Emergency Vehicle Detection System

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***Abstract* — A system designed to interface with a preexisting audio system, that provides an automobile driver with advanced warning of an oncoming emergency vehicles. Using an array of microphones integrated onto the vehicle, the system will allow the driver to move to the safest shoulder of the road for both themselves and the emergency personnel. By analyzing the changes in frequency and comparing the fluctuation in amplitude at each microphone, the system will be able to both identify and locate an emergency vehicle. Once detected the system will then alert the driver by interrupting the audio system and playing a warning response.**

***Index Terms* — Analog-digital integrated circuits, emergency services, logic programming, microphone arrays, and microprocessor**

I. Introduction

With the technological advancement of automobile's audio systems and soundproofing technology in cabins, emergency vehicles have had a more difficult time getting driver's attention. This has caused an increase in accidents involving Emergency vehicles, especially at wide intersections. The Emergency Vehicle Detection System is an automated system installed on the roof cars/trucks that alerts the driver of the presence of an Emergency vehicle.

The system uses an array of four cardioid microphones and frequency filtering to detect the sirens produced by the oncoming ambulance, fire truck or police car. By using four microphones, the system has sufficient data input to extrapolate the location of the Emergency vehicle relative to the car with an accuracy of +- 45 degrees. This location process is done by analyzing the intensity of sound at each microphone simultaneously. This analysis begins by filtering out the noise detected by the microphone with a bandpass filter. This filter is constructed of cascaded 2^{nd} order Butterworth high and low pass filters for optimal results.

Once filtered, the sound is then digitized by the dedicated ADC portion of the processor. This chip uses 12 bits at a sample rate of 100 kHz to map the sounds, and create the bit stream that is passed on for the digital signal processing. Using a discrete Fourier transform of the bit stream, the amplitude and frequency of the sound are determined by the processor. Those readings are then compared to the known frequency patterns of the sirens. If a match is detected, the processor compares the amplitude at each microphone and extrapolates the general location of the emergency vehicle. The system will decide if the driver needs to be alerted.

II. System Components

In order to accurately describe this system, it is best broken into a series of components that once integrated together will form the complete operational system. The following sections introduce the components on a semi-technical level

A. Microphones

The sensors of the system are Shure SM58. These are dynamic, cardioid microphones that were chosen for their specific pickup pattern, linear frequency response, and low price point. These characteristics makes them easy to work with and prove the concept.

B. Filter

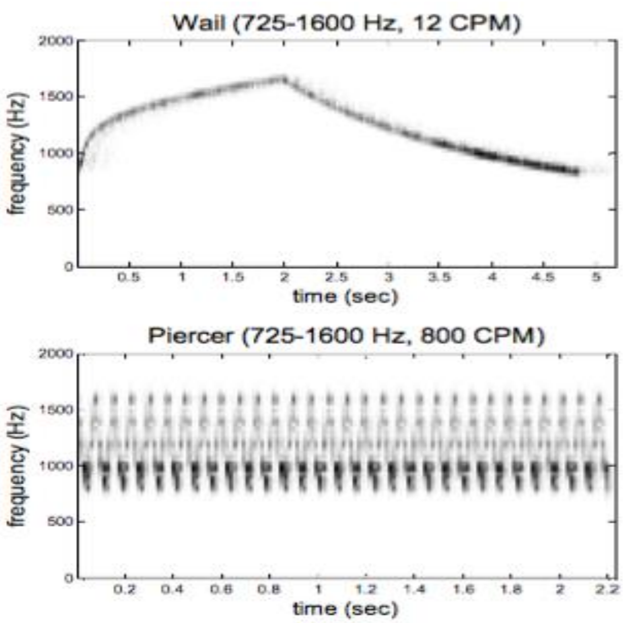
By implementing a filter, unwanted noise will be eliminated at the start of the system. This filter is a 2^{nd} order Butterworth filter, built with TL084 general purpose op-amps. It was tailored to have a bandpass range roughly from 300 to 3000 Hz, a gain of 6943.75, and an offset of 1.5 volts. This will allow our processor to have the clearest signal possible to convert to digital format for processing.

