Emergency Vehicle Detection System

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1.0 Executive Summary

With the technological advancement of cars/trucks stereo systems and soundproofed 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 triangulation process is done by analyzing the intensity of sound at each microphone simultaneously. This analysis begins by filtering out the noise seen by the microphone with a bandpass filter with a pass band in the range of one hundred yards. This filter is constructed using back to back 5th order Butterworth high pass and low pass filters for optimal results.

Once the filtered sound enters the system, the signal is then digitized by the dedicated ADC chips. These chips use 8 bits at a sample rate of 50 kHz to map the sounds, and create the bit stream that is passed on for the digital signal processing. This bit stream is then processed in order to find the amplitude and frequency of the sound. Those readings are then compared to the known frequency patterns of the sirens. If a match is detected, the processor takes the amplitude at each microphone and extrapolates the general location of the emergency vehicle. After the relative location of the emergency vehicle has been established, the system will decide if the driver needs to be alerted. If the emergency vehicle is traveling away from the driver, the system will ignore the input. However, if the driver needs to be alerted, the system will tell the driver what direction to travel or to not cross the intersection.

This warning will be both auditory and visual. One of two LEDs on the driver's dash will blink, giving the driver a direction in which to travel. Also, the EVDS will take control of the vehicle's sound system to mute any music playing and play a prerecorded message.

2.0 Project Description

2.1 Motivation

There are many sources of motivation that have propelled this project forward. The most striking are from our own personal experiences. Since the beginning of the semester we have encountered both on our own and together countless Emergency Vehicles in Orlando and across the country. Despite the apparent volume and appearance of these vehicles, they are seemingly invisible in many situations. This proves to be very problematic when driving around. Especially when you are listening to music, or even just driving with your windows up. With modern innovation creating essentially soundless cabins for new cars, it is becoming increasingly harder and harder for Emergency vehicles to alert drivers of their presence.

These seemingly invisible emergency vehicles are proving to be a safety hazard that has caused a significant number of injuries across the nation. These vehicles are supposed to be protecting and saving the citizens of our great nation, not adding to the body count. In 2012, there were over 100 non-emergency workers killed in accidents involving emergency vehicles. This is not even counting the emergency workers. Additionally, there were over 200 more that were injured as a result of emergency vehicles. These numbers are entirely too high. Accidents, such as the one shown below in Figure 2.1, where not only harm the members of the accident, but also the people that the Emergency Vehicle (EV) was originally on its way to aid, or in this case, the person in critical condition that was already in the back of the Ambulance. Not only are these accidents tragic and unnecessary, they are preventing the emergency response team from doing their job, and requiring more emergency teams to be utilized, when there are already plenty of places that need their presence.

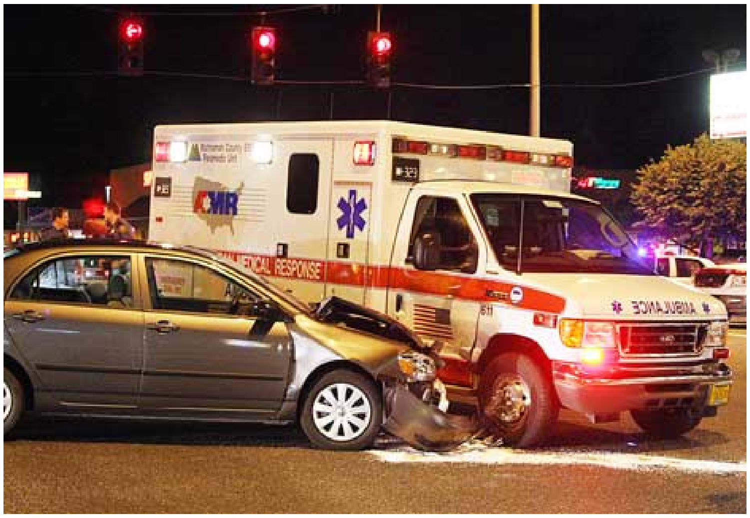


Figure 2.1: *An ambulance carrying a patient in critical condition is struck by an unaware driver in Portland, Oregon.*

The other problem with emergency vehicles, is that even if one driver is aware of their presence, others might not be. In this case, the other oblivious drivers have the potential to react erratically to the driver that is slowing down to make room for the emergency vehicle. This erratic behavior has potential to cause even more accidents. So despite the attempts at being noticed by the public emergency vehicles still cause a significant number of accidents, due to non-uniform recognition of the emergency vehicle.

This touches on another part of our motivation for the project. Human hearing is not nearly good enough to be able to decipher all the things that are going on in the world around. By having a machine that is fine tuned to specifically look for the audio signatures of various sirens, there will be less pressure on the driver to decipher the various sounds around him. Roads are noisy places and with all the commotion is it very hard to hear one distinct sound, even if it is one as distinctive as a siren, and not hearing the siren is one of the main things that makes an emergency vehicle invisible.

Another problem that contributes to the non-recognition of emergency vehicles is the presence of sirens and siren like noises both in music and on the radio. There are numerous radio advertisements and hit songs that feature sirens of some sort. From experience, these sirens sound real enough that they produce one of two reactions. One, is a standard reaction to a siren, look around and start to move over. This can be problematic because, since your car is the only one hearing the siren, you now seem like you are driving erratically. This will eventually lead to an accident. The other is a case of being desensitized to the sound. Meaning that the next time you hear a siren you will not be a quick to move over or even recognize the emergency vehicle. Essentially these radio ads and songs camouflage the true emergency vehicles and aid in their seemingly invisible nature.

This brings us to the major goal of our project. By improving the recognition of emergency vehicles across the board we will be able to positively affect the safety of the roads. Though this system will not be immediately implemented in every car, each one that is implemented is a step toward safer roads and highways. Our system would provide a second set of ears for the driver that will serve to help decipher the increasingly noisier roads and transmit the message to an increasingly quieter cabin.

2.2 Requirements and Specifications

The emergency vehicle detection unit is going to be held to high standards for its specifications. The goal is to have a system capable of detecting a siren up to 50 yards away with and accuracy of +/- 45 degrees. This will allow the system to choose the correct quadrant the sound is in, giving the drive sufficient information. As in any experiment, the team would like to have as few false positive responses as possible, with no false negatives. Those requirements will determine if the system can differentiate between a siren and other random sounds on the road. The system will be clocked to sample the sounds at 50k samples a second, and should be able to give a response to the driver within 3 seconds of the sound being detected.

There are no limiting constraints from standards that will constrain the size of the system on top of the car. The only standard that applies to our system is the national emergency vehicle siren standard. This will not alter the physical design of the system, only change circuit elements for filter elements.

2.3 Goals and Objectives

With these requirements, it is the goal of our team to decrease the number of accident involving emergency vehicles. By drastically increasing the driver's situational awareness the emergency vehicle detection system will help the emergency personal travel in more direct and safer traffic lanes. It will also help the driver detect the emergency vehicle when entering a blind intersection, which is where most accidents occur.

2.4 Design Constraints

The size of our system will be constrained by the size of the microphones. This will determine the height, width, and length of the system. The PCB board that contains the intended processors will not have a constraint on the design because it will be comfortably located in a casing between the four microphones. The Butterworth filter design will be constrained by the frequency range of the four emergency vehicle sirens. Since the emergency vehicle detection system is powered by a car battery, all of the electronics will be constrained to be compatible with a 12V battery source. This will not be an issue because most micro-electronics run off voltages far less than this. The mounting of the system will be constrained by the need to attach to a car. This will lead to the need to arrange the microphones in a compact manner.

3.0 Research

3.1 Microphones

3.1.1 Array

Determining the layout of the microphones on the vehicle is the starting point of the emergency vehicle detection system. There are several points to consider when finalizing the layout that will affect the rest of the design. These elements include the number of microphones being used, the amount of overlap desired in the pickup pattern of each microphone, and the physical placement on the car.

The number of microphones in the system will determine the steps taken during the decision logic for the playback. This application must use three or more microphones, as the use of less than three microphones will not give the system enough information to triangulate the origin of the sound wave to the desired degree of accuracy. However, the use of more than four microphones would be unnecessary and an inefficient allocation of funds. Therefore, the choice is between a triangular or a square pattern for the physical layout of the microphones, each of which has unique advantages and disadvantages.

3.1.2 Array/placement

The triangular pattern with three microphones will minimize the cost of microphones, however will not have the precision of a square pattern with four. Because the emergency vehicle detection system requires a higher degree of precision to locate the origin of the sound, this design utilizes the square pattern. This will allow the Emergency vehicle detection system to determine the location of the siren in a 360 degree range with an accuracy of +- 45 degrees.

Referencing Figure 3.1, shown below, it can be seen that the pickup range of three microphones is a full 360 degrees. However, with the additional microphone, the square array has an extra 120 degrees involved in the system. This addition allows for overlap of each microphone, which will be crucial to the directionality of the system.

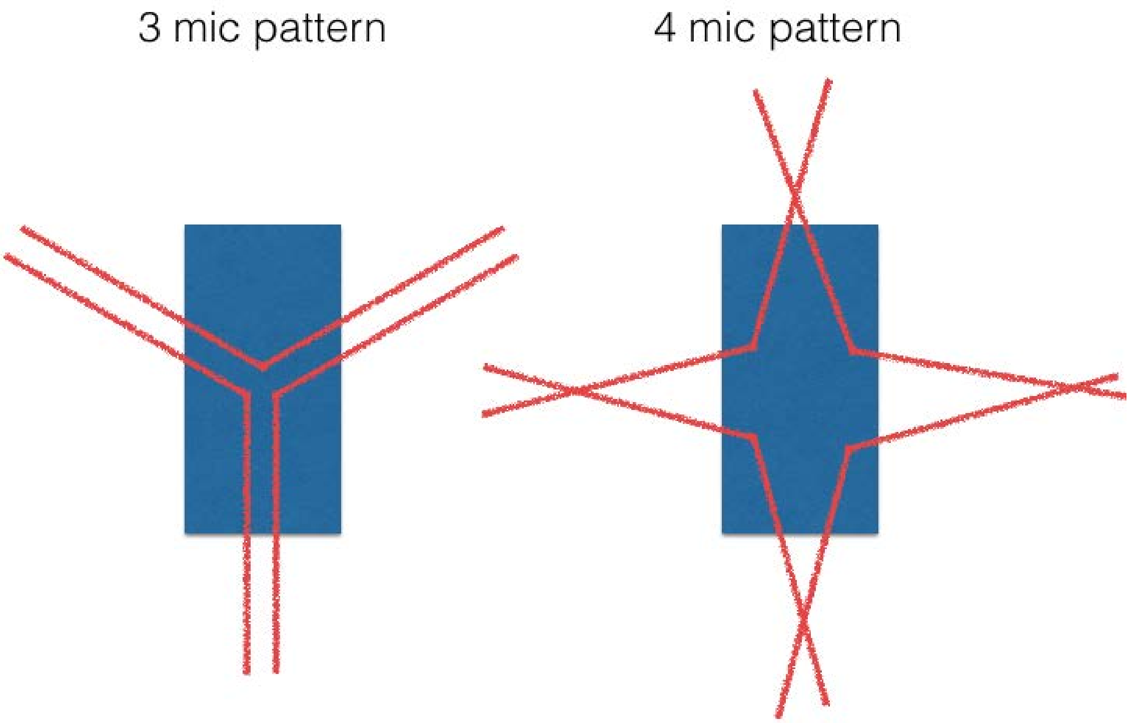


Figure 3.1: *The figure to the left is an example of a three microphone design of the emergency vehicle detection system. The figure of the four mic system is depicted on the right.*

3.1.3 Pickup patterns/overlap

In able for the Emergency vehicle detection system to pinpoint the location of the siren, the system must have multiple inputs simultaneously to triangulate the position. The steps taken in the software to analyze the aptitude of the sound at each point will be discussed later. However, the type of pickup pattern and amount of overlap in each microphone's pickup pattern must be finalized before the code is written. These two elements influence the type of microphones being used and where those microphones will be arranged on the car.

The Emergency vehicle detection system will have four microphones in square, with the diaphragms pointing outward. Ideally, this set up will create a kind of blind spot in the center of the microphones. By creating this void in the center of the microphones, the system will be better able to triangulate the origin of the sound in question. The void will insure the system will always have one microphone that has no input, as it can’t hear the sound being analyzed. This concept will help with the triangulation of the sound by cutting the angle of view by 180 degrees without doing any in-depth amplitude comparison. Cutting the one microphone out of the equation, the systems decision logic can be done more quickly by streamlining some of the amplitude calculations. This concept can be visualized in figure 3.2.

