin will have its current and voltage monitored. The hardware will also deal with configuring the pins properly. Following a successful hardware test the software test will follow. The software will test to eliminate the tones as specified in the tones section

6.1. ADC

The ADC aspect of the system will handle receiving the input signal from the microphone. In order word, the output signal from microphone is analog and will go through ADC to become digital signal. Therefore, it can go through digital amplifier circuit to amplify the signal. There are two components to the ADC aspect of the system. The first component of the ADC aspect is the microphone which will handle receiving analog audio signal from the user. The second component is the ADC aspect is the ADC device itself which will process the audio for digital amplifier circuit to handle the digital signal. The ADC must be tested properly, if not the Portable Karaoke Microphone aspect of the code will produce the incorrect signal which will cause the system to produce noise and not clear sound from the microphone.

The procedure for testing the ADC software is to have an audio device connected to the ADC. The microcontroller will then be connected to the computer. Then, the audio signal will be sent to the computer after the I2C communication. The audio signal will then be saved on the computer which will be later used in MATLAB to compare the input signal and the output signal. Using MATLAB, the team can compare the quality of the audio signal and to see if there is any degradation of the audio signal.

6.2. DAC

The DAC aspect of the system will handle send the audio signal from the microprocessor or from the digital amplifier to analog sound. There will be digital signal from digital amplifiers; therefore, it will need to be converted to analog signal to the speakers. In order words, there are two components for the DAC aspect of the system. The first component for DAC aspect is the DAC device itself. The second component for the DAC aspect is the audio jack which will output the audio from the microcontroller. Those digital signal will go through digital amplifier anyway. The DAC aspect must be tested properly to ensure that it will output the audio signal properly without any noise. If the DAC produce any noise, it would make the speakers of Karaoke Portable System redundant due to the additional noise.

Similarity, the first concern before testing the DAC device is to ensure that the software has the correct pin that the DAC device is connected to. If the program does not have the correct pin value, it will not be able to communicate with DAC device to send the output. The second aspect to consider before testing the DAC device is to ensure that send the output to correct channel, since each channel of

DAC will represent each side of the speakers. The final aspect to consider before testing the software component of DAC device is to ensure that the SPI structure is initialized, and that the SAI structure is properly calibrated to handle the communication between the device and microcontroller by ensure that it the system allow interrupt.

The procedure for testing the DAC software is to have a constant audio signal store in the microcontroller to ensure the microcontroller have enough space to store the signal, since the microcontroller cannot support USB transfer with a high enough transfer rate without delay or jitter. The microcontroller will constantly loop the audio signal. The team will then connect a headset or a speaker to test if the DAC can play the audio signal.

6.3. Analog and Digital Chip Test

For testing this component (PT2399), this chip is a delay digital chip that combines ADC and DAC inside the chip, so we thought we can put a sine wave from the function generator, then the output will be delay the phase. The output is shown on the oscilloscope. The purpose of this testing is to know the chip is working or not and what problems is happened when we build the circuit. The testing circuit that we built is from H&G Amplifiers website [59]. The picture below is the breadboard that has two circuits. The circuit above is an analog circuit, but it is not working properly. The second circuit that we build is using digital chip (PT2399) that worked properly.

These two circuit which we build are on the same breadboard. The first circuit on the top of the breadboard is using analog amplifier, but its results are not the results that we are looking for. The second circuit is using a digital echo chip, and the results are shown on the oscilloscope of the picture below. The result is that there will be a delay in the phase from the output to the input. The picture for testing is below:

In this Figure 26a, there are two circuits, but only the circuit 2 (bottom) is working when we use an echo digital chip (PT2399), there are 5 V voltage supply that we use. The results we have for the circuit 2 on the oscilloscope. In the figure 26a, the input signal is the yellow wave, and the output signal is light blue wave. The delay from signal 2 to signal 1 is -172.7^{o}, and it is clipping since there is problem on the limit of the chip. They all are analog signal, and the voltage gain is 2. The voltage gain is almost the same to the datasheet (2.5 max).