C. Microcontroller

At the center of the system is a TMS320F28377S. This processor was chosen for its ability to handle the multitude of tasks that will be needed. The TMS320F28377S has a dedicated ADC portion that will convert the four analog signals from the microphones to digital format with twelve bit resolution at 100 kHz.

Once converted the processor will use its floating point architecture to perform a discrete Fourier transform of each bit stream in order to find the frequency and amplitude of the sound being detected. These values will be stored 100 at a time.

These 100 stored values are compared two at a time to identify the shift in frequency verse time of the sound. If these changes follow the same pattern as the emergency sirens, the algorithm will set a flag to initialize the warning system. For reference, the patterns of the two crucial sirens in question are provided in Figure 1. These are the most important sirens because one is the fastest changing in frequency vs time and the other is the slowest.



Time vs Frequency Graphs of Sirens

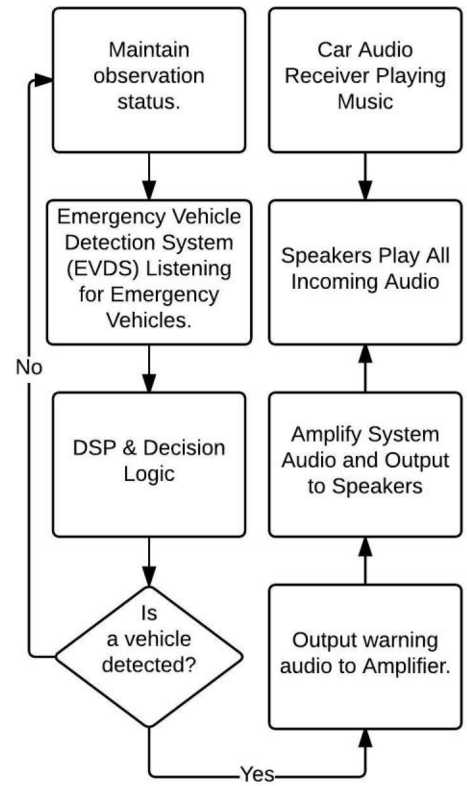
Fig. 1. Frequency vs time graphs of the fastest and slowest changing sirens produced by emergency vehicles. These are used in calculations shown in section V below.

D. Warning System

An off board audio amplifier is use to amplify the DAC generated audio file from the flash memory through the speakers in the vehicle. The Kinter MA-180 is an independent amplifier that will only be used by the EVDS. This will allow operation of the system even if the vehicle's main audio system is powered off. This portion will also have a visual aspect in the form of an LED display. The LEDs will light in specific patterns depending on the location of the siren.

III. System Overview

To better understand the integration of each module described above, a flowchart of the system can be helpful. Figure 2 on is the overview of the EVDS decision process. The flowchart depicts the non-conclusive nature of the system. This will allow for continuous operation while the vehicle is in operation.



Flowchart of system

Fig. 2. Flowchart that outlines the steps taken by system

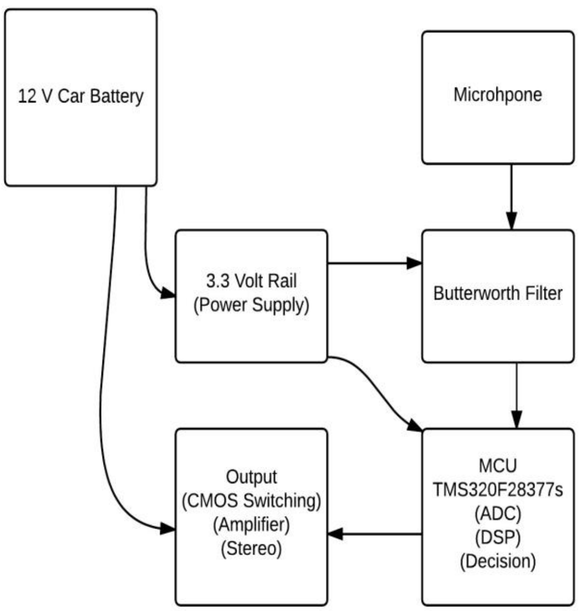
Figure 3 shows the order in which each of the major modules are connected to create the EVDS. As seen in the flowchart, the system will be powered by the existing battery, but the 12 volts will be regulated and dropped to 3.3 volts to compensate for the processors power requirements.