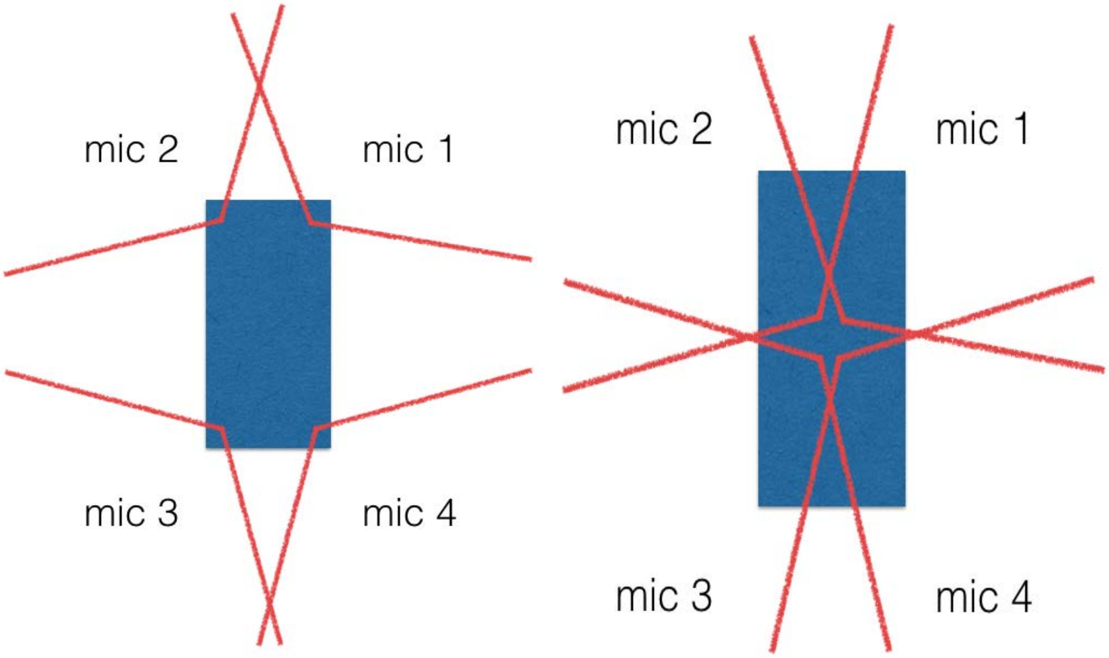


Figure 3.2: *The two figures above show how the placement of the microphones will drastically affect the size and shape of the void in the center. The figure on the left shows the microphones at the corners of the vehicle. The right figure shows the microphones very close together. Each still offers the opportunity for one microphone to read no input to help with calculations.*

These amplitude calculations stem from the amount of overlap in each microphone's pickup pattern. Overlapping the pattern will allow for an algorithm that compares the intensity of the sound at each microphone. The microphone that is at the point of the triangle facing the sound will have the highest intensity, while the two microphones on the sides will have slightly lower intensities. As these two intensities vary, the triangulation algorithm can determine the location of the sound even further than it has already with the zero input of the fourth microphone.

The pickup pattern on the microphones is the most significant design element in the entire system. Implementing the optimal pickup pattern will give the system its ability to determine direction. There are seven standard patterns to consider. These are shown in Figure 3.3 below. Of these seven patterns, three can be eliminated as possible choices the emergency vehicle detection system. The omnidirectional, bidirectional, and sub-cardioid pattern do not offer enough directionality to create the desired blind spot in the center of the square.

Eliminating these three as options leaves four polar patterns to consider. To narrow further, the shotgun pattern is designed for picking up sound that is only directly in front of the diaphragm. The small angle of acceptance in the pickup pattern does not allow for any overlap. This eliminates the shotgun microphone as an option.

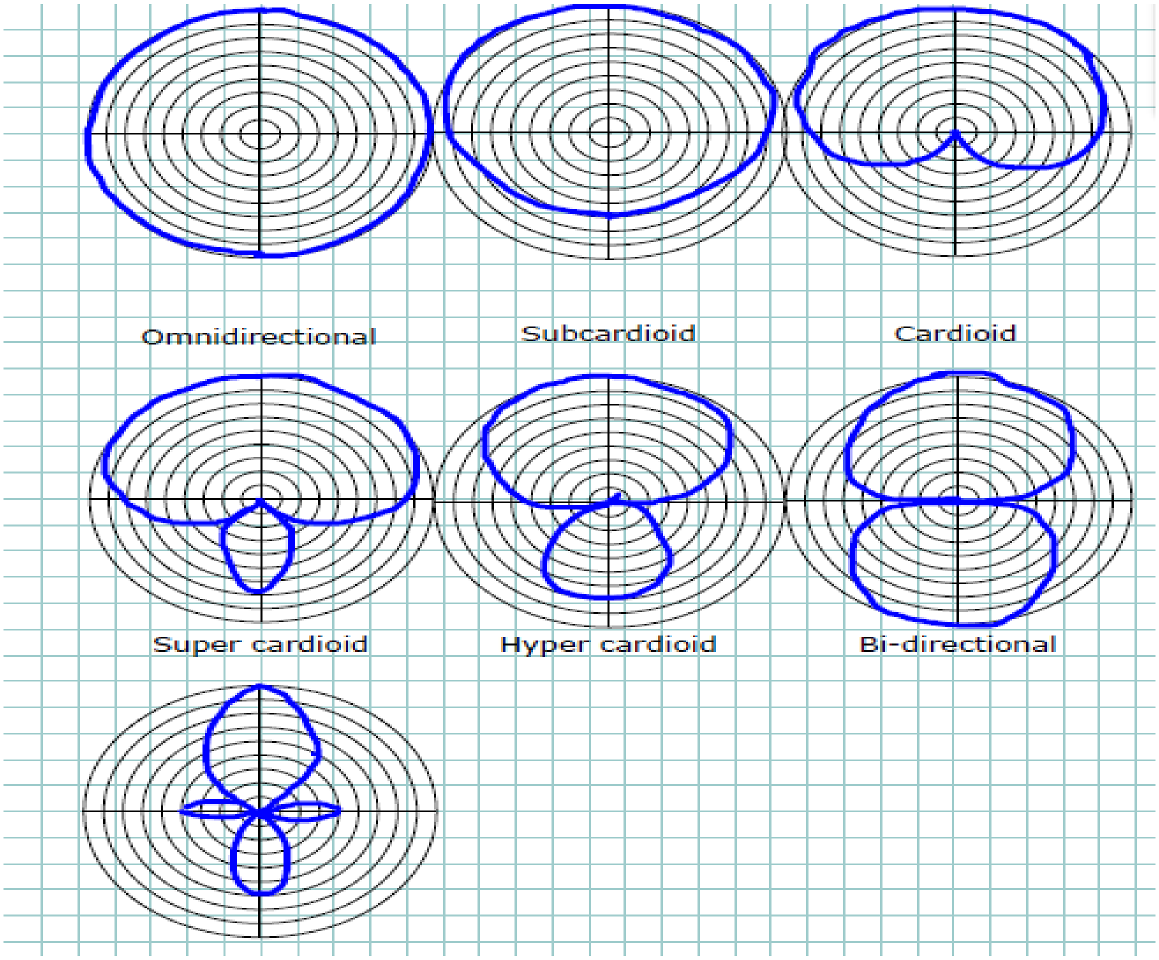


Figure 3.3: *This figure depicts the different shapes of each pickup pattern considered for the emergency vehicle detection system.*

This leaves the hyper cardioid, super cardioid, and the cardioid polar patterns. These three options have the best directional capabilities and will all create the desired void in the middle of the array. To decide between the three, further analysis of their pickup patterns is necessary. The hyper-cardioid and the super-cardioid both have angle of acceptance in the range of one hundred to one hundred fifteen degrees. While the angle of acceptance for the cardioid pattern is in the range of one hundred twenty to one hundred thirty degrees. If the sound is produced outside this angle of acceptance, the microphone sensitivity is greatly reduced. This means using the larger angle of acceptance offered by the cardioid microphone will not only offer the most overlap in the pickup patterns, but also give the most reliable reading if the sound is produced between sixty and ninety degrees from the center of the microphone.

The one hundred twenty degree pick up range of the cardioid microphone has significant roll off at each of the extremes, +- 60 from the center of the microphone. The amount of roll off at these points is unique to each microphone. In some cases, moving just past the sixty degree mark can cause a 5 dB loss. Those losses will change not only change based on the angle of detection, but also based on the frequency of the sound being detected. This frequency dependency gives the cardioid microphone is symmetry along the microphones axis, and must be accounted for both in the selection of the microphone and the coding. These variations in intensity are the key to using the microphones to triangulate the origin of the sound. The algorithm discussed later will account for the variation in intensity based on both the angle and frequency.

3.1.4 Frequency Response

When discussing the frequency of the sound in question, one must also consider the frequency response of the microphone. There are two major types of responses to consider, a flat and a tailored response. The choice between these is based on the intended application and the allocated budget.

If the microphone has a tailored response, there is a range of frequencies that cause a built in gain, regardless of the angle of intersection. These microphones are used primary for vocalist who will want the range of their voice to be custom fitted with the microphone. The emergency vehicle detection system could utilize a tailored frequency response, however, the desired budget did not permit. If the budget was of no concern, the microphones could be designed to have built in gain in the range of the sirens, that being roughly 500-3000 Hz. This amplification would help filter the noise around the microphone by making the sound of the siren have a larger response. This feature can be added later simply by using custom designed microphones. These microphones would also have a greater range for detecting the siren. The built in dB gain for the range of siren would allow for much less intense sounds, meaning the emergency vehicle is further away, to be detected.

Eliminating the tailored response as an option, leaves the flat frequency response as the only option. This means that there is mostly a uniform response to all frequencies within the range on the microphone. However, this “uniformity” is not always perfect and is different for each microphone. For the emergency vehicle detection system, the microphones will need to have a consistent and uniform frequency response in the 500-300 Hz range. Any variation in frequency response in this range will need to be accounted for in the triangulation algorithm. Therefore, it is best to simply eliminate the variable all together and choose a microphone with a very flat frequency response.

3.1.5 Types

After determining the desired polar pattern and frequency response, the last variable when choosing a microphone is the diaphragm design. The options in this category are either a dynamic or a condenser microphone. Each type uses the same principles to convert the sound waves to electrical signals, however, the two microphones use different transducer designs. These two designs are meant for different applications. Figures 3.4 and Figure 3.5 below show the different transducer designs.