6.4. Testing the Battery Charging Station

The charging station is an important part of the project due to the battery dangers that could occur if improperly used. This portion describes a simplification of the steps that will be measured out in order to test the device and be confident enough for further applications and system be able to reach a certain size this could result in over charging. A way to bypass this safety issue is to test the battery with a max voltage charge less than what the battery can hold. Since the battery we’ll be using is roughly 4V we will make the make the max charge of that batter be 3.5 by changing the diodes and resistors values.

Once implemented the user will check the battery charge value, if the value fits with the value trying to be achieved for a long period of time, the charging system is capable of producing a safety feature that can be implemented onto the actual device.

Afterwards we can implement the device safety at a max voltage which should work, however there may need to be some adjustments with the resistor values along with diode position. This should be watched carefully as battery over charging can be dangerous. If the device works, the safety feature should be complete.

The regulator portion should be implemented next. This I regulator will be the input line from the outside power source to the battery. This portion will also be the efficiency factor of the battery charging unit as we will be using a PWM switching regulator, to charge the battery up rather than using a steady equivalent voltage source.

Afterwards LED light testing can be implemented using the switching regulator by changing the voltage of the battery which can be checked by changing a battery source. When changing the pretend battery source different LED will turn on for each type of voltage. It will need a minimum voltage to be powered which will represent the battery and will be changing at different voltages for confirmation that the device properly displays its objective.

Lastly, we can implement the female USB port which will act as our external power supply that can charge the battery as the basic power source was doing before. This will be the finalization of the external source.

One thing that should be noted is that the LED lights should not be powered by the battery, only when the battery is being charged for this component, or when the device is on. To implement this a switch must be connected that actives when the USB port is being powered. To test this the user must only use the USB supply power source, and check the light for connection.

In last section, we test the battery. The battery that we use is Lithium 3.7V, 2200 mAh. To test the battery, we use the multimeter in the electronic lab at University of Central Florida. The batteries that we bought is from Digi key. We bought two of them. When testing these two batteries in the lab, the first battery one we have is 3.912 V, and the second battery which we have is 3.993 V.

Therefore. These two batteries are met our requirements since most of our components are using the voltage range from 1.8 V to 3.6 V. Also, we order a step-down voltage regulator is 3.6 V, in case some of our component has the voltage supply needs to be step down to 3.6 V since the actual value of our batteries that we bought are almost 4 V.

6.5. Testing the Power System

Testing the device powering system is important in order to provide a safe and reliable power source that can be used with multiple devices that the user implements.

With the now fully functioning battery charger device the second process can begin. The first implementation is to create a switch that can send a voltage line to the system. This switch must create a voltage source when turned on. If not on there must be no voltage source draining the battery from the source.

When a voltage is sent towards the line, it must turn on a light to demonstrate that the power source is on. This can be easily implemented by having the user use an LED light and a switch. The user would just have to connect the switch to the load line acting as a VCC and when the device is on the LED light should turn on ad when the switch turns off the LED light should dissipate.

What the user wants to implement next should be a way for the power supply to work as it is being charged or vice versa. This can be accomplished through using a switch that turns on the system which should be also connected to the charging station.

With both parts of the system working the user can use the LED lights to be turned on from the battery if there is no device charging the battery. This can be implemented similar to the switch where the led are also connected to the battery source by a switch, however it should prioritize the charging systems USB input.

Once that is all completed there must be a line that can produce both voltage sources. The best way to do this is by using a step-down regulator to convert from the voltage of the battery to the desired voltage which is 3.6 and 3.3. Once that is implemented the user should check the stability of the device by checking fluctuation of the current as well as drop in voltage with an oscilloscope.