IV. System Hardware

Now that the system can be understood at the macro-level, the hardware used to create it can be described in detail. The following sections take each module and describes its construction in detail.

A. Microphones

The sensor used in this project is the Shure SM58. This is a dynamic microphone that has a sensitivity of –54.5 dBV/Pa (1.85 mV) and a frequency response from 50 to 15000 Hz. This microphone works by converting positive pressure on the diaphragm to positive voltage on pin 2 with respect to pin 3. A 3.5mm aux converter is used as the conduit between the microphone and filter. It should be noted that these microphones were chosen because of their availability and affordability. If this system was to be implemented during the manufacturing of the vehicle, multiple (more than 4) MEMS microphone could be built directly into the frame and body.



Interconnections of the different modules

Fig. 3. Flowchart depicting how the different modules of the emergency vehicle detection system are connected and powered

B. Filter

The bandpass filter is comprised of two 2^{nd} order Butterworth filters, one high pass and one low pass, made with TL084 general purpose op-amps and 5% tolerance resistors. The op-amps are operated with +/- 3.3 volts supplied to the respective VCC's. Using two 470 Ohm offset resistors at the end of the low pass filter entering the ADC module, the offset of the output has been tailored to be 1.5 volts. This will allow for the ADC portion of the processor to more accurately convert the signal. Because of how the processor is designed, the output of the filter needed to be purely positive to ensure no data is lost. Also, prior to the filter, a pre-amplification is done to amplify the micro-volt size signal produced by the microphone to useable values in the millivolts rage. The equations used to tune the high and low pass filter are as follows:

$$K = \frac{R_A}{R_B} + 1 \quad (1)$$

$$\frac{1}{Q} = K - 1 \quad (2)$$

$$\omega^2 = \frac{1}{R*R*C*C} \quad (3)$$

$$B_n(S) = S^2 + 1.4142s + 1 \quad (4)$$

Using equations (1) – (4) and the values provided in table 1, the gain, K, of the high pass filter is roughly 2.5; making the quality factor, Q, 1.5. The cut off frequency, ω, is calculated to be 338.63 Hertz. Recalculating these values for the low pass filter yields a gain of 2.5, quality factor of 1.5 and a cut off frequency of 3386.27. Cascading these filters, with the high pass coming first, produces the necessary bandwidth of 3047.64. A higher order filter could be implemented for faster roll off rates. However at this stage of design, the second order filter and its roll of rate of -40dB/decade were considered to be sufficient.