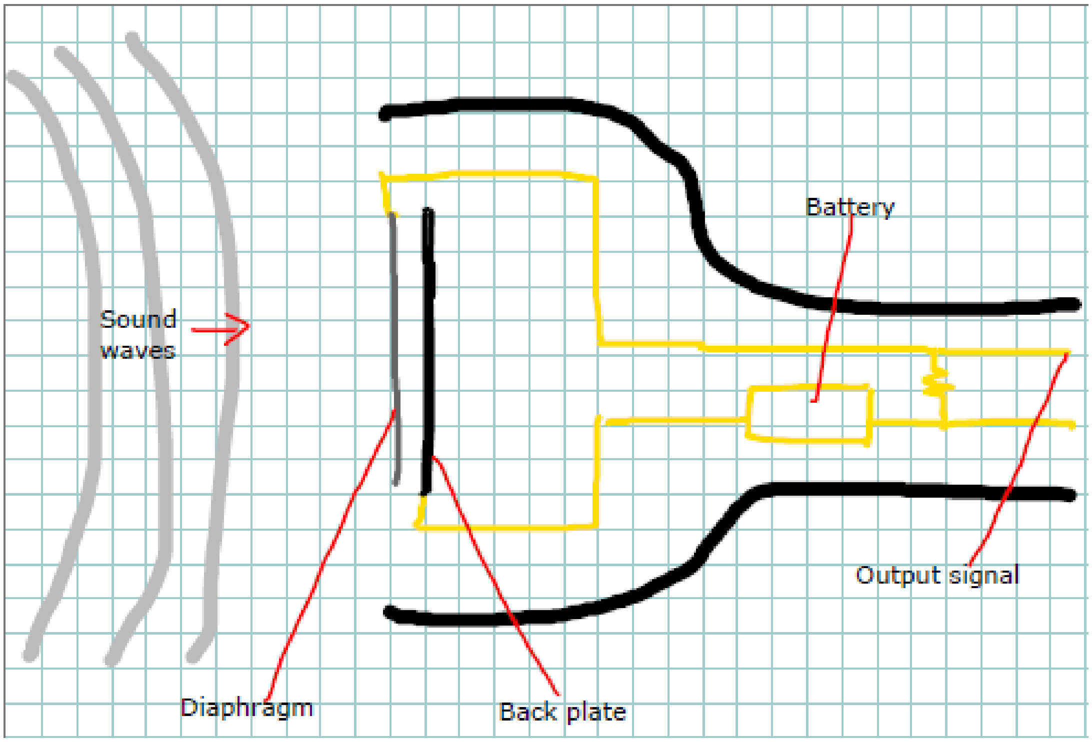


Figure 3.4: *A cross section view of a condenser microphone's transducer.*

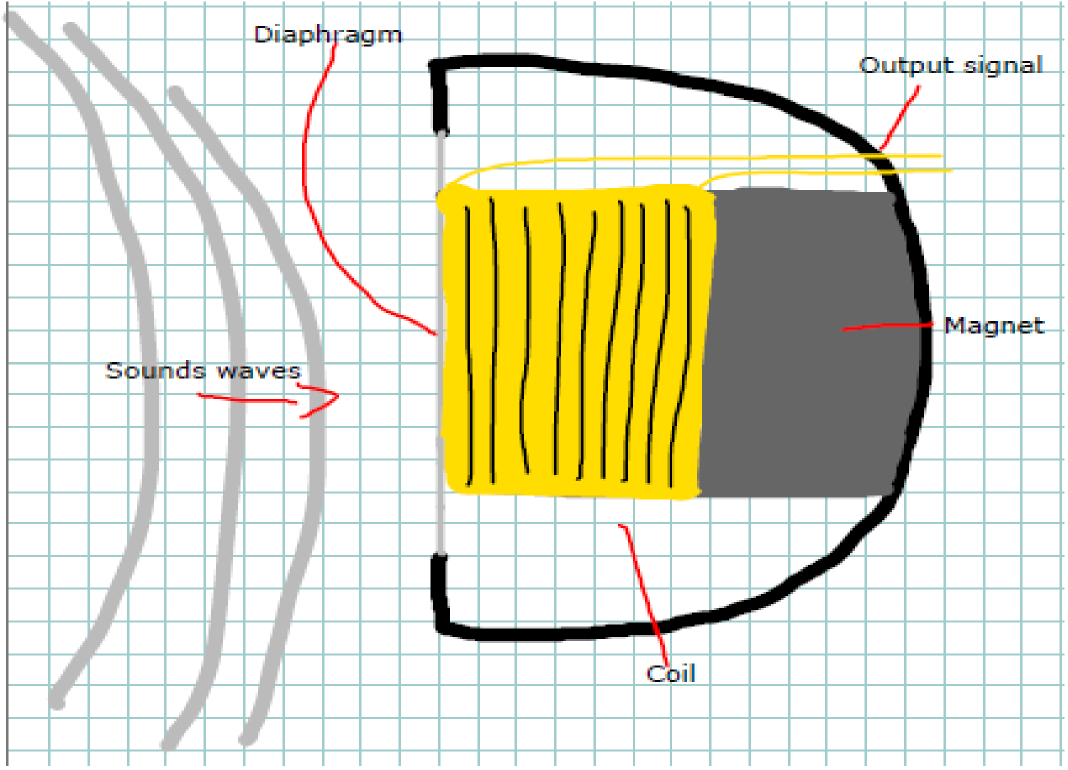


Figure 3.5: *A cross sectional view of a dynamic microphone design.*

Condenser microphones are much more delicate that dynamic microphone but deliver much better audio fidelity. This stems from the diaphragm and back plate design. The back plate design is very sensitive and cannot handle high pressure sound waves. The dynamic microphones coil design makes it much more rugged and slightly less sensitive because it is much less susceptible to vibrations. Another point to consider when deciding between the two types of microphones is the extra power that will be needed to power a condenser microphone. However, this has little effect on this design because power consumption is not a major concern with the car battery and alternator available as power sources.

The fact that the microphones in the emergency vehicle detection system will be subject to random vibration and high pressure winds, makes the dynamic microphone the optimal choice. This design will handle the excess movement of the coil much more effectively than the condenser microphone.

3.1.7 Microphones

Now that the design requirements have been identified, the decision on which brand and model of microphone to use can be made. The emergency vehicle detection system design needs a dynamic microphone with a cardioid pickup pattern that is relatively small. The frequency range of the sounds being analyzed is roughly 500-3000 Hz and the response should be as flat as possible in that range. Given that most microphones have a frequency range of about 50-20000 Hz, the main requirements when deciding on a microphone are the amount of roll off outside the angle of acceptance and the linearity of the frequency response in the range of 500-3000 Hz.

There are countless microphones that fit these specifications. The main variable in choosing the microphone for the emergency vehicle detection system that fits these criteria is cost. The cost difference in microphones depends on a number of factors, but these factors have little to do with the specifications of the microphone. More often, the more costly microphones are much smaller and lightweight. For the emergency vehicle detection system, size and weight are important to consider, but not completely necessary when looking for proof of concept. For comparison purposes this paper will discuss two different microphones. One of which is the optimal choice if cost was not a concern and one that will be implemented in the system because it costs much less.

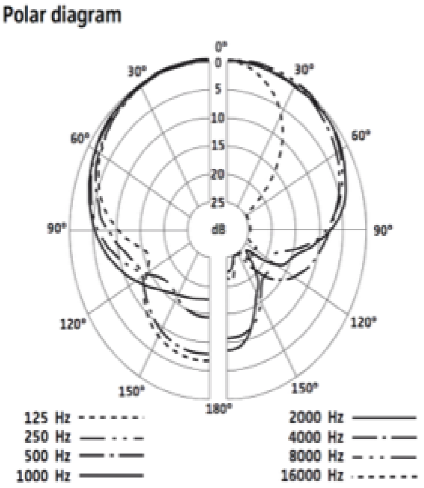
3.1.8 Sennheiser e604

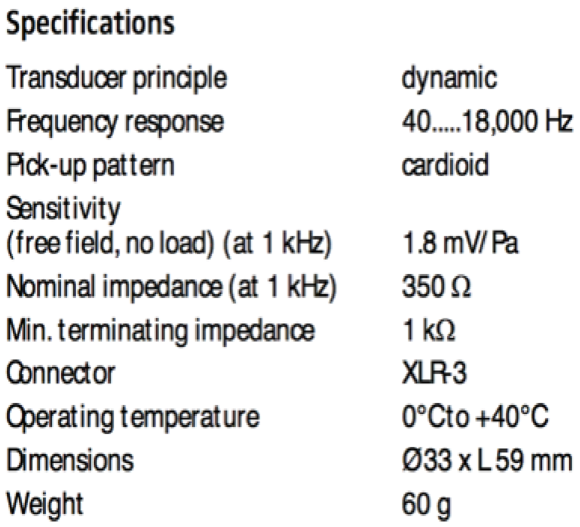
To begin, the ideal microphone for the emergency vehicle detection system is the Sennheiser e604 or equivalent. These microphones are roughly $140 each, making the total investment for the microphones $560. Because this microphone is designed to be used on drum sets and built with a Humbucker coil, it is built to handle sound pressure higher than 160 dB. This feature would help the system handle the wind pressure and the excess vibrations of the car. Another advantage to this microphone is its stainless steel casing and mount have been designed to have very little sensitivity to impact. All of these features come together in unit that weighs only 60 grams.

These features complement the Sennheiser e604's standard technical specifications like sensitivity, roll off, and frequency response. This model has a sensitively reading of 1.8mv/Pa at 1 kHz giving it the capability to detect the siren, but not overly sensitive to help noise cancellation. The roll off rate of the e604 is ideal for the emergency vehicle detection system because it does not attenuate too quickly outside sixty degrees. Referencing the polar pattern below, at a frequency of 500Hz, the e604 has roughly a 3dB and 5 dB drop at 60 and 90 degrees, respectively. At the top siren frequency of 3 kHz, the pickup is roughly the same with a 2 dB and a 5dB drop at 60 and 90 degrees respectively. These roll off rates will allow for the loss in amplification needed to help triangulate to origin of the sound, but will do not dissipate too quickly as to eliminate necessary information.

The frequency response of the e604 is ideal in the range of frequencies created by the siren. Referencing the table above, this model has a -58 dB and a -54 dB rating at 500 Hz and 2000 Hz, respectively. With some inspection, it can be seen that the slope of the frequency response in this region is .0026 dB/Hz. This linearity would ensure that there are steady and constant reading across all frequencies, eliminating the need for extra steps in the code to account for spikes or dips in intensity at certain frequencies. All of the above values are taken from Figure 3.6 below.

Polar diagram





Specifications

Transducer principle	dynamic
Frequency response	40.....18,000 Hz
Pick-up pattern	cardioid
Sensitivity (free field, no load) (at 1 kHz)	1.8 mV/ Pa
Nominal impedance (at 1 kHz)	350 Ω
Min. terminating impedance	1 kΩ
Connector	XLR-3
Operating temperature	0°C to +40°C
Dimensions	Ø33 x L 59 mm
Weight	60 g

Frequency response curve

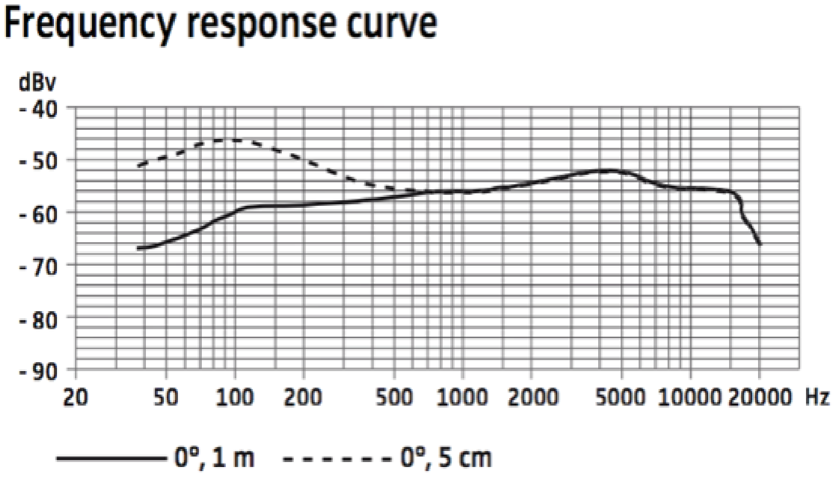


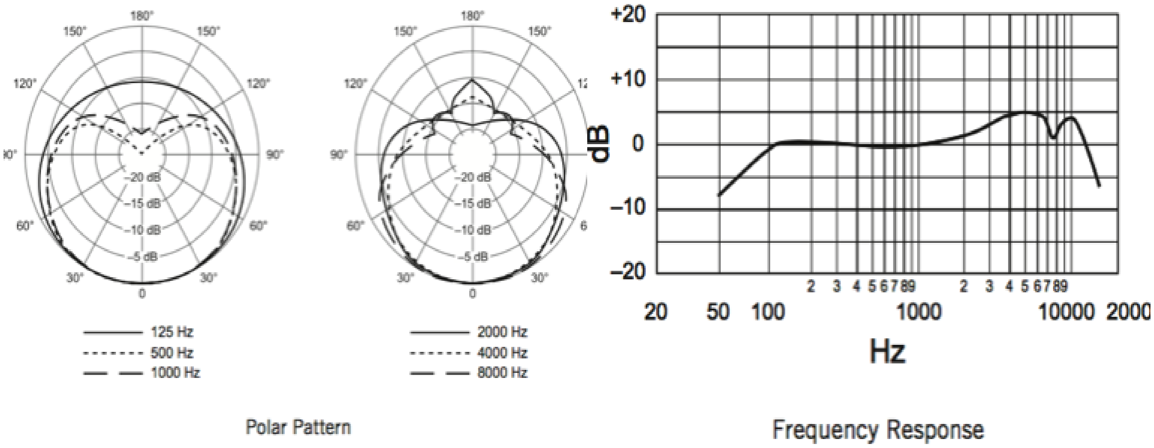
Figure 3.6: *Depicts the important information from the Sennheiser e604 including its polar pattern, frequency responses, and other specifications that were considered.*

3.1.9 Shure SM58

Taking all the data into account, the Sennheiser e604 is the ideal choice for the emergency vehicle detection system. However, in order to save 10% of the budget, some compromises had to be made. With only $400 located to the microphones, the Shure SM58 was chosen to be used in the emergency vehicle detection system. This model costs roughly $99 a unit and meets the technical requirements needed. The only major difference between the two models are the size of each unit and the extra noise cancellation built into the e604.