Finally, the implementation of test voltages and sources should be used so that the system cannot harm the components that will be placed at the end. Should the system work it should be ready for implementation with the actual components themselves

6.6. Volume Amplifier Circuit Testing

TEA2025 is an audio volume amplifier chip, and it can have the gain 100. We use this chip to design the volume amplifier of our project. We build two designs of this chip. One is for music volume with two speakers, and another design is for echo volume amplifier from the microphone. Both are the same design. To design this volume amplifier, first, we make sure the power supply is met our requirement with the range from 3 V to 12 V. the input of this design is an analog music input, and the input can get it from the jack with three pins. First pin is the tip (hot wire), second pin is ring (cold wire), and the last pin is the sleeve is the ground wire. After making sure where the music come from, we start to build the design. It has two potentiometers. Each of them is 10K. Because these potentiometers are the same, so we use only one potentiometer as dual potentiometer to adjust the volume of the left and right channel inputs. The voltage gain itself is 45dB, so after the inputs go into this chip, the outputs have the voltage gain of 45dB. Our outputs are two speakers, and each of them is five watts. The first test of this design, we know that the chip is working well, we can hear loud enough in the big room. However, there are some noises, so we must improve the circuit again. All in all, the TEA2025 chip is very good for design an audio volume amplifier

6.7. Bluetooth Testing Plan

For this testing, we are planning to use the Microcontroller Arduino and a Bluetooth Module HC-05. In the Bluetooth module, there are five pins: VCC, GRD, TXD, RXD, and KEY. The VCC pin is for the power supply that connect to the power supply (typically 5V). GRP pin is connected to ground. TXD pin of HC-05 is connect to the microcontroller RXD pin. RXD pin of HC-05 is connected to the UNO Arduino microcontroller TXD pin. The Key pin is connected to the air of communication mode. When the Key pin is low, this is a mode before fairing mode, and when Key pin is high, this is a paired mode or AT mode. The VCC or power supply is connected to HC-05 and microcontroller UNO Arduino. The switch is used to tur off and on the power supply. When completing hardware and source code installation on Arduino UNO, the next step is setting up PC site. the Bluetooth device plug is used to connect to the PC to communicate to our HC-05 Bluetooth module. We must connect the microcontroller to PC to code for that and setting the communication between the HC-05 and PC. The Key default password is 1234, so after the password is corrected, the window will show which serial COM is connected to HC-05 Bluetooth Module. To write the code to the PC, we need to open software "Mybotic Serial Com Tool". In this software, we can set the Baud rate, and comp serial. At this point, we can the data from the PC and connect the Bluetooth from the smartphone. All in all, HC-05 Bluetooth Module is good device to help us get the music from any devices that support Bluetooth. Also, HC-05 is very reliable and easy to use and fast connection.

6.8. Base and Treble Circuit Testing

Audio transistors are working in the low power supply, and they are easy to build than using an audio chip. When testing this schematic, we want to make sure this circuit is working when using these audio transistors, so the input signal of this testing is from the jack that has three pins (tip: hot, ring: cold, sleeve: ground), the input is an analog signal. The input signal is connected to the potentiometers to adjust the music effect (bass and treble). Before the inputs go to the transistors, we need to make sure the input signal can adjust, so we use four potentiometers 100K. They are the same values, so we can combine them as two dual potentiometers. One is for bass adjustment known as VR1, and another one is for treble adjustment known as VR2. The output signals are connected to the speakers. We use two audio transistor 2SA1015 low power supply, and the other are 2SC1815. The whole circuit is built to test how the music can change when adjusting the bass and treble. The result of testing this circuit is not working very well since we still do not have the treble effect on the music. All in all, by using these audio transistor, we still have the music from the speakers with bass and treble effects, but the volume is still low, so we decide to build a volume amplifier from them.

6.9. Echo Testing

PT2399 echo chip are working in the low power supply, and they are easy to build than using other IC echo chips because it has an ADC and DAC that are built inside the chip. It is good since we can change the echo effect by the potentiometer. When testing this schematic, we want to make sure this circuit is working when using this PT2399 echo chip. The first thing that we want to make sure is the power supply must be low. The PT2399 is working on the range of supply voltage (4.5 – 5.5 V). this is met our requirement. The input signals are an analog signal that comes from the microphone. In the chip, the pin 16 is for analog input signal that connects to the microphone. We are making sure the microphone is working on the frequency range (20Hz – 20KHz). Also, there is a potentiometer 100K that connects to pin 16 and pin 6. In PT2399 chip, pin 6 is for adjusting the echo effect by using a potentiometer. Now, we can change the echo effect by using a potentiometer that connects to pin 16 and pin 6. In PT2399 chip, pin 15 is for an analog signal output. On this pin 15, we will connect a speaker (5 watts), so we can hear the sound. The result of testing this echo circuit is not working very well since we still do not have the small noise out from the speaker. All in all, by using this PT2399 echo chip, we still have the sound that is recorded from the microphone with the echo adjusting by potentiometer, but the volume is still low and small noises, so we decide to build a volume amplifier from them