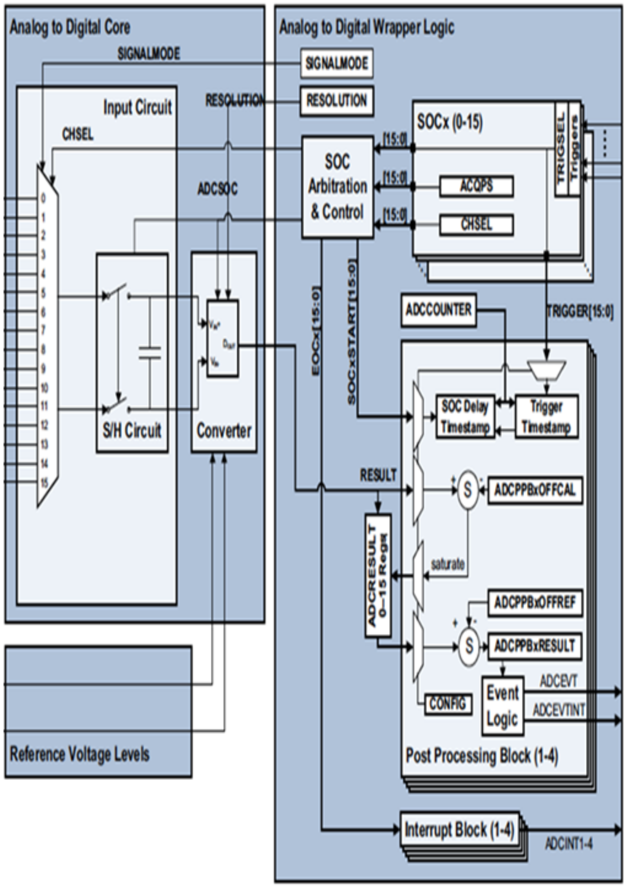
TABLE 1

VALUES USED IN BUTTERWORTH FILTER

Variable	Value
R_A	1500 ohm
R_B	1000 ohm
R	470
$C_{Lowpass}$	.1 micro-Farad
$C_{Highpass}$	1 micro- Farad

C. Analog to Digital Conversion

The ADC portion of the microcontroller runs at a sampling rate of 100 kHz and uses 12-bit resolution. This sampling rate is well over the Nyquist rate of 6 kHz. That rate was calculated using 3 kHz, which is the highest frequency sine wave produced by emergency vehicle sirens. The schematic for the microcontrollers built in ADC module of the TMS320F28377S is shown in figure 4. In the figure, it can be seen that the processor has multiple ADC inputs, which will be used for the different microphones. These signals are run through a sample and hold circuit to prepare for conversion. Once converted, the output of the converter runs into the "ADCResult" register coinciding with the input and passed to the post processing blocks of the processor. It should be noted that at the conversion stage, the signal is compared to the reference levels. These levels are 3.3 volts and 0 volts, dictating the 3.3 volt operation of the op-amps mentioned before.

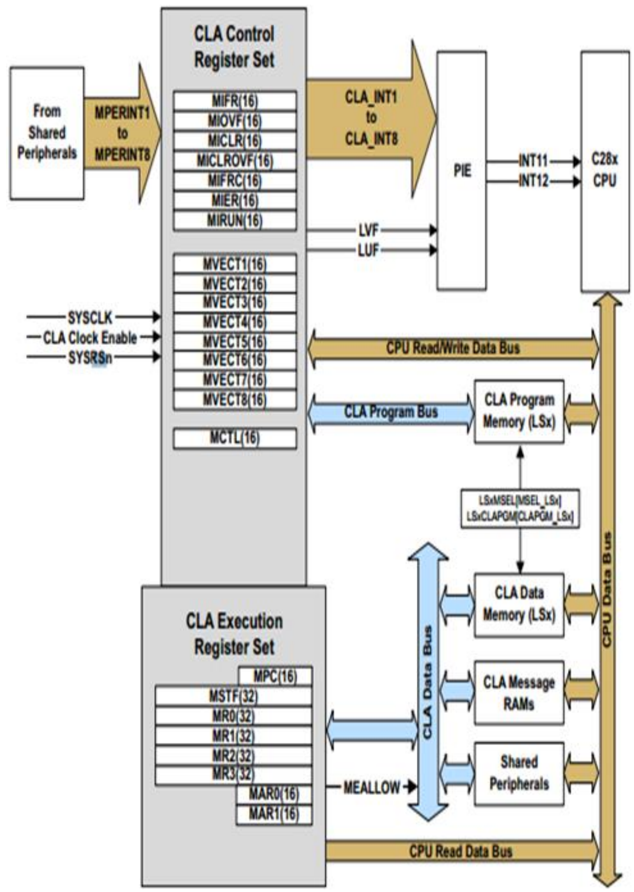


Block diagram of ADC module in processor

Fig. 4. A block diagram showing the workings of the ADC module including data buses, outputs, and reference voltage levels.