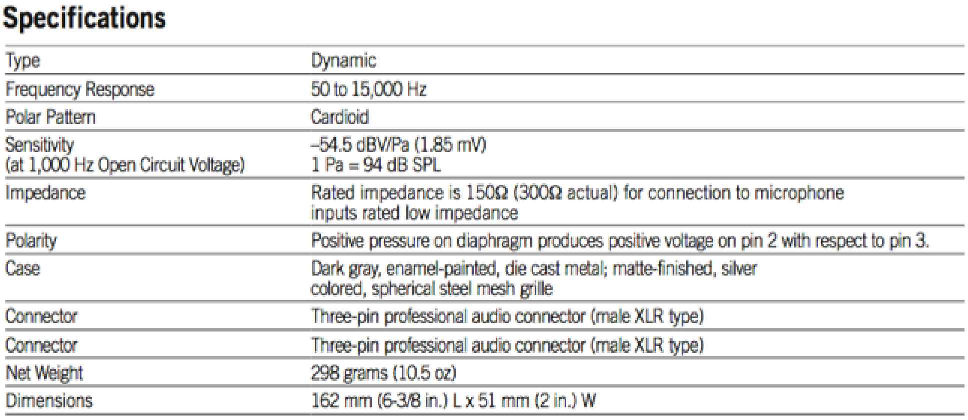
As for the standard technical specifications like sensitivity, roll off, and frequency response, the SM58 is very similar to the e604. With the SM58 having a sensitivity rating of 1.85 mV/Pa at 1 kHz, the two models are almost identical in sensitivity. The roll off rate of both models are similar as well. The SM58's response at 500 Hz at 60 and 90 degrees are -2dB and -5 dB, respectively. These readings only differ from the e604 by 1 dB. At 4000 Hz, the difference is slightly larger, The SM58 has a -5 dB and -10dB response at 60 and 90 degrees. This is going to be a point of concern later in the design because at -10db the attenuation of the signal will be almost negligible regardless of the proximity to the microphone. However, some testing will be done to optimize placement and angle of the microphones to help minimize these losses.

With regards to frequency response, the SM58 slightly outperforms the e604. With a slope of .0006 dB/Hz in the region of 500-2000 Hz, the SM58 has flatter response across the frequency range. However, this variation is very small and almost insignificant. Figure 3.7 below outline some of the key elements from the data sheet



Polar Pattern

Frequency Response



Specifications

Type	Dynamic
Frequency Response	50 to 15,000 Hz
Polar Pattern	Cardioid
Sensitivity (at 1,000 Hz Open Circuit Voltage)	–54.5 dBV/Pa (1.85 mV) 1 Pa = 94 dB SPL
Impedance	Rated impedance is 150Ω (300Ω actual) for connection to microphone inputs rated low impedance
Polarity	Positive pressure on diaphragm produces positive voltage on pin 2 with respect to pin 3.
Case	Dark gray, enamel-painted, die cast metal; matte-finished, silver colored, spherical steel mesh grille
Connector	Three-pin professional audio connector (male XLR type)
Connector	Three-pin professional audio connector (male XLR type)
Net Weight	298 grams (10.5 oz)
Dimensions	162 mm (6-3/8 in.) L x 51 mm (2 in.) W

Figure 3.7: *Depicts the important information from the Shure SM58 including its polar pattern, frequency responses, and other specifications that were considered.*

3.2 ADC

3.2.1 Sampling Rate

In order to process and analyze the incoming sirens from the emergency vehicles an analog-to-digital conversion (ADC) is needed. For this conversion the sampling rate will be based off the cycles per minute (CPM) that the four different sirens types occur. The four different EV sirens are as follows in Table 3.1:

Name	Signal Frequency (Hz)	Cycles per minute	Cycle time duration (S)	Cycle Frequency(Hz)
Wail	725-1600	12	5	.2
Yelp	500-3000	180	.33	3.3
Piercer	725-1600	800	.075	13.33
Hi-Lo	550-650	60	1	1

Table 3.1: *Siren characteristics*

The Nyquist rate will be based off of the yelp since it has the highest frequency: $N > 2 * F_{High}$. Therefore .33 KHz is the frequency that must be sampled above in order to properly abide by Nyquist's criteria. Most analog-to-digital converting units have sampling rates that extend into the MHz Because of this the Nyquist rate will not cause any problems in the system.

The Nyquist rate is very low compared to the sampling rates of modern analog-to-digital converters, and also much slower than modern embedded processors computing power. Because of the high capabilities of those two the chosen sampling rate will be tested at 44,100 Hz, the sampling rate of an audio CD. The sampling rate will be tested at this frequency down to 11 KHz in order to find the sampling rate with the highest success rate. The reason that our sampling rate doesn't have to be double twice the frequency of audio, which is 20KHz is because the sirens are in that very discrete range of 500 Hz to 3000 Khz, therefore the Nyquist rate is at a much lower 6 Khz.

3.2.2 Bit Resolution

The bit resolution for the ADC portion of the Emergency Vehicle Detection System must be compatible with the resolution needed for proper Digital Signal Processing to obtain the optimal decision-making. In most analog-to-digital conversion systems that are designed for audio systems the bit resolution is typically above 16 bits. A CD uses 16-bit resolution while for greater accuracy 24 bits are also used in audio application. Looking at the problem at hand the analog signal of the siren will be very high intensity with distinct filtered frequency ranges.

For best accuracy the higher bits would be used, however the siren signal is not complex, therefore a system that is designed with a lower bit resolution should be able to still properly. Common analog to digital conversion units from manufactures such as TI typically start at 8-bit resolution and work their way up to 24 bits. The benefit of these processors is that no matter what the bit resolution is, they all have high sampling rates which is a beneficial offset for the emergency vehicle detection system.

The sirens that are used in the emergency vehicle systems operate at very distinct frequencies, for usually long periods of time. Because of this a low bit resolution, the 8-bit option, will supply the digital signal that comes out of the conversion with 2^n points of reference. With 8-bit resolution this would leave our system with 256 points of reference. With no large variances in frequency because of the distinct siren patterns, these 256 points should be ample for high accuracy. If after tests the results show poor digital signal resolution then a higher bit rate of 16 will be tested.

The other consideration with the bit-sampling rate for the analog-to-digital portion of the system is if it will be compatible with the job that needs to be done with the digital signal-processing end of the system. Since the bit resolution is the same between the two systems the compatibility will make it so that the resolution has to be a joint decision between the digital signal-processing scriptwriter and the analog-to-digital converter scriptwriter.

3.2.3 Processors

Because the final design of the emergency vehicle detection system calls for four microphones, four different analog-to-digital conversion units will have to be used. One will be placed at the end of each of the four filters off of the microphone lines.

Texas Instrument TLC0834CD: An 8-bit analog-to-digital successive approximation unit. This unit runs on a 5-volt supply voltage. The input current is +- 5 mA, and the input voltage range for an analog signal is 0 to 5V. This range is fits into the output range of the microphones that are being decided. The TLC0834 has an easy microphone input interface, with four different inputs. Only one input will be needed, since there will be four analog streams from the microphone being converted simultaneously. The other three channels have mux connections so they can be turned off. This processor can sample up to 20 kSPS, which is well above the Nyquist rate of 6000 Hz. The outputs run with a serial data link, and the output voltages are dependent on the current. Roughly range from .34 V to 4.6 V or 2.8V (depending on the current in relation to Vcc). The processor matches all the specifications that were called for the system. The error of this processor is +- 1 LSB and it runs in the temperature range that will be faced by the system. This processor will only be viable, however, if the sampling rate of 11 KSPS is unable to obtain proper results than this processor will not be able to handle the CD rate of 44 KSPS.

Texas Instrument ADC084S021: The ADC084S021 is another 4-channel analog to digital conversion unit. However, in comparison to the TLC unit that was previously mentioned, the ADC has a sampling rate range of 50 ksps to 200 ksps. Like the previous unit this one also has an 8-bit sampling resolution. The output is binary and is compatible to most DSP interfaces, which is where the ADC unit's output will be delivered in the system. With a 3-volt power supply this unit typically consumes 1.6 mW, and with a 5-volt supply the power consumption increases to 5.8 mW. This processor, however, also has a mode called power-down which reduces the 3 and 5 volt supply voltage power consumption down to .12 uW and .35 uW respectively. The advantage of this unit comes from it's low power use, this is important in our system since the device will be powered off the car battery supply. The unit has a typical signal to noise ratio of around 49.6 dB, which tends to be a standard for industry analog-to-digital processors. All four channels have an input range from 0V to Va, where Va is the supply voltage for the system (3 ~ 5 Volts). The maximum output voltage of the system is Va - .5 V and the minimum is roughly .4 Volts. Both of these fall inside the range of the accepted input voltages for the digital signal processor being used in the system. The processing rates for this unit range from 50 KSPS to 200 KSPS. Even though both of these rates are well above our Nyquist frequency of 6000 KHz, we might face an issue of oversampling if we try to operate even at the lowest

Sampling frequency of 50 KSPS. Because of this, this unit might face issues in giving viable data for the detection system.

Texas Instrument ADC081S021: This unit is a single channel 8-bit resolution analog-to-digital conversion unit. For this unit the sampling rate can range from 50 KSPS to 200 KSPS. Given the Nyquist frequency of the system of 6 KHz, this unit can properly function within the criteria. The supply voltage can range from 2.7 volts to 5.25 volts and the depending on this supply voltage the unit runs at different rates. The power consumption for this unit is 1.3 mW at a 3.6V supply and 7.7 mW at a 5V supply. The output is supplied as straight binary and is compatible with most digital signal processing serial interfaces. The input voltage range will be 0V to the supply voltage. This processor is at its best performance when it is driven by a low impedance source. This will eliminate capacitor distortion. The microphones that were chosen for the system are rated as low impedance, so they will be matched for this processors best performance. Since the microphones will be continuously picking up noise this unit will have to run on its continuous mode, because of this it cannot take advantage of the power saving mode built into the unit. The output voltage ranges from .4 V to supply voltage -.2. The supply voltage level will be dependent on the input voltage range of the digital signal-processing unit. The supply voltage will have to be tuned down if the input is only a 3V range. The typical signal to noise ratio of this unit is 49.6 dB.

Texas Instrument ADC081S051: The unit operates with a supply voltage from 2.7V to 5.25 V. The typical power consumption ranges with the supply voltage from 1.6 mW to 8.5 mW for the previous two voltage values respectively. The output is a serial data binary stream that has compatibility with most digital signal processing units. For an analog input signal the voltage range is 0V to the supply voltage and its output voltage range is 0 V to supply voltage 0.3 Volts. The signal to noise ratio of the unit is typically 49 dB, which is a level that is theoretically compatible with the pickup levels we need. Like the previous processor this unit is at its optimal performance when it is driven by a low impedance source. The SM 58 microphone that was chosen has a low impedance rating that matches the need for this unit. The frequency range for this processor ranges from 200 samples per second to 500-kilo samples per second. This means that the unit can sample well above the system Nyquist frequency of 6 KHz. This system also has the capability to reach a range of 500 KSPS, this means that with the processor there is flexibility to take tests with different sampling rates that can cover up to the rate of an audio CD, which is 44KHz.

3.2.4 Processor Comparison

Similarities:

First for the comparison of the analog-to-digital conversion unit the similarities between the four different units will be taken into consideration. In response to the ideal bit resolution of 8-bits, all four processors have the same bit resolution of 8. This leaves us with 256 points of reference for all four processors, with roughly the same error of +- 1 LSB. Similarly, all four units have the same input voltage range of 0V to the supply voltage, and similar output voltages of 0V to the supply voltage -.3. The signal to noise ratio of all three units is also a standard 49.6 dB, which is a high enough level to properly pick up the intended signals.

Differences:

One of the major differences between all of the models is the sampling rate difference. The two ADC081S models both operate near frequency levels that are similar to each other, and both of their frequency ranges are well above the Nyquist rate and can operate at higher sampling rates if need be for optimal use. However the ADC084 unit has a minimum operating frequency of 50 KSPS. This is well above the Nyquist frequency, but it may cause issues with oversampling. The TLC operates at a single frequency of 20 KSPS. This again is above the Nyquist rate, but testing will have to determine if this sampling rate will produce proper results.

The next biggest factor from the four different units is the power consumption for each. Even though all four are considered low power units, the ADC081S units have the lowest power consumption. Out of the two of those the S021 has the lower rating. However, the difference between the two power consumption levels at a supply voltage of both 3 and 5 volts is negligible to the over consumption of the unit. Since the biggest differences between the two units is the minimum frequency range that the can sample at, the S021 has the advantage since the minimum frequency being analyzed is well above both of the minimums on the system.

The analog-to-digital conversion unit that is most compatible with the system intended for design is the ADC081S021. This unit has the frequency level that is most compatible to the design constraints, and it also has the lowest power consumption. Each ADC081S021 is only roughly a little more than a dollar. This means that very little is added to the total cost of the design in the overall system.