6.10. Microphone and Speaker Testing

In this section, we are testing the speaker (4W output) and the microphone. To test these components, first, we build an analog circuit with an amplifier gain 100. The input of this circuit is the voice that record from the microphone, and the output is an analog signal to the speaker. The result of this testing circuit is quite successful since we have some noises, and the output sound is not clear. We though the reasons for that is from the microphone. The microphone that we bought is too small, and does not work well. So, we bought another microphone and testing again. At this time, the result is getting better than the first time using a small microphone that we have less noise and clearer sound. To make sure for the microphone and speaker that met our requirement.

We bought a testing PCB board from prime amazon. This PCB board is using for audio that we can change its volume, bass, treble, and balance. The power supply is 3.7 V. the battery using for this PCB board is the same battery that we bought for our project, so we think this is a good idea for us to test the battery, the microphone, and the speaker at the same time at the same PCB board. When testing this PCB board with microphone is the input, the speaker is output, and the battery is power supply for the whole PCB board. The result of this testing is a lot of better. We have clear sound coming out from the speaker with adjusting the volume, the bass, the treble, and the balance of this PCB board. The figure 26b is the whole circuit connection by using the testing PCB board that bought from prime amazon.

6.11. Frequency detection and Output Color testing

The testing done for the VTVD is described in this section. Includes the prototyping with the assembled parts that makeup the VTVD to check functionality.

6.11.1. Prototyping

Figure 35d and Figure 35e what seems to be simplicity, has proven to be quite difficult in execution. The code for the Arduino is reading the Analog input given by the Op Amp. The Arduino's ADC is being used to interpret the Audio input. However, when complete computation will need to be much faster, and higher resolution will be needed, the Arduino's ADC will no longer be a viable option for design. However, it does prove to be essential for testing audio inputs to validate code for interpreting frequency. In Figure 35d

the note F is played through a computer speaker, which is then picked up by the microphone. This note, in particular F_4, lies around 349.23 Hz. This note range was chosen as a simple test case and allows for a wider interval between its harmonic neighbors E_4 and $F_4^{\#}$; approximately 20 Hz difference. Using the mapped colors for frequencies table cited early, the note F should correspond to a Red-Violet color. Which is clearly shown in the figure above.

Figure 35e is another test case for the algorithm developed to use to detect frequency. Once again, a friendly frequency was chosen in the same octal range as before. This time, B_4 was played from the speaker, which corresponds to a frequency of 493.88 Hz. It's nearest neighboring frequencies are not 30 Hz apart. Further testing must be done for the complete range of frequency that a human voice can sing. Multiple iterations will be done to observe the behavior of the algorithm and to see if the ranges dictated for frequency are accurate enough to continue to use for the VTVD. Prototyping PCBs will be made for the Audio input, Op-amp, ADC, and power supply options.

7. Administrative Content

The content that follows tabulates a current milestone of the tasks to be completed and the status of each task for the completion of the project. The section also includes the budgeting for the modular system and the financing of KPS. A table for the budgeting and financing is shown below with a simple list of items that will be purchased in order to build and test the microphone and the LED system enhancement for the project.

7.1. Budget and Finance Discussion

Below is an estimated budget for the Karaoke Portable System. Currently the design of the KPS requires hardware components to build five amplifiers and analog digital converters (ADC). Breadboards and all the simulated equipment will be provided in IT lab or Senior Design Lab. Those parts are cheaper and extra will be ordered to account for design flaws during the prototyping stage. Purchasing the electret microphones and speakers to test the sound and amplifiers will be evaluated and thoroughly researched before purchasing and will be included in the budget. As of now those will cost approximately $35.

