Karaoke Portable System

Lam Hai Dinh, Tuan Dao, and Jennifer Franco

Dept. of Electrical and Computer Engineering
University of Central Florida, Orlando, Florida, 32816-2450

***Abstract* — In this paper, a karaoke system, KPS, is proposed. The system will incorporate Bluetooth® 2.0 LS for interconnectivity to multiple devices. Personal videos or an installed application selected from the user's multimedia database on their device, is utilized as the musical background. A third-party application will be used from the user's device to obtain lyrics if desired while performing. Additionally, different settings adjust the output in the system. These are bass, treble, echo or delay. Furthermore, open source software is embedded to display an array of LEDs mapped to musical notes based on pitch transcription by use of a frequency detection algorithm.**

***Index Terms* — pitch transcription, karaoke, octave, threshold, absolute pitch standard.**

I. Introduction

Karaoke is a relatively newer form of entertainment in America. Its origins lie within the Asian Market, starting with the first Karaoke machine made in the 1970s. The basic principle of Karaoke is to provide a back tracking for artists or hobbyists 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 Karaoke System 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.

II. System Components

To have a successful product, each of the components of the system will be designed to work properly together. The components will be either purchased or designed. The technical introduction of each of the components is described in this section.

A. 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. There three common types microphones are dynamic, condenser, and piezoelectric. Finally, the electret condenser microphone is decided to use. An electret condenser microphone (CMC-2742WBL-25L) from Digi key is the component to use in this project. This microphone is taking analog signals.

Electret condenser microphone offers a 60 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. This device costs roughly $ 2.42 per unit price and works within a 30 Hz – 15 kHz digital range which is downside of this microphone. 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 [1].

B. Echo Amplifier (PT2399)

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 [2]. 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 [3].

C. Volume Amplifier (LM386)

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 4-12v. 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).

D. Stereo Amplifier:

The amplifier will tone control the audio of your voice. It has four tone controls option: volume, balance, bass, and treble. It required 12 volts for power supply to operate. The stereo amplifier will take the analog audio signal from echo amplifier, tone control the audio, and send it to post-amplifier before going to a 10 Watts speaker. It goes up to 20 dB of voltage gain, and +/- 15 dB for Bass and Treble control. The main part for this amplification process is TDA1524 op-amp. This integrated circuit is manufactured by Philips and has a low total harmonic distortion (THD).

E. Bluetooth Module:

For wireless function of the project, Bluetooth technology is chosen. 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.

F. Battery:

Lithium-Ion battery will be used to supply for the whole system. It has high energy density, rechargeable, and low self-discharge. The LED and the speakers consume a lot of power. Therefore, the capacity of 9800mAH can make the KPS to operate 5 hours continuously. In addition, most of the amplifiers takes 12 volts for power supply. Lithium-Ion battery could supply that with small size as 5”x3”x1”. The battery can be recharged by LM3622 circuit. It takes up to an hour to have the battery fully charged.

G. Speakers:

There are three speakers in the project. Two 4 Ohms, 5 Watts speakers for left and right channels for Bluetooth module. To balance the volume, one 4 Ohms, 10 Watts speaker is connected to the microphone.

H. ADC

An Analog to Digital Converter does just as it states. It takes an analog input (the LED display) 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 sued 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 [5]. When researching different ADC's to use for the VTVD these parameter specifications will be taken into consideration for part selection and ADC type. The ADC used for the system was the ADC121S101, it has one of the highest resolutions, highest speeds and an affordable price.

I. *Microcontroller*

A Microcontroller is a small computer on a single integrated circuit. A microcontroller is designed to govern specific operation an embedded system. The microcontroller was used to program the ADC to ensure proper conversion from an analog signal to a digital signal. The microcontroller was also used to program the LED display. The ATmega was an ideal choice for the system. It 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.

III. System Hardware Concept

All the components were described from the previous section. The connection of the hardware will be presented in a block diagram, showing the I/O flow of the system hardware

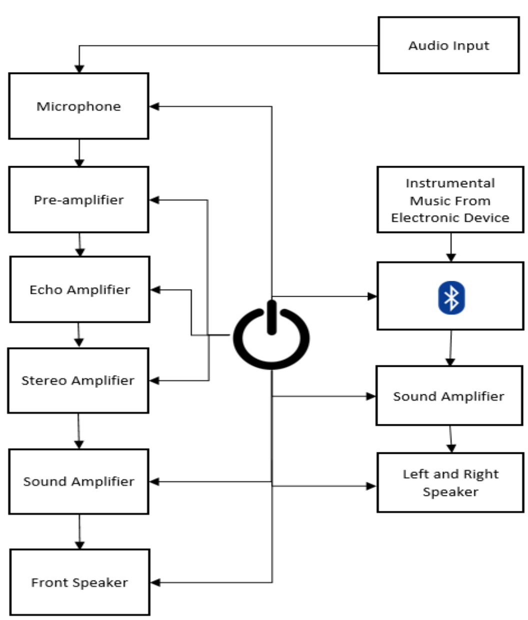

Figure 1. Block Diagram Presenting Major Hardware Components.