Table 3.2 below shows the comparisons of the four different analog to digital conversion processors. The main variables that the table compares for the four units are the sampling frequency, supply voltage, power consumption, SNR and bit resolution. These were chosen because they are the most relevant variables of the processor for out project. The table shows that many of the processors are similar, but the deciding factor being the sampling frequency.

Name	Sampling Frequency	Supply Voltage	Power Consumption	SNR	Bit Resolution
ADC081S051	200ksps-500ksps	3.6V or 5V	1.6mW/8.5mW	49.6	8
ADC081S021	50ksps-200ksps	3.6V or 5V	1.3mW/7.7mW	49.6	8
ADC084S021	50ksps-200ksps	3 or 5	3.6mW/5.8mW	49.6	8
TLC0834CD	20ksps	0-5V	-	-	8

Table 3.2: *ADC Processor Comparison*

From the table above it can be seen that all the analog to digital conversion processors are comparable. With the cost being irrelevant, all processors are around only one dollar per unit price, the deciding factor comes down to sampling rate and the bit resolution. Since all bit resolutions are 8, the sampling rate will be the deciding factor. Both the ADC 021 models have the compatible 50 ksps sampling rate. Because of this power consumption is the next factor. The ADC081S021 model consumes less power, even though the emergency vehicle detection system will be powered off the cars alternator and the power will not be an issue, the choice for the processor will be the ADC081S021 model.

The ADC081S021 model output has digital signal processor compatibility. The processor runs standard SPI connection, this will make it easy to interface with the Texas Instrument digital signal processor that will be chosen. The input coming from the Shure sm58 microphone is in the form of an XLR female connection. This connection will then be converted to 3.5 mm audio input. On the PCB board there will be a 3.5 mm audio input. The converted XLR microphone input will be connected to this audio input. The 3.5 mm input will then be filtered through the fifth order Butterworth filter. Then the signal will be inputted into the analog to digital conversion where it will output to the digital signal processor. This will be an SPI interface connection. The process can be seen explain in figure## below.

The above Figure 3.8 shows a block diagram of the connection types that will be used for the emergency vehicle detection system up until input to the digital signal processing unit. As can be seen, the Shure sm58 microphones are connected to and XLR connection, this will then be run through and xlr to audio 3.5mm adaptor. On the PCB board there will five inputs for 3.5 mm plugs, four for the input microphones and one for the output to the speaker. From this point the input from the 3.5 mm on the PCB board will be filtered through the Butterworth filter, which will then be directly connected to the input of the analog to digital conversion processor. After this the output will then be connected to the DSP processor. The analog to digital converter and digital signal processor are compatible with each other through SPI interface.

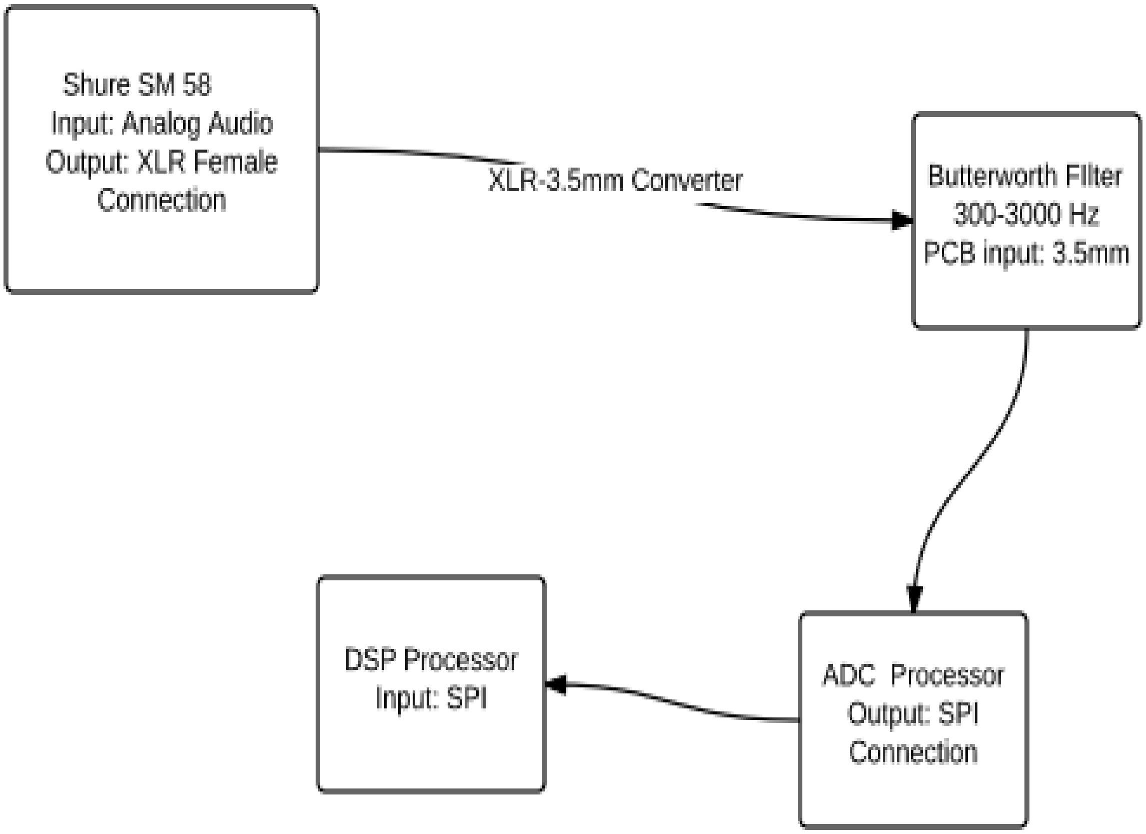


Figure 3.8: *ADC section, input/output connection types.*

<u>Final Part Decision:</u> The final decision for the analog to digital conversion processor is the ADC081S021 unit. This is a cheap unit with the lowest sampling rate out of all the low power devices. It fits all the needed criteria and the easy use make it a bonus in the decision making.

3.3 PCB

3.3.1 Filter Overview

In order to limit the frequency levels of the signals that will be received by the microphone a filter will need to be fabricated. As can be seen from table 3.1 the lowest audio frequency of Emergency Vehicle signals is 500 Hz while the largest occurring frequency occurs at 3000 Hz.

The decision in the filter is whether it will be best for the system to use an active band-pass filter or a passive band-pass filter. The designs for each of the filters are below, all the filters were designed using LTSpice. Below in Figures 3.9 and 3.10 are a passive and active filter that we used to test the concept of signal filtering using a test microphone.

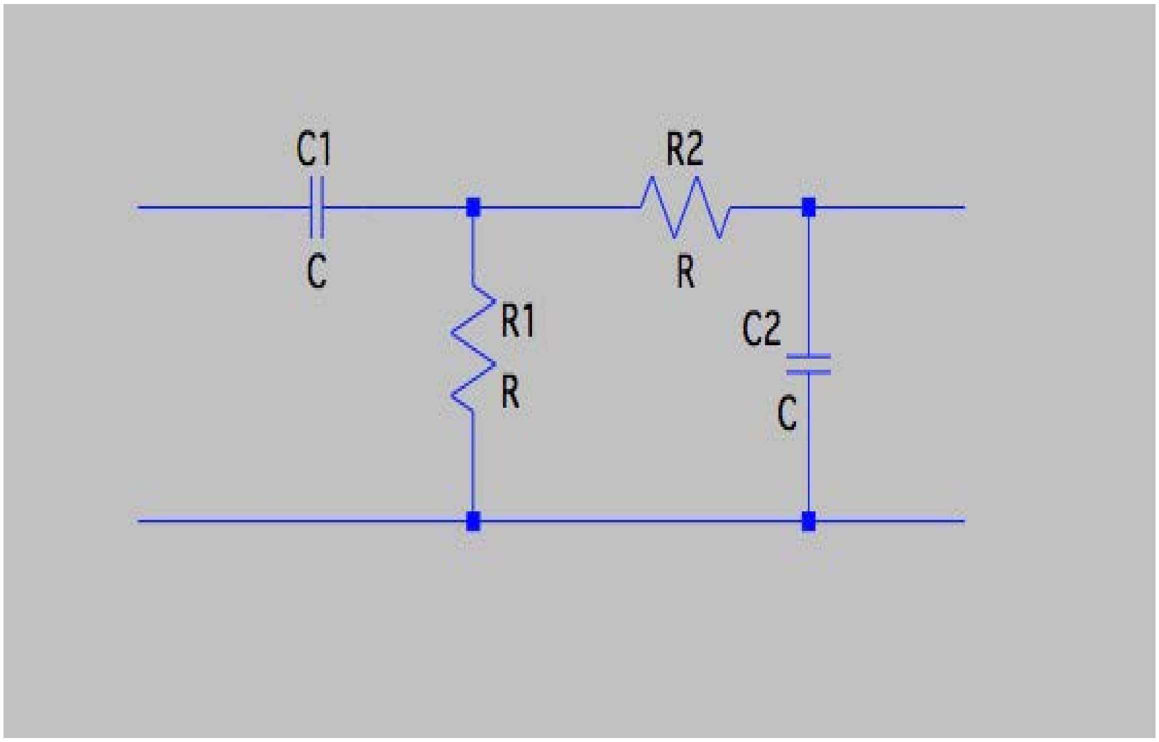


Figure 3.9: *Passive Filter*

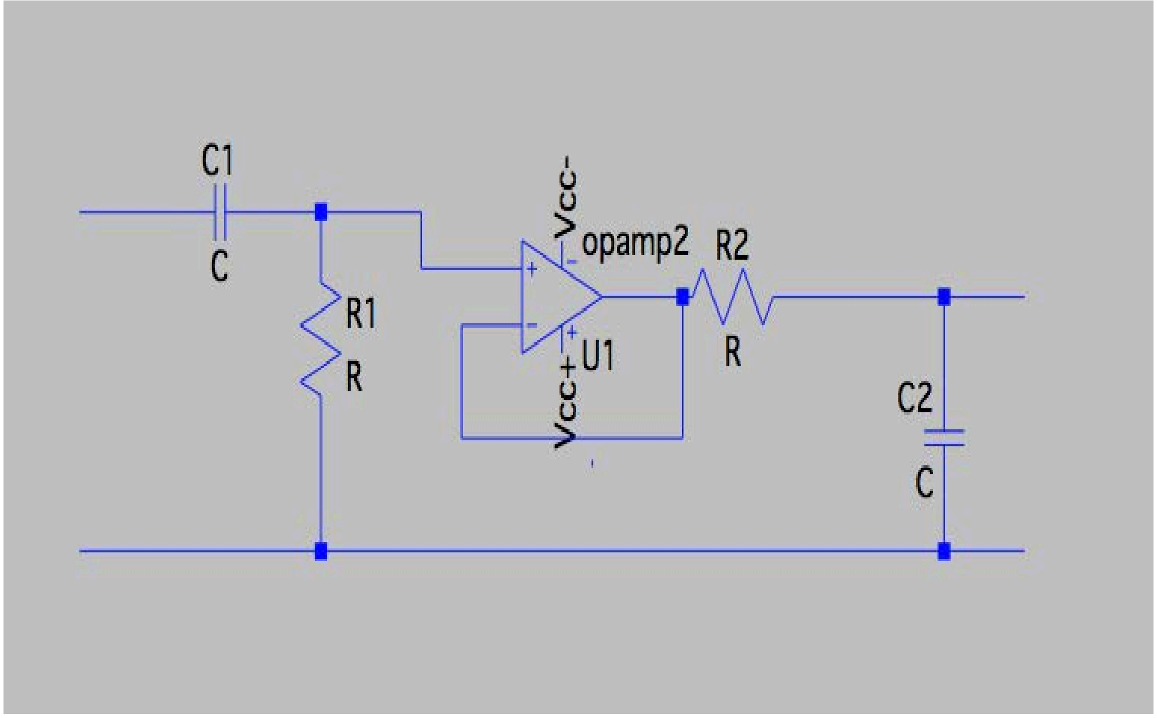


Figure 3.10: *Active Filter*

3.3.2 Passive Filter

Figure 3.9 shows the ideal schematic of a passive band pass filter. R1 and C1 comprise the high pass filter, while R2 and C2 comprise the low pass filter. Together the signal will be filtered between the two cut off frequencies in order to avoid excessive ambient noise being processed by the system. For both of the filters that comprise the band-pass filter the cutoff frequency is:

$$f_c = \frac{1}{2 * \pi * R_n * C_n}$$

The frequency range to be filtered between is 500 Hz to 3000 Hz. All resistors/capacitors will have 5 % error margins. The relevant values for this equation are listed below in Table 3.3 and Table 3.4 shows the results.