Consideration of any design laws is being accounted for and extra minor components will be purchased. The KPS will need two microprocessors to one read the digital data from the Bluetooth device to communicate with the microphone and the other microprocessor will be used to make the appropriates calculations for mapping the LEDs to each of the 12-notes of the western scale and trigger the LEDs to flash. The microprocessor that runs the system will cost approximately $15 minimum. Two PCB boards will be designed, one for the amplifiers, wireless device and DAC, the other for the microprocessor, LED driver, ADC and LEDs. As indicated in the requirements section the LEDs for the KPS will either be mounted unto the microphone or will be separate from the microphone. The reason for having a PCB board for the LED portion of the project is to account for separate modules each team member will complete. For the LEDs to flash while the performer is singing there will be buttons the performer can choose from. The performer can choose what options such as choosing to have LEDs flash based on the 12 notes of the western scale the performer sings. (i.e. If the singer sings in the key of A then an LED color will flash corresponding to that specific note) the performer will also have the option of choosing a preset option, these LEDs flash as programmed and cannot change. These features are being considered based on budget and any design constraints.

Table 20. Budget

KPM Parts List					
Description	Price per Unit	Developing Amount	Developing Total	Project Amount	Project Total
Charging Cable	Donate	1	$0	1	$0
TDA1524A	$2	10	$20	1	$2
PT2399	$2	10	$20	1	$2
LM386	$3	10	$30	2	$6
Audio/Recording Cable	Donate	1	$0	1	$0
PCB Boards	$20	10	$200	6	$120
Electret Microphone	$2	10	$20	2	$4
Cover (Fiber Glass)	$100	0	0	1	$100
Speakers	$8	5	$40	3	$24
Rechargeable Battery	$20	2	$40	1	$20
Bluetooth Module	$15	3	$45	1	$15
Microcontroller	$15	1	$15	1	$15
LEDs	$0.75	30	$40	30	$40
Misc.	$30	2	$60	1	$30
TOTAL			$530		$378

The second reason for budgeting two PCB boards is to compensate for any mistakes and for each engineer to work on sections of the KPS. Budgeting extra parts will also prevent spending extra time to replace any components and fixing any design flaws due to mistakes. The estimation cost for each PCB board would be around $50. This will be the bulk budget cost for the project. The rechargeable battery will be a 5V 22mAh inside and easy to remove. Those batteries would be fully charged at around eight hours and cost around $20. The batteries will be used to power the microphone module of the system. As for the LED module the power will be received from a wall wart to power all the components that make up the LED module and a possible voltage regulator to maintain the voltage at a steady voltage needed for the IC's being used. For aesthetic purposes the casing of the KPS will be a clear plastic material. This visual appeal will allow the design of the product to be seen. There will be LED lines with multi-colors attached around the microphone, however this is subject to change if the LED portion of the microphone does not fit the microphone design. This will be evaluated during the building stage.

7.2. Milestone Discussion

The table below depicts the current milestones of the project. With six header titles describing the purpose of the table. These identifiers include the tasks that will be completed and the status of each task. Such as the completion deadline and who is responsible for each task, as well as the status of the task. Each member is required to work diligently to complete the tasks designated to each member. To ensure the goals and deadlines are met.

Table 21. Project Milestones

Tasks	Responsible	Start	End	Days	Status
Senior Design I&II					
Meeting/Ideas	Group 17	8/23	8/23	1	**Complete**
Agree on objectives	Group 17	23-Aug	25-Aug	2	**Complete**
Initiation					
Project Selection	Group 17	23-Aug	23-Aug	1	**Complete**
Project Roles	Group 17	28-Aug	28-Aug	0	**Complete**
Development					
Project Documentation	Group 17	6-Nov	10-Nov	4	**In Progress**
Table of Content	Group 17	10-Nov	20-Nov	10	**In Progress**
Draft Document	Group 17	2-Oct	27-Nov	56	**In Progress**
Final Document	Group 17	5-Sep	4-Dec	90	**In Progress**
Prototyping					
Hardware Config.	Group 17	11-Sep	15-Sep	4	**Not started**
Filter Amplifier 1&2	Lam Dinh	2-Oct	31-Oct	29	**Research**
Filter Amplifier 3&4	Tuan Dao	2-Oct	31-Oct	29	**Research**
Power Supplies	Group 17	12-Sep	15-Dec	3	**Research**
LED mic display	Jennifer Franco	2-Oct	31-Oct	29	**Research**
Wireless Device	Group 17	12-Sep	16-Dec	4	**Research**
Design					
PCBs	Group 17	1-Oct	30-Nov	60	**Not started**
User Interface	Group 17	12-Oct	1-Nov	20	**Not started**
Website	Group 17	2-Oct	1-Nov	30	**Not started**
System Prototype	Group 17	11-Sep	4-Dec	90	**In Progress**
Launch		4-Dec	4-Dec		