D. Floating Point Unit

Once the analog to digital conversion is complete, the data that is stored in the "ADCResults" registers will be ready for processing. The floating point unit in the microprocessor will be responsible for performing a Fourier transform on the bit stream from each microphone. The block diagram of the microprocessors interconnections is shown in figure 5. The floating point processor in the TMS320F2837S is known as the Control Law Accelerator. It is an independent single-precision floating-point unit. With its 32 bit design, it can handle up to eight individual CLA tasks. These tasks originate in peripherals, in this project those being ADC. Each of the incoming ADC streams are taken in by the CLA to perform a Fourier transform. Once one stream is done, the next steam is taken through the same process. This is all done at the processors clock rate of 200 MHz and independently from the main CPU.



Floating Point Unit Block Diagram

Fig. 5. A block diagram of the inner working and data flow inside the floating point unit of the processor.

E. Warning System

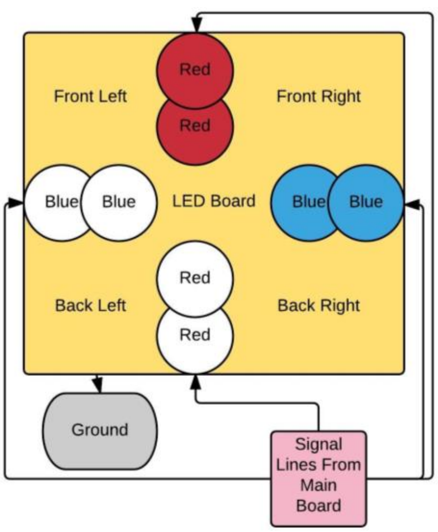
The final portion of the hardware in the system is the external amplifier and because it is an off the shelf unit it has not been described in detail here. Some essential information is provided in table 1 below.

The audio notification produced by the microcontroller when a siren is detected needs to be amplified in order to play the three tones over the speakers in the vehicle. The additional amplifier level will be set higher than the amplifier used to power the vehicle's audio center. This will allow the tones to overpower any sound already coming through the speakers. Implementing this design will offer the best listening experience for the passengers, while still warning them about the emergency vehicle. The user selected audio will continue with the alert overlaid, offering the best warning with minimal interruption.

TABLE 1.
IMPORTANT INFORMATION ABOUT EXTERNAL AMPLIFIER

Specification	**Value**
Power Output	2*20 watts RMS
Input Impedance	47k Ohms
Frequency response	20 Hz- 20 kHz
Input Sensitivity	200 mV
Operating Voltage	Up to 14.6 Volts

To accompany the auditory warning, an array of LEDs will be built to have a visual warning. These LEDs are oriented in a square pattern. They will illuminate based on which microphone is being triggered. For example, as seen in figure 6, the front and right lights are lit up. This would signal the driver that the siren is in the front right quadrant of the vehicle.



Led Display

Fig. 6. Image showing the directional output of the warning system

V. SYSTEM SOFTWARE

Before the software portion of the system can be described in detail, the overall objective of the system must be complete understood. The success of the project is determined by the system's ability to determine if the sound inputted into the microphones is a siren. This is done by comparing the frequency patters of the sirens to the sound being analyzed.

Because the system is continuously running, the algorithm will take a frequency sample and compare it to the next sample taken. The algorithm is looking to see that the change in frequency is at the correct rate to call the sound a siren. Based on the calculations done in equations (1) and (2), the frequency should change more than 300 Hz but less than 7500 Hz in one second.