Name	Ideal Value	- 5% Value	+5 Percent Value
R1	470 Ohms	446.5 Ohms	492.5 Ohms
R2	470 Ohms	446.5 Ohms	492.5 Ohms
C1	1 uFarads	.95 uFarads	1.05 uFarads
C2	.1 uFarads	.095 uFarads	.105 uFarads

Table 3.3: *Filter elements value table.*

Name	Ideal	Worst Case (High)	Worst Case (low)
F(cut-off high pass)	338 Hz	375.21 Hz	307.7 Hz
F(cut-off low pass)	3386 Hz	3752 Hz	3077.7 Hz

Table 3.4: *Ideal and worst case cut-off frequency calculation*

With the desired circuit element values the filter should not cut out any unwanted frequencies, even in taken the worst-case values into consideration. For better accuracy, and if the proper funding is available, the five percent resistors and capacitors can be swapped out with one percent resistors. This will keep the filter working in a more densely defined frequency range.

3.3.3 Active Filter

Figure 3.3.2 shows the ideal schematic for an active band-pass filter. R1 and C1 comprise the high pass filter, while R2 and C2 comprise the low pass filter. Together the signal will be filtered between the two cut off frequencies in order to avoid excessive ambient noise being processed by the system. The op-amp will be running at unity gain for no amplification between stages. For both of the filters that comprise the band-pass filter the cutoff frequency is:

$$f_c = \frac{1}{2 * \pi * R_n * C_n}$$

Again, circuit elements have 5% margin of error, filtered frequency range 500 Hz – 3000 Hz. The relevant values for this equation are listed below in Table 3.5 and Table 3.6 shows the results.

Name	Ideal Value	- 5% Value	+5 Percent Value
R1	470 Ohms	446.5 Ohms	492.5 Ohms
R2	470 Ohms	446.5 Ohms	492.5 Ohms
C1	1 uFarads	.95 uFarads	1.05 uFarads
C2	.1 uFarads	.095 uFarads	.105 uFarads

Table 3.5: *Filter elements value table.*

Name	Ideal	Worst Case (High)	Worst Case (low)
F(cut-off high pass)	338 Hz	375.21 Hz	307.7 Hz
F(cut-off low pass)	3386 Hz	3752 Hz	3077.7 Hz

Table 3.6: *Ideal and worst case cut-off frequency calculation*

The selected elements for the active band-pass filter theoretically meet the needed frequency range of the desired filter, even when the worst-case error margin is used in the calculation.

3.3.4 Butterworth Filter

The final filter consideration is that of a Butterworth filter. Butterworth filters can be both low pass and high pass, and they are higher order filters. The reason that a higher order filter is important in the emergency vehicle detection system is because it will allow the out of band frequencies to be cut off at an even sharper level than the 20 dB per decade filtering that can be expected out of the first order filters.

For the Butterworth filter, the filtering level is at 20dB/decade times the number of orders in the filter. So for our project the main consideration on the number of orders was if the amount of noise that we expect the emergency vehicle detection system to “listen” to at those frequencies right near the fringe of the desired low pass, and high pass frequencies. After testing some common road sounds on the Sparkfun test microphone, we decided that there would most likely be high levels of noise occurring near those cut-off frequencies.

Also, the Shure sm58 microphones have a lower frequency pick up of 15 Hz. The lower frequency of the filter will need to be established at 300 Hz, due to the frequencies of the emergency vehicle sirens. This shows that the lower cut off frequency will be a little above a decade greater than microphone lower frequency. Because of this close relation, the group decided that a fifth order Butterworth filter would be the best for the emergency vehicle detection system. By running a fifth order low pass, and fifth order high pass filter in series we will create a fifth order band pass filter that will have a gain that can be varied to best fit the analog to digital conversion processor. Figure 3.11 is the proposed Butterworth Low pass filter. The figure below was constructed on Multisim.

The figure is a Butterworth low pass filter the high pass filter will be the same, but with the capacitors and resistors switched. For the Butterworth filter is composed of a first order input filter, followed by two-second order filters. The low pass and high pass filters vary only in the resistors and capacitors in the image labeled C1, C2, C3, C4, C5, R1, R2, R3, R6, and R7. These resistors and capacitors only need to be swapped for the high pass section of the filter that will be following this low pass section.

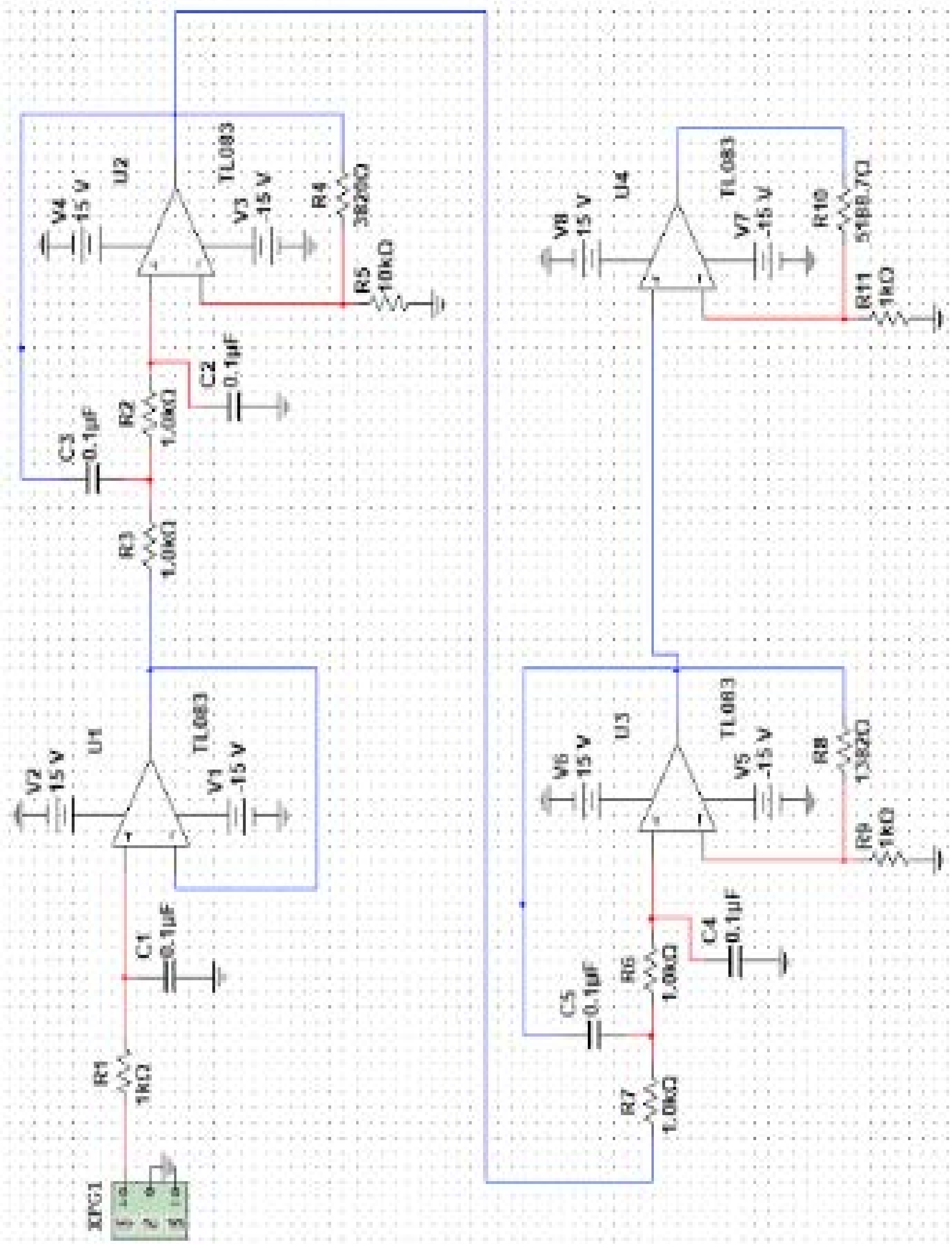


Figure 3.11: *Proposed Fifth Order Butterworth low pass filter.*

3.3.5 Filter Comparison

The passive filter has the advantage of the active because it has no need for an extra dc power supply to be powering an op-amp. Because of this it makes the power consumption of the active higher than the passive since the Vcc terminals of the op-amp section of the circuit are constantly connected to DC voltage.

The active filter has an advantage over the passive filter in that it allows for a stage of signal amplification in replacement of the unity gain portion of the op amp.

The passive filter will experience insertion loss even within the pass band. Also, the source and load form part of the characteristic impedance of the passive filter. In the active filter there will be no insertion loss, and the op-amp provides a buffer against the output impedance. This insertion loss is the deciding factor between the different filter methods that are being discussed. If the insertion loss in the passive circuit is to great for a high-resolution signal to be delivered to the analog-to- digital conversion unit, then the rest of the system will be affected by the missed accuracy. However, if the insertion loss in the passive circuit is found to be at a low enough level for the analog-to-digital conversion unit and the digital signal processing unit to efficiently and accurate complete the tasks then the passive filter will be the better choice.

For the frequencies and signal levels that will be in use for this system the active filter does not have that great of an advantage over the passive system. The active filter has the benefit of amplification, but the expected signals to be received from the microphone are at a high enough voltage level that the insertion loss through the passive system should theoretically be low enough to prevent any problems in the system.

However, depending on the analog-to-digital processing unit that will be used in the system, there may be need for amplification in order to cover a large enough voltage range for proper conversion. This will be dependent on the specifications of the exact integrated circuit that will be used to take in the output of the filter, whether active or passive.

Both filters will be tested and system success rates will be compared find the filter that has the highest success rate. The factors that were mentioned above will be measured and their effects on the final outcome will be taken into consideration.

The third filter that is compared is the Butterworth filter. For the emergency vehicle detection system a fifth order Butterworth filter was considered. This filter will be comprised of a Butterworth low pass filter in series with a high pass filter, this combination will allow for a 5th order Butterworth bandpass filter. Because the filter is fifth order which means 100 dB/Decade filtering outside of the cutoff frequencies.

Like the passive first order filter, the Butterworth filter will run on low power TL084 op-amps. This means that the power consumption of the Butterworth will also be low enough that it can be considered irrelevant in the determination process. However, the one area where the Butterworth filter differs from the other two is the space it will take. Since each Butterworth band pass filter will take two op-amps, this means there will need to be a total of eight op-amps in order to properly filter all four microphones.

The size issue comes into play with the overall design of the PCB board. The PCB board will need to fit inside the mounting system; because of this the Butterworth may cause an issue with the overall size of the system. The three different filters are compared in Table 3.7 below.

Name	Out of Band Filtering (dB/Decade)	Number of TL084 processors	Amplification Level
Passive (1st Order)	20	0	Loss (Passive)
Active (1st Order)	20	1	Vcc- to Vcc+
Butterworth (5th Order)	100	8	Vcc- to Vcc+

Table 3.7 *Filter comparison table*

Table 3.6 compares the different filters with three variables to help decide the one that best fits the system. The three variables chosen were out of band filtering, we want the steepest drop off so the Butterworth filter is optimal. Number of TL-084 processors was the next variable because for the system we wanted to keep the number of active components to a minimum to avoid power consumption. The third variable of comparison is the amplification level, since the analog to digital conversion unit has an input level from 0 to Vcc, the filter cannot amplify past this voltage source level.

From the table above it can be seen that filter with the highest level of out of band filtering is the fifth order Butterworth Filter with a drop off of 100 dB/decade. Since this filter will also allow for 8 stages of amplification for the input to be filtered before the analog-to-digital conversion processer it seems to be the best fit for the emergency vehicle detection system. The input is compatible with the 3.5 mm audio input that will be coming in from the microphone, and the output voltage levels are compatible with the chosen analog-to=digital conversion. Four Butterworth band pass filters will be needed for the emergency vehicle detection system, one for each microphone.

3.3.6 3.5 mm Connector

Since the main source of external input and output connection will be the 3.5mm connector this section will talk about the pin output of this specific connector, and how this output signal will be implemented by the different processing units in the book.

Figure 3.12 below shows the pin set up of the 3.5 mm connector. All pins are stacked on top of each other.