There are two inputs going to the system: analog audio signal from physical voice and digital signal from any electronic device. First, the analog signal goes to the electret condenser microphone. Since the signal magnitude is so small, it then goes to the pre-amplifier circuit. Echo and Stereo amplifiers would filter the analog signal and make it sound more professional. However, the signal is still very low; hence, it will be amplified one more time before projecting through the front speaker. Secondly, the Bluetooth module connects to an electronic device. The digital signal converts to analog signal by the module. The magnitude of the signal then increases by a sound amplifier. Finally, left and right speakers will project the music.

IV. Hardware Design

Each of the major system components outlined in section II, System Components will now be described in detail.

A. *Echo Amplifier (PT2399)*

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 4.5V-5.5V 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.

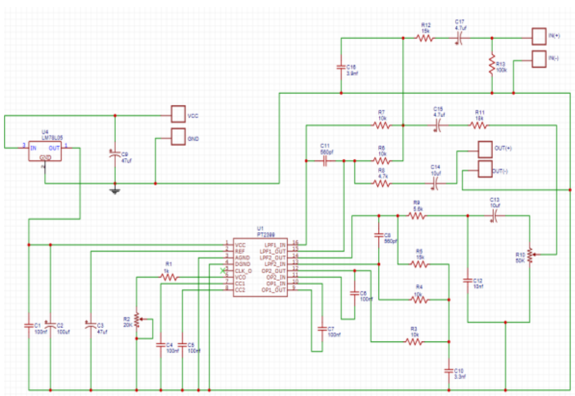

Figure 2: PT2399 Echo Amplifier Circuit Diagram

B. Volume Amplifier (LM386)

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 pre amplifier for the microphone, and the second one is used to amplify 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.

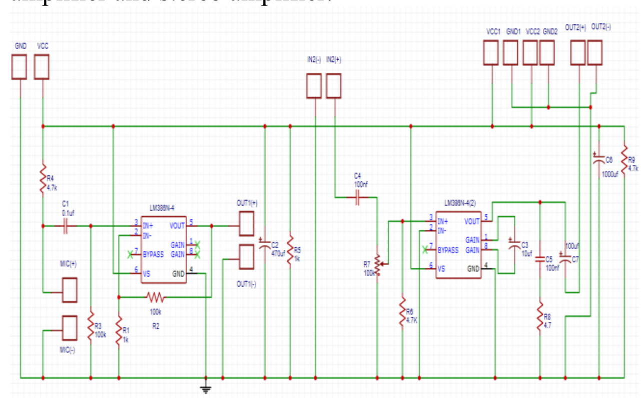

Figure 3: LM386 Volume Amplifier Circuit Diagram

C. Bluetooth Module:

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.

D. Battery:

The 9800mAh 12 volts Lithium-Ion battery require a circuit to charge. IC LM3622 is a powerful chip that could make a 12.6V, 4A Li-Ion battery charger circuit.

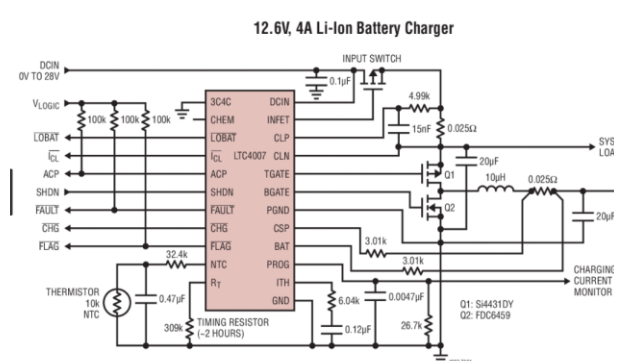

Figure 4. LM3622 Battery Charger Circuit.