Slowest Rate of Frequency Change

$$Min = 1400\ Hz$$

$$Max = 1700\ Hz$$

$$1\frac{sec}{cycle}$$

$$\frac{1700 - 1400}{1} = 300\frac{Hz}{sec} \quad (1)$$

Fastest Rate of Frequency Change

$$Min = 750\ Hz$$

$$Max = 1500\ Hz$$

$$.1\frac{sec}{cycle}$$

$$\frac{(1500 - 750)}{.1} = 7{,}500\ Hz/sec \quad (2)$$

A. Analog to Digital Conversion

Now that the goal of the software is defined, the first step in the coding is to perform the analog to digital conversion. The input pins of the processor are initialized and giving each a sample window of 15 system clock cycles. The pulse width module is used to trigger a sample on the rising edge at a rate of 100 kHz. Each microphone is feeding an "ADCResult" register filling a 511 bit converted component array. At every rising edge, an interrupt is triggered that fills one bit of the buffer. When the buffer is full, the Fourier transform is done one that buffer. Each of the inputs are handled systematically one at a time as each buffer fills.

B. Fourier Transform

As stated before, the Fourier Transform of each buffer is done as it fills. This process is handle by the CLA and not the CPU. The ADC peripheral has built in connections for data transfer from the CPU to the CLA, making the coding less complicated. Texas Instruments also provides libraries with real time Fourier transform equations. This algorithm takes the information in the 511 bit buffer and copies it to RFFTin buffer. Using this buffer and the given TI function, the Fourier Transform is calculated. Multiple pointers allow for many buffers to be filled. These include a magnitude buffer, a phase buffer, and a twiddle factor buffer. The algorithm used in this project is concerned with the magnitude buffer. By scanning through this buffer and finding the bin with the highest magnitude, and multiplying it by 195 (the ADC sample rate divided by the buffer size). This calculation will convert the bin number to a frequency value, thereby finding the fundamental frequency of the sound in question.

C. Decision Logic

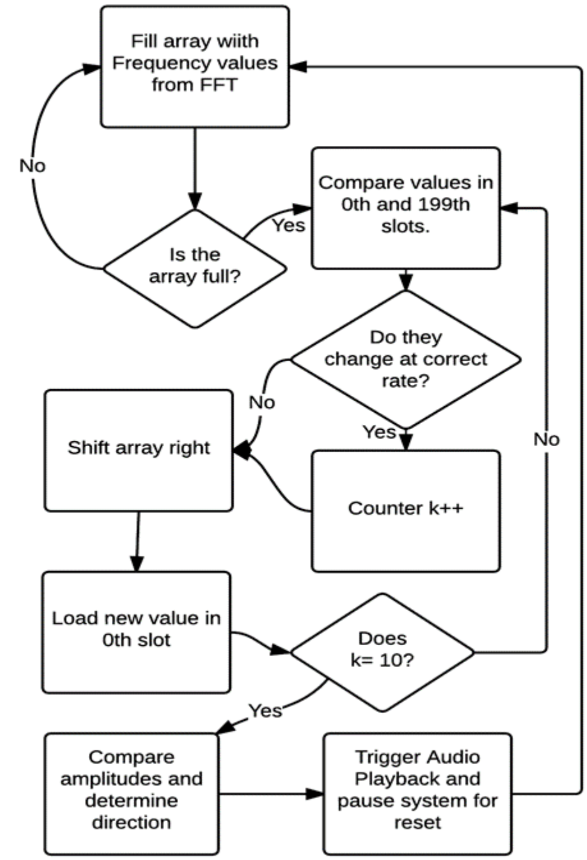
The frequency values calculated are stored in arrays titled “Freq_A#” with the number corresponding to the microphone number. These arrays hold 200 values and there is an initial delay to populate all the slots. Once slot 199 is populated, slots 0 and 199 are compared. Then a logical shift to the right is performed on the array. The new value in slot 199 is compared to the new value in 0. If the two values are equal or change to quickly with respect to time to be a siren, the system keeps listening. However, if ten values follow the correct pattern in a row, the system will set a flag that a siren is being detected. The trigger value can be tuned for each microphone to adjust sensitivity. Below in figure 6 is a block diagram for this process for better understanding.

VI. System Implantation

This section discusses the physical layout of the system as well as how the hardware was built and configured.