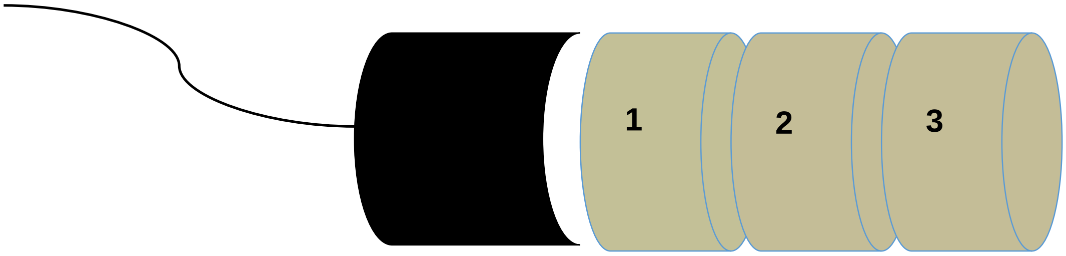


Figure 3.12: *3.5 mm Head connector pinout*

Figure 3.12 shows the pin representation of a 3.5 mm connector. Pin 1 on the connecter is the wiring run to ground. Pin 2 represents the audio out signal. This pin will be used for the output of the 3.5 mm connector at the output of the system. This wire will run on the PCB board from pin 2 of the 3.5mm connector to the output of the analog to digital conversion unit.

Finally, pin 3 on the 3.5 mm connector is the signal to the microphone. This is the pin that will be used for the four inputs from the microphones. The PCB will have four of these ports that will have similar wiring. The wiring will connect the pin three from the microphone directly to the band pass filter. Here the signal will be filtered at the appropriate levels and sent into the input of the four separate analog to digital conversion devices.

The 3.5 mm connector was chosen because of this simple schematic diagram, as well as good compatibility with common audio systems, such as microphones and speakers.

3.3.7 Digital to Analog Conversion

For the output of the system a signal will be sent from the arm processor that tells that the driver where the emergency vehicle is coming from. These outputs will be pre-recorded and stored in the digital form on the arm processor. In order to run these signals to the amplifier and speaker, they first need to be converted back into an analog signal. For this the digital to analog converter will be used.

Since there will be an external amplifier for the speaker, the digital to analog conversion system will only have to convert the digital bit stream to analog. The Texas instrument DAC030 is the processor being chosen for this application. This is because it runs at low power dissipation, only 20 mW. Also, this processor is compatible with the selected digital signal processing unit/ARM processor unit. The DAC is an 8 bit converter, which is the same bit resolution as the DSP and ADDC converters.

3.4 Digital Signal Processing

The digital signal processing portion of the emergency vehicle detection system is where the most intensive computer computation takes place. Once the sound wave has been converted to the digital signal by the ADC processor, it will be sent to the digital signal processing processor for analysis. It is here where the digital signal is analyzed to determine the intensity and the frequency of the sound being heard by the microphones. The frequency and relative intensity of each sound wave must be passed through a decision algorithm, which will determine if the sound is a siren and where it is coming from.

In order to find the information needed out of the bit stream created by the ADC processor, a dedicated digital signal processor will be needed to implemented. The processor in question must be able to handle the computation. To find out what specifications the processor must meet, the ADC sampling rate and Nyquist rates must be considered. Another point to consider is the bit resolution used to convert the signal. After these specifications have been defined, the next step is to consider other parameters such as power consumption, built in memory, and cost of the digital signal processing chip itself. As discussed in the ADC processing in the emergency vehicle detection system, the digital signal processing is dependent on the frequencies of the sirens and the unique wave patterns they have.

3.4.1 Sampling rate

To review, the sirens have four unique frequency variations and cycle times. From the table of each sirens data above, the highest frequency comes from the yelp. This means that the Nyquist rate should be based off this frequency. With a frequency of 3 kHz, the yelp siren has a Nyquist rate of .33 kHz. This rate is extremely slow compared to the sampling rate of most processors. This leaves the sample rate open for testing. The sampling rate of an audio CD is 44,100 Hz, and will be used as a benchmark for testing. Then the sample rate will varied and tested to as low as 11k Hz to find the ideal rate.

3.4.2 Bit Resolution

As for the bit resolution of the ADC, 8-bits will be more than sufficient. The eight bits gives the processor 256 points of reference. Because the sirens have such unique frequencies and patterns, the 8-bit resolution should suffice. However, if testing proves that 16 bits gives the system greater accuracy with minimal effects on speed, 16 bits may be implemented. In either case, most processor can handle in excess of 32 bit resolution.

3.4.3 Processing Speed

Now that it is clear that the sounds being analyzed by the emergency vehicle detection system can be sampled at very low rates with low resolution, the next step in choosing a digital signal processor is to determine the speed in which the processor needs to operate.

The processor needs to be extremely efficient to handle the computation. The best example to put it into perspective is if the system hears a “wail” pattern. This siren has a cycle time of 5 seconds, meaning the processor is going to need at least 2.5 seconds of sampling in order to see the change in frequency. With a sample rate of 44k SPS, that will produce 110,000 samples from each of the four microphones. Each of those samples will be represented with 8 bit resolution making the amount of information being processed to grow exponentially. If the emergency vehicle is traveling 40 miles per hour, it will travel almost 150 feet in that time. That is almost 50% of the target warning range of 100 yards taken away in the time to simply hear the sound. In this scenario, there is very little time left to process the information, determine the sound is siren, turn off the audio being played in the car, and play the audio recording from the emergency vehicle detection system. All of these processes need to be done quickly enough to give the driver time to respond and move the vehicle to the proper location.

3.4.4 Other Parameters

These other parameters will help narrow down the endless choices in digital signal processors. First, the power consumption of the processor is of no concern here. The car battery and the alternator in the vehicle will supply the whole system will enough power that the few watts of power consumed by the digital signal processing chip will not matter. However, the digital signal processing chip will need to have built in memory capabilities to store the voice recordings, as well as a built in ARM processor to perform logic functions. Given that most digital signal processing chips are relatively cheap, ranging from $2 to $150, and that this chip is going to be doing the majority of the computation, cost is one of the last points to consider in this application. The very last point to consider is the operating system the chip is compatible with. This comes down to preference of the user. Based on the previous knowledge of the group, Linux and Windows Embedded CE are the options for the emergency vehicle detection system.

3.4.5 Comparison

Now that the requirements of the processor have been laid out, the process of comparing and deciding between processors can begin. For the purpose of this paper, four processors have been compared.

Texas Instrument OMAP3525: This unit is a high performing digital signal + ARM processor that runs on Linux, Windows CE, and Android operating systems. This chip is based on the OMAP 3 architecture, giving it the capabilities to process images and audio in high resolution. With its multi-core design, the OMAP3525 has a digital signal processing speed of 430 MHz and a ARM processing speed of 600 MHz. With these speeds, the processor is able to to up to 179 GFLOPS and up to 358 GMACS. It also has the necessary storage capabilities needed. It has 32KB of program RAM and Cache, as well as, 32 KB of SRAM and 16 KB of ROM. All of this processing power only consumes 2-10 watts of power. The 64 32-bit general registers will be more than sufficient for coding purposes. With this processors multiple supply voltage modes, it is capable of being tuned for optimal synchronization with the ADC processor chosen. A major advantage to this processor is its multi core design. With its quad core architecture, the OMAP3535 could potentially run each of the four microphone inputs on each of its chips. The main issue, however, is that this processor is designed for high definition video processing. This is not a major concern, but must be addressed. There will be certain programing and software requirements for the chip to be utilized for audio processing.

Texas Instrument DM3725: This processor uses Linux, Windows Embedded CE or Android operating systems to perform high performance image, video, and audio processing. This chip is designed to consume very little energy and operates using .9 to 3.0 volts. With its ARM processor capable of running at 300, 600, and 800 MHz, it has ample processing power for the decision logic. It also has 256 KB in Chache, allowing it to store the audio recordings needed for the emergency vehicle detection system. This processor also has 32K-Byte L1P Program RAM/Cache, 80K-Byte L1D Data RAM/Cache, 64K-Byte L2 Unified Mapped RAM/Cache, 32K-Byte L2 Shared SRAM, and 16K-Byte L2 ROM. The DM3725 also has a multicore digital signal processing unit capable of operating at 660 or 800 GHz and up to 179 GFLOPS. This design also utilizes chip select pins which will allow for simultaneous chip operations.

Texas Instrument TMS320VC5510A: The TMS320VC5510A is a dedicated digital signal processing unit that was designed for high performance and low power consumption. It operates using 3.3-V I/O Supply Voltage 1.6-V Core Supply Voltage. With its 1800 MHz clock rate and dual multipliers and a 6.25 ns instruction cycle time, this chip has the processing power needed in the emergency vehicle detection system. It also has 24 kBytes of instruction cache, 32 kbytes of on-chip ROM, and 8M x 16-bit addressable external memory space, giving this chip the memory needed to perform the task at hand. This design also has two built it arithmetic/logic units. The major issue with this processor is that is runs on DSP/BIOS or VLX, two systems that are unfamiliar to the members of this group. This is not a major concern, but should be taken into consideration. Also, if this chip is being utilized in the emergency vehicle detection system, a separate ARM processor may be needed to help with the processing of information. Also, this processor may not be able to handle all four microphone inputs at once, meaning that there will be the need for multiple digital signal processing chips.

Texas Instrument TMS320VC5502: This is another dedicated digital signal processing unit, that if implemented would possible need a separate ARM processor. This unit is capable of operating at speeds of 3.33 ns per cycle time and with a 250 MHz clock rate. This design utilized two built in arithmetic units that are capable of helping with the decision logic needed in the emergency vehicle detection system. With 16 KBytes of instruction cache, 64 KBytes of RAM, and a 8M x 16-Bit addressable external memory, this processor has the capability to store the audio files. Because of its universal design, this processor has interface capabilities with SRAM, EPROM, SDRAM, or SBRAM if needed. However, if this chip is selected for implementation, there will need to be a dedicated chip for each microphone to allow for simultaneous calculations.

Table 3.8 below summarises some to the specifications of the processors discussed above.

Processor	Speed	Number of cores	ARM	ARM speed	Memory
OMAP3525	430 MHz	4	yes	600 MHz	32KB- RAM and Cache, 32KB -SRAM 16KB -ROM.
DM3725	660 or 800 GHz	4	yes	300, 600, and 800 MHz	32K-Byte L1P Program RAM/Cache, 80K-Byte L1D Data RAM/Cache, 64K-Byte L2 Unified Mapped RAM/Cache, 32K-Byte L2 Shared SRAM, and 16K-Byte L2 ROM
TMS320VC5510A	1800 MHz	1	No	-	24 kBytes of instruction cache, 32 kbytes of on-chip ROM, and 8M x 16-bit addressable external memory
TMS320VC5502	250 MHz	1	No	• -	6 KBytes cache, 64 KBytes RAM, 8M x 16-Bit addressable external memory

Table 3.8: *Outline of possible DSP chips.*

3.5 Decision Logic

3.5.1 Frequency Analysis Research

This portion of the code will be used to determine if there is a siren detected by one or more of our microphones. The standard method for doing this is to analyze the signal pattern of a preexisting recording and then match it to the recording that is obtained through the microphone array. Since we have four sirens that we will be listening to, there will need to be a loop that determines which siren in particular the microphones are hearing. In the long run, this information can be used to identify the vehicle that is responsible for the siren. These sirens will be located anywhere between 500 Hz and 3000 Hz. Once we have recordings of the individual sirens an analysis can be drawn up that accurately represent the frequency output of each siren.

From here there are two ways that the analysis can be approached. The first option is to take the incoming recording and sample it at a designated time interval. From our ADC evaluation we will be able to obtain a frequency and magnitude measurement for the incoming noise. This frequency will then be compared to the various samples that we have tested. If the value recorded falls within the designated range of any of the sirens, a follow-up sample will then be tested. If the follow up sample also matches the same curve as the initial sample, then we are on the right track to identifying a siren. Once we get an accurate representation of the frequency patterns of the sirens, we will be able to determine how many positive samples in a row we will need to properly identify a siren. Additionally, there will need to be redundancies in the searching so that even if two consecutive recordings do not match a specific pattern due to noise pollution, the processor will continue to compare the samples in order to catch a siren that might have otherwise slipped through the cracks.

The other way of comparing the samples would be very similar with respect to the incoming recording. The difference would come in the analysis of the preexisting recording. Where the last method would require the original pure recording to be segmented up in order to match it with the incoming recording, this method would require actually finding the frequency response of the siren due to time. The processor would take our recorded values and figure out which of the sirens contain the specific frequency in their pattern through the solving of these frequency response equations with respect to time. Then using our set sampling rate and the same equations can predict what the next value for the frequency should be. This would make the first value less important in identifying the siren and more important in identifying various starting points for evaluation of later recordings. By comparing these predicted values with the later recorded values the processor will be able to identify of the incoming values are representative of a siren or not. This method is a more uniform method of identifying a siren, as it would be able to find any frequency within the reference time, and not rely on the frequency of our predetermined samples. If our processor has enough power, this is the favorable method for identifying a siren.