The following graphs shows the charging voltages and the capacity over time of the battery:

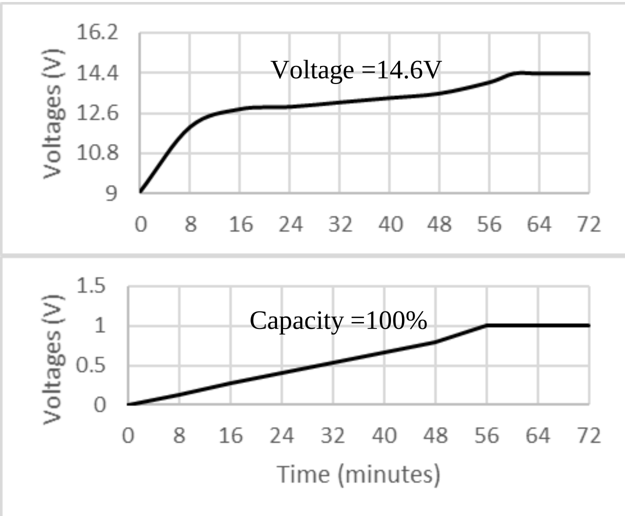

Figure 5. Charging characteristic of 9800mAh 12V battery

Some important values in charging battery are:

1. Input Voltage: 20V
2. Charge Voltage: 14.6V
3. Charging Current: 10000 mA (1 Charge)

By observing both graphs, the battery is fully charge in an hour and reach 14.6V.

E. Stereo Amplifier:

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.

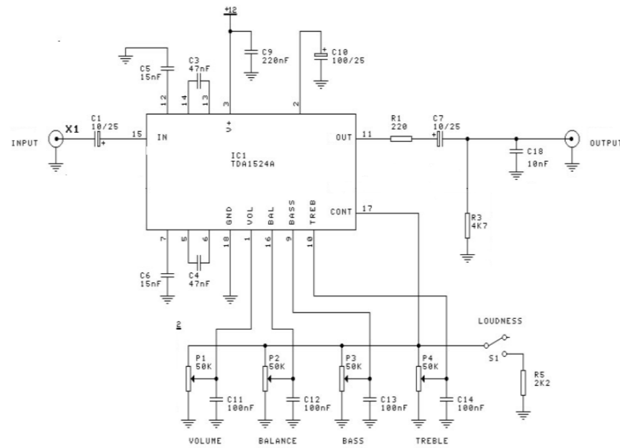

Figure 6. TDA 1524A Stereo Amplifier Circuit with one input and one output.

IV. LED DISPLAY

The LED 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 system. Not only will the Portable system allow the singer to have the ability to hear their favorite songs but perform to their favorite songs while an arrangement of lights display the performance for an exciting and interesting show for all the enticed viewers watching.

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.

V. 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. Below in the figure 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

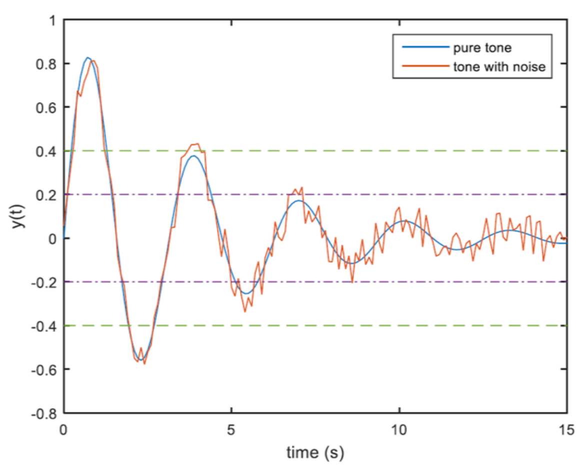

Figure 7. Simulation of decaying audio signal with and without noise. Dashed lines representing the threshold on the signal.

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

$$T(\text{Period}) = 1/f\,(\text{frequency})$$

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.

To achieve optimal display capturing all the octaves was crucial. An octave is the interval between one musical pitch and another. The absolute pitch standard is defined so that all musical Cs are integer powers of 2. Therefore, by doubling the frequency all the notes and octaves were captured. The process is accomplished by using the following calls:

```
Void pitch (float frequency)
     Float freq ()
     Int reading_adc()
        void musicalNotes(RGB color, mask[][] )
   void turnoff()
     void scrollText(String textToDisplay)
```

Figure 8. Nested function calls path upon system "hearing" a musical note.

Table 1 lists the note frequencies starting from C4 at 261.2 Hz and moving to the next octave by doubling the frequency. The range used in the algorithm was starting from two up to seven. This encapsulates the human voice range. The various functions used in the system were fundamental to achieving the correct output. The freq() function, was vital in taking the readings from the reading_adc() function and making the appropriate calculations to obtain the period over a signal. The musicalNotes() would map the LEDs to the respective note by using a 2D mask array. The next function would turn off the LEDs if needed.