A. Microphones

Because this system needs to detect locations of sounds, the orientation of the microphones is crucial to the design. Each of the four microphones has a cardioid pickup



pattern with significant roll off outside 60 degrees from center. This makes the pickup range of each microphone

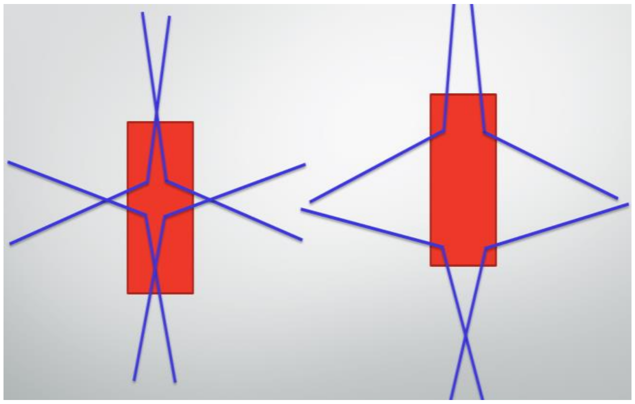
Decision Matrix

Fig. 6. A block diagram of the inner working and data flow inside the floating point unit of the processor

roughly 120 degrees. This allows for overlap in each of the microphones pickup patterns. The amount of overlap on each microphone can be adjusted to fix or create blind spots.

The amount of overlap in the pickup patterns must be addressed and compensated for in the algorithm used for locating the siren. Testing was performed and an ideal layout for the microphones was determined. Some of the orientations tested are shown in figure 7 and the final alignment is depicted in figure 8.

This orientation was chosen because it offers the most accuracy in the regions that the driver is most blind to; those regions being the sides and rear of the vehicle. However, this orientation creates a blind spot in the front of the vehicle. This was determined to be acceptable because the driver will be looking this direction and the average human has a 150 degree field of vision. This orientation will give the driver a theoretical 360 degree field of “vision.”



Different pickup pattern overlaps that were tested

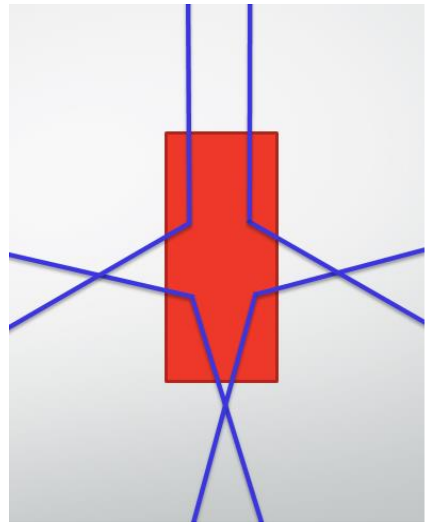
Fig. 7. These images show how different orientations of the microphones produce different overlap patterns.

B. Hardware

In the presented design, the microcontroller and filters are mounted at the center of the microphone array. This was done for testing and debugging purposes. If this design was added to a vehicle at manufacturing, the circuit boards, filters and wiring could all be hidden in the dashboard. A PCB was designed using the developmental board as a reference. By removing several pin headers and the booster pack extensions an initial design was created and printed. As for the filters, they too were designed to be placed on a small PCB that was separate from the main processors board. The separation of the boards was done for testing purposes; these two boards could be combined into one board to save space in the dashboard. The PCBs were printed using a two layer design, and were intended to have surface mount parts ranging from in size from 1/10 0402 to 1/16 1210. Multiple layers could be implemented to make the boards smaller in size and better integrate the two boards together. However, this would have required a more advanced version of EagleCad and would have been more expensive to get printed. Because the size of the board is not a major concern, the larger boards were deemed acceptable for this prototype.

During the population of these boards, several boards were destroyed. These boards were deemed unusable after many failed soldering attempts. Multiple problems occurred on each of the attempts, such as non-wetting solder pads, shorts, and melted silk screens. The group’s inexperience with soldering surface mounted devices was considered the main problem.