The other part of our signal analysis is using the various microphones to identify the location of the source of the siren. From our ADC we should be able to obtain a frequency and a magnitude of the incoming recording. We should also get at least three positive magnitudes from our microphones. The microphone with a zero value will be roughly 180º from our source sound. In order to determine distance, we will use the magnitudes of each of the microphones compared with a magnitude response diagram to determine distance from the microphone. Once we have distances, we will be able to use simple geometry to find overall distance from the unit and the angle from it as well. This is the standard method for noise triangulation, and as the name says, we will need at least three microphones active in order to locate our source sound.

Below, in Figure 3.13 is a flowchart describing briefly how this function will take place. There will be many more components to the code than what is shown, but the skeleton for the process is well and clearly outlined. This is one of the standard ways that these types of audio recognition codes are written.

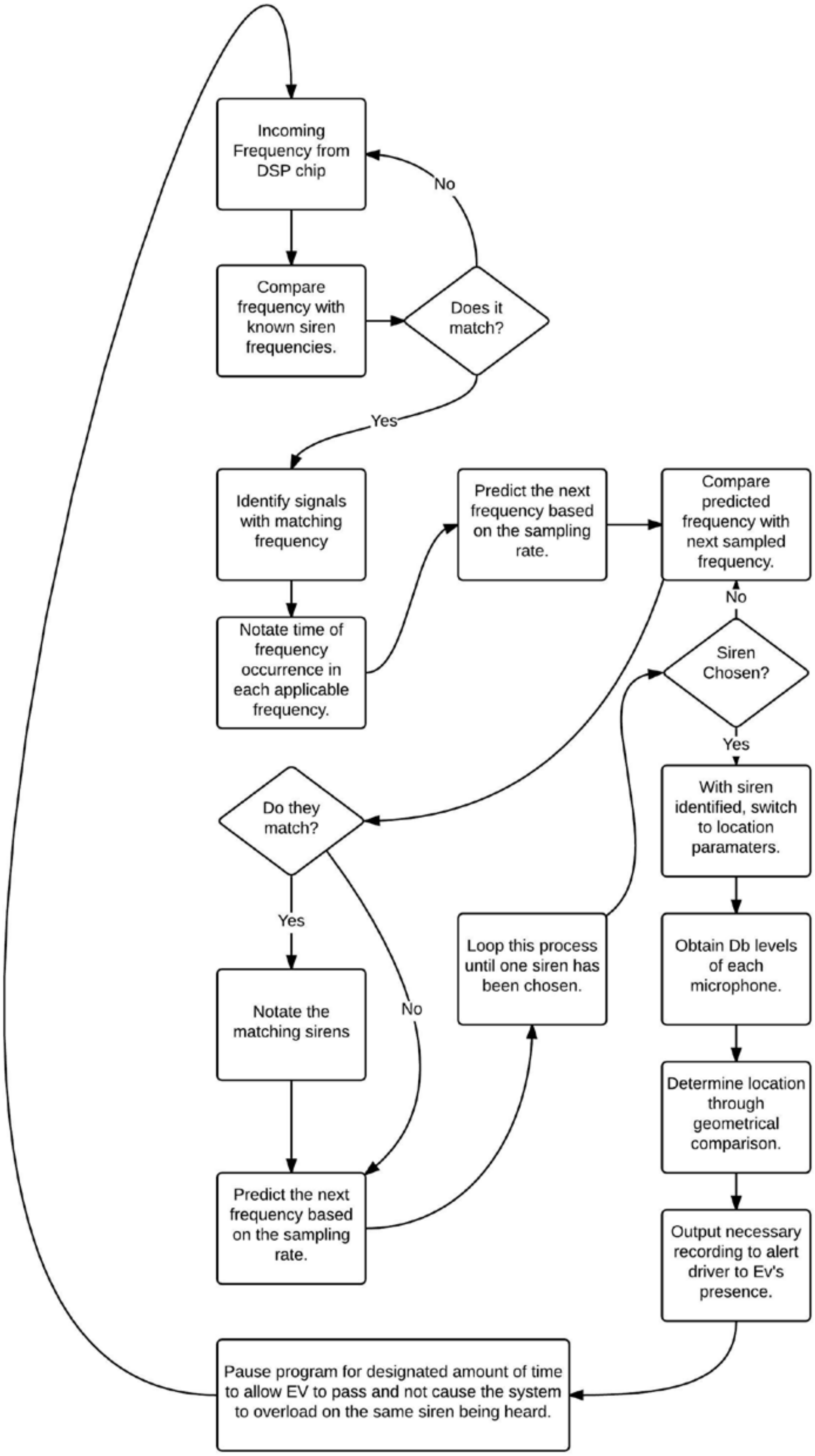


Figure 3.13: *Flowchart outlining decision logic.*

3.5.2 Switching Function Research

After the processor has distinguished weather or not the incoming recordings are sirens, the next thing that will need to take place is the switching function. This can take place in something as simple as a 2-1 multiplexor function. On one input we will have the car audio. On the other we will have the audio signal that we want sent from out processor, be that recordings or something else. The control line will either need to be a separate signal that is on when there is a siren present and off when there is no siren present, or we can just tie the control line to our generated audio and place the generated audio on the 1 side of the multiplexor. This would dictate that whenever our audio wanted to play something over the speakers, the car audio would be shut off momentarily, our message would play and then after it is finished the car audio would resume command.

There is very little difference between these two options of control. If utilizing the first option, the state of the audio lines would not matter unless there was actually an emergency vehicle detected. If there was a slight voltage over our audio lines it would not matter, this is important because our audio lines would be coming straight from the amplifier and it might have a dormant leak voltage. If it did, and we were using the second option of tying our audio lines to the control line, then our processor would take over the speakers at random instances that would not correspond to the presence of an emergency vehicle. Due to this, it would be prudent for us to have a control line that goes straight from the processor to our switching function instead of having it run through the amplifier and then into the control line. The flow chart, Figure 3.14 below, is a visual depiction of the switching function that will be utilized in our system, and how the logic will look in the big picture.

Overall, we are going to need to build a function that takes in two audio signals and plays back one. The one signal passed through will be determined by the EVDS generated audio signal. If there is any signal over the EVDS line, then the 2-1 multiplexor will select the EVDS audio to be played. If there is no signal over the EVDS line, then the car audio will be allowed to play. This will allow for seamless transition between our two audio sources.

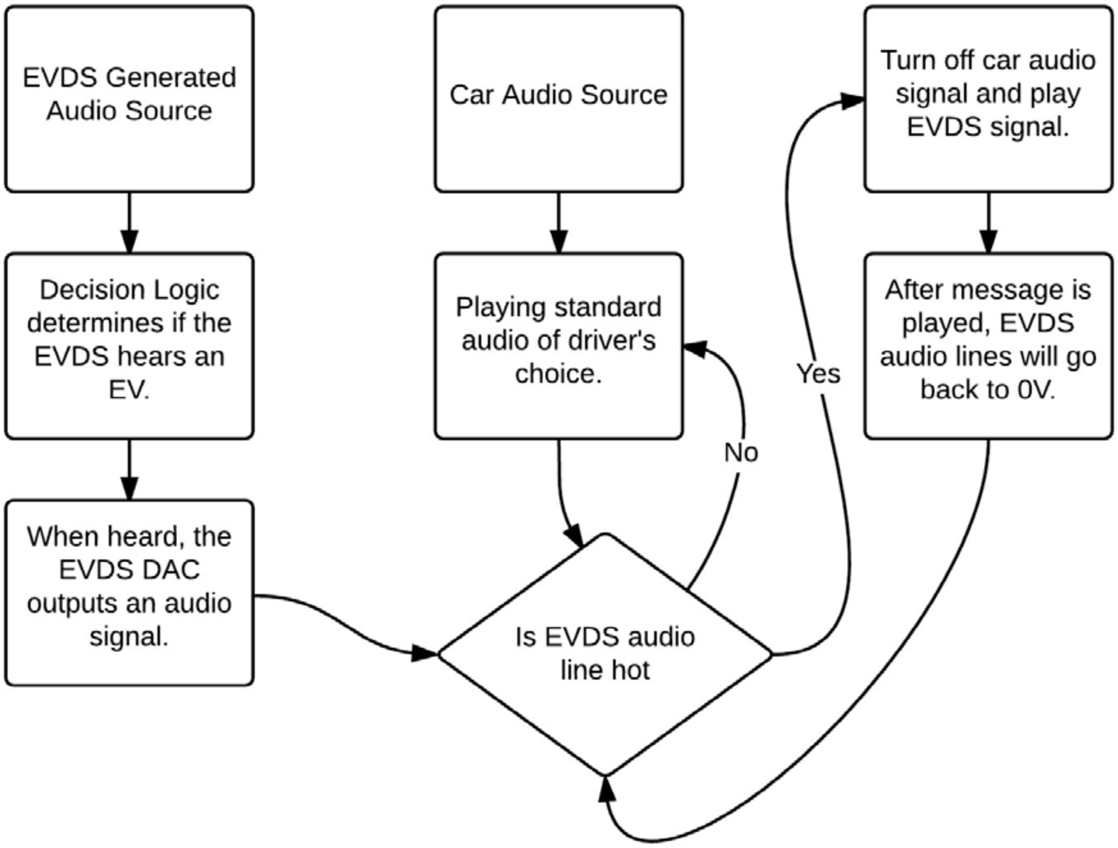


Figure 3.14: *Flowchart outlining switching function.*

3.6 Audio Interface

3.6.1 Receiver

In order to adequately test our product we are going to need to have some sort of audio receiver that will act as a car stereo. This pseudo car stereo will need to be powered and be able to drive the speakers that we are going to be wiring to it. It will need to contain an amplifier within the unit in order to accurately represent a car. In order to accurately test our product we will need our interface to be as realistic as possible.

This leaves us with two main options when looking for playback devices. We can either use a home stereo receiver or a car stereo receiver. They are almost identical in their functionality both will provide amplified outputs and a variety of inputs that we can use to play our car audio. For testing purposes however, a smaller more compact car receiver would probably work better. It would be impractical to be wheeling around a seventy-five pound home stereo receiver when we can simply carry around a 5 pound car receiver. Also, by using a car receiver we would be able to more accurately depict what will be inside a car that would be using our product. Though there will be few differences between the two receivers, there might a few nuances that are different between the two sources that could upset our design in the long run. Due to this, our optimal choice of receiver would be an old car receiver so that we do not have to actually cut into anyone’s car. If it happens that the car receiver that we have acquired does not work, then a home audio receiver will work just as well.

Currently we have two choices that we are looking into. We have in our possession both a car audio and a home audio receiver. The home audio one is a Yamaha HTR-5540. This is a internally amplified home audio solution. As stated above, it is incredibly bulky, at 17-1/8"W x 5-15/16"H x 15-1/8"D, weighing in at a beefy 75 lbs. it would be entirely impractical lugging this device around for our testing purposes. On the other hand, we have an old car audio receiver available for our use. It is a Fujitsu 86120-AA160. This is the standard receiver for a Toyota Camry, one of the most common cars on the market today, and a great platform for our testing. It would be very representative of many of the vehicles that would eventually get the EVDS installed in them. It also has multiple audio sources internals such as a CD player and a FM radio. This would save the hassle of plugging input sources into our receiver while we are testing. It is also a significantly smaller form factor, measuring in at 9”x5”x8”, and weighing only 10 lbs. The only downside to this solution is the output wires and the power wires ill need to be replaced as the ones that were originally hooked up to it were left in the car that it was pulled out of. This is but a minor problem, one that can be easily overcome. Below is Table 3.9 comparing the two receivers at our disposal.

Name	Inputs	Output Channels	Power Source	Cost
Fujitsu 86120-AA160	2	5	12 V DC	$0
Yamaha HTR-5540	8	5	120 V AC	$0

Table 3.9: *Receiver Comparison*

Both of these would work for our purposes. It is very useful that we have both of them at our disposal, because there is a high chance that the Fujitsu will end up not working, as we obtained the part second hand. If that is the case, then we will end up using the Yamaha regardless. Other than this consideration, both parts will work equally well, with the most accurate testing environment coming from the Fujitsu. Unfortunately we will not know if it works until we have been able to adequately test our system further down the road. The final decision on parts will have to come then. Either way, we do have a backup lined up if either of our options does not work.

3.6.2 External Amplifier

In order to amplify our sound from the EVDS, there will need to be an External Amplifier built into our system. While this is something that could easily be built by us, there are many