8. Project Summary and Conclusion

In the beginning the group 17 had an idea to create a portable karaoke wireless (KPA) system. This device can work as a speaker for music, and a record our voice for karaoke. The whole system is a karaoke portable wireless. We planed the device the group divided the work into 4 subsystems. Three of which are hardware; microphone, signal and power system. While the final block was focused on software for LED and the nose. Each block was and is crucial to the success of the project. After planning the project into four parts, we start project and divide the works for each member. CPE is doing the software for the LED system, 2 ECPs re doing the hardware and power system. To make the project successful, our group is planning to research everything that useful for our project.

After all research and planning is complete, the next major step is to test each block component and to test the whole design in the lab and software. At this point, the 2 ECP are working very hard to test each of the components very carefully. We test in the lab and discuss the input and output of the design. There some discussions between us, but finally, we have a common decision. A CPE is testing the components that are related to the LED system. These tests are not the same time, but the inputs are the same, so we combine them at the end to make sure all the components are working. While testing the components and design the system, we learn a lot of things and experience. these experience and knowledges have given the group significant insight in successfully in our senior design 2. Each member has to critical thinking of how to test and how to fix the problems. The most problem which we have are combine the components together and working in the same power system because wants everything working in low power supply. This has each member in our group can think and have the same language to talk, so we are right now feel very confident toward the senior design 2. In conclusion, the device took many hours of research, studying, asking, and testing to accomplish the project. When the project is completed, it is used for everybody in our society. It is either a karaoke with music, or a speaker, or a megaphone for the teacher. In the future, the device will play the music via Bluetooth from either a computer or phone and sing to together by the design of tone control and echo effect with very low noises. The device if anyone choses to mass produce should be available to any consumer. This would require that the price per unit is increases, but it is very good quality music and device.

Appendices

APPENDIX A: Citation

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APENDIX B: Permission

One thought on “TRRS and TRS Plugs and Sockets Explained”

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Leave a Reply

Dear David Miles,

My name is Lam Dinh and I am student of University of Central Florida. I am currently working on a paper for my senior design class. I would like you use the figures in this blog of TRRS and TRS Plugs. Thank you for your time.

Best regards,
Lam

Fill in your details below or click an icon to log in:

Lamdinh93@knights.ucf.edu

Lam Dinh

Website

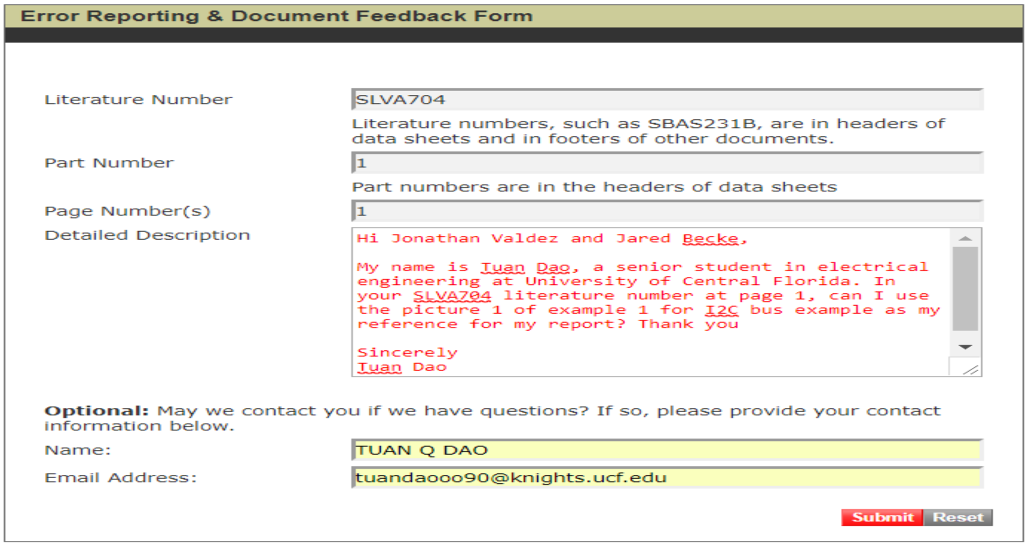
Permission of Using Figure

Lam Dinh <lamdinh93@gmail.com> 2:48 PM (0 minutes ago)
to info

Hello whom it may concern,

My name is Lam Dinh and I am a student of University of Central Florida. I am currently working on a paper for my senior design class. I would like to ask your permission to use the figures of "Microphones: Polar pattern / Directionality" article on your website.
http://www.shure.eu/support_download/educational_content/microphones-basics/microphone_polar_patterns
Thank you so much for your time

...



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Literature numbers, such as SBAS231B, are in headers of data sheets and in footers of other documents.

Part Number: 1

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Page Number(s): 1

Detailed Description:

Hi Jonathan Valdez and Jared Becke,

My name is Tuan Dao, a senior student in electrical engineering at University of Central Florida. In your SLVA704 literature number at page 1, can I use the picture 1 of example 1 for I2C bus example as my reference for my report? Thank you

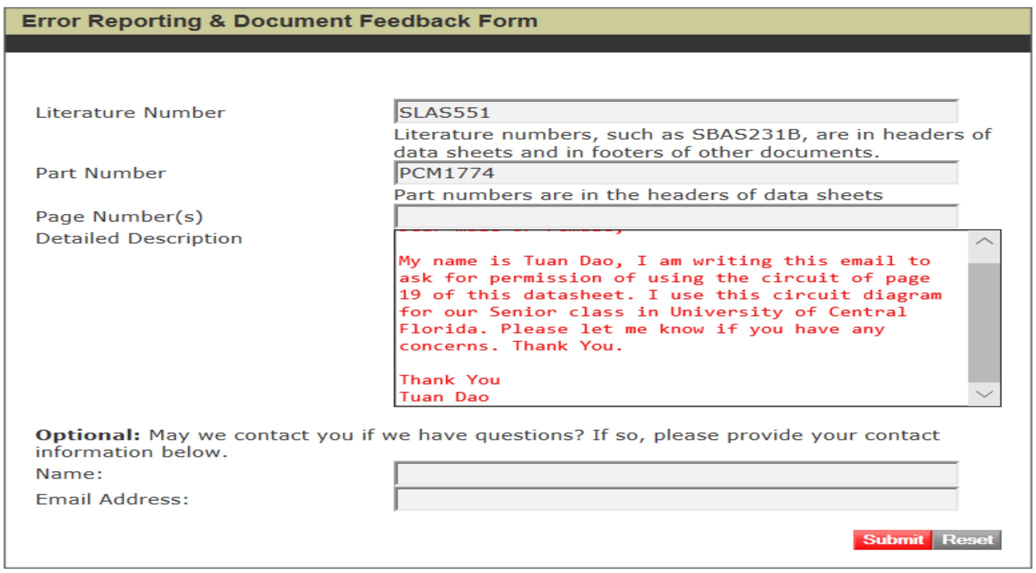
Sincerely
Tuan Dao

Optional: May we contact you if we have questions? If so, please provide your contact information below.

Name: TUAN Q DAO

Email Address: tuandaooo90@knights.ucf.edu

Submit Reset



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Detailed Description:

My name is Tuan Dao, I am writing this email to ask for permission of using the circuit of page 19 of this datasheet. I use this circuit diagram for our Senior class in University of Central Florida. Please let me know if you have any concerns. Thank You.

Thank You
Tuan Dao

Optional: May we contact you if we have questions? If so, please provide your contact information below.

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