	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

Table 1. Summary of Note Frequency (Sample)

VI. Case Design

The purpose is to build the box that can display all the components of the system. Acrylic sheets are the best material option. First, it can be easily laser cut by TI lab by using AutoCAD to draw the dimension and holes. Second, acrylic sheets could be assembly easily by acrylic glue and hardly break. Finally, the committees could see all the PCBs, components, and wires inside the box.

There are three small boxes for the speakers which will places on the bottom sheet. The battery also mounted on the bottom sheet which makes the system has strong foundation. Next, the PCBs are mounted on the speakers' box and the potentiometers are placed on the top sheet. The top and front speakers connect with two clamps so it can open easily for maintenance. The goose neck microphone cover is used to cover the microphone. It delivers the flexibility of the user. The LED display is placed in front of the box and connect to the top sheet.

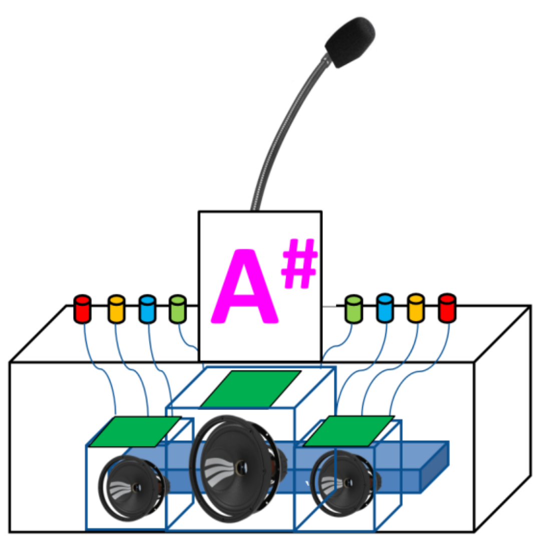

Figure 8. The Animation of the Karaoke Portable System

VII. PCB Board Design

The complete system, with the obvious the karaoke system and LEDs display, is implemented on a dual-planed perforated prototyping connection board. This is big enough to allow for complex wires and components to connect on the PCB boards. There are three through-hole components for our karaoke system and two surface mounted components for the LED display system. The reasons that this project has five PCB boars are to fix them easier. Each PCB board has its own function, so testing and fixing each PCB are more reliable.

The design of PCB is a highly-efficient method of prototyping with Eagle-cad and EasyEAD software, and one that allowed for this system to maintain relatively low-noise operation. All in all, the PBC boards design meets the highest accurate in dimensions and good quality to reduce noise and produce and good quality of music.

VIII. Safety and Standards

Throughout this project, the safety and standard are very important. First, any heat dissipation from our power PCB would cause potential injury to the user, so using a relatively small voltage to accomplish the goals for both the electret microphones and the overall run-time of the design is a solution. Second, all potentially between the connection in wires have been wrapped with heat shrink to prevent any shock for the user. All wires that run from the microphone and battery to the power PCB have been collected and wrapped in heat shrink as to limit the amount of separate wires in the project.

IX. Conclusion

Overall this system was design for the user to sing and enjoy the music in everywhere. The product is used to be more amusing in music. The product is made in high quality of acrylic sheets, and all the components and PCB are meet the safety and industry standards. The karaoke system is included the microphone, echo, bass, treble, and volume control is working perfectly with very low noises. The music is played from Bluetooth device that work perfectly. Future work in the design would consist of a full product prototype, as well as making the smaller to be more portable.

THE ENGINEERS

Lam Dinh is a 24 -year old graduating Electrical Engineering student who is taking a job with Duke Energy in Lake Mary, FL, as a transmission substation engineer, specializing in grounding analysis, substation design, and protection scheme design for high quality and reliability performance of substation systems.

Tuan Dao is a 28 -year old graduating Electrical Engineering student. Tuan hopes to pursue master degree in electrical engineering after working for a few years to obtain the experience. He specializes in area of sound engineering: loudspeaker and amplifier design, working for company such as Beats by Dre, Bose, JBL, Sony, or Samsung.

Jennifer Franco is a Computer and Electrical Engineer, she hopes to pursue fields in the software industry. Jennifer worked on the design of the LED display and programming of the Karaoke Portable System. Upon graduation she has been giving the offer to work as a Substation engineer for Duke Energy.

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