Final of



orientation

microphones

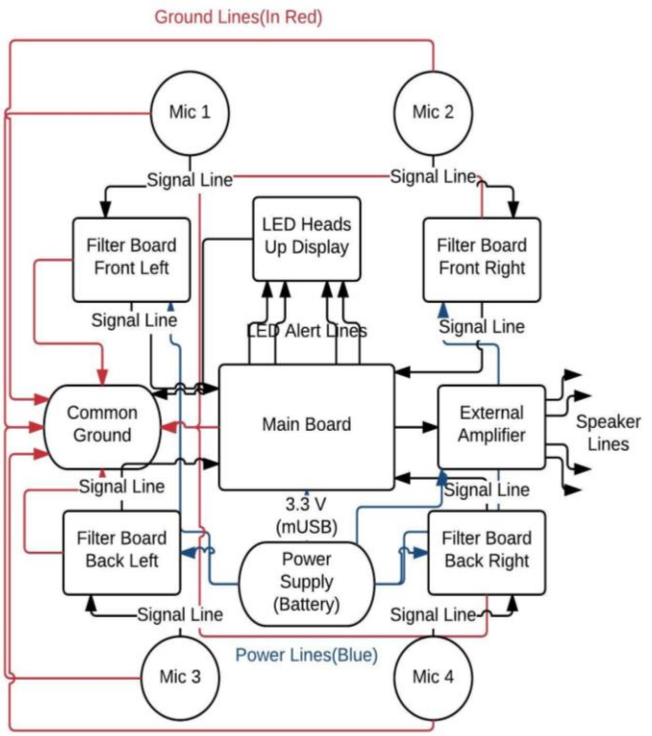
Fig. 8. This is the orientation of the microphones chosen for optimal results.

With no more funding available, and the end of the term fast approaching, the advisor gave permission to use the developmental board in replace of the designed PCB. To fix the problems seen while soldering the filter together, the board was changed from a PCB to a through-hole design. This allowed for larger parts to be used, making the soldering easier. With these changes, the prototype seen was built and tested. Figure 9 is a breakdown of how the components are wired together.

VII. Conclusion

During this two semester long project, the concepts and teachings from all prior classes were used. This project was a great opportunity to combine the information presented in each class into a real world design process. By undertaking and completing the design process, the authors were able to see how complex and unique designs could be created using the fundamentals taught in class. During the production stage of this process, the authors came to see how different real world applications acted in comparison to simulations.

Debugging any issues that arouse was a great lesson in how to test, diagnose, and fix issues. These lessons could not have been taught in a classroom, and have given the authors great respect for design and test engineers.



Wiring Diagram of System

Fig. 9. This is a wiring diagram showing common grounds, power connection and signal lines of the system

The managerial portion of this project was great practice in recording, reporting, and documenting the design and production process, as well as mitigating the numerous failures encountered throughout. These skills are essential to the success of any engineer and will not be forgotten by the authors. In conjunction with these technical skills, this project tested the groups interpersonal and communication skills. The group aspect of this project gave the authors the chance to apply their skills in collaboration and communication.

Overall, this project demonstrated the complex process of creating, designing, and producing an electronic system and the multitude of disciplines needed to accomplish such a task.

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Ryan Chappell is an Electrical Engineering student at the University of Central Florida. After graduation in August 2016, he plans to continue to work with Precision Test Solutions as an Applications Engineer. Ryan also plans to eventually pursue his Masters of Business admiration.

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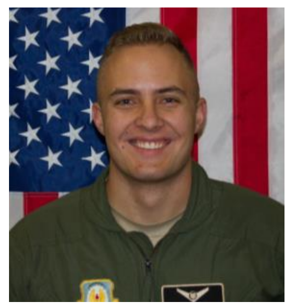

John Fick is Electrical Engineering Student at the University of Central Florida. He will graduate in May 2016 with his Bachelors of Science and a minor in Aerospace Studies. Upon graduation he will be attending Undergraduate Pilot Training for the United States Air Force. John plans to one day earn his Masters of Computer Science or Electrical Engineering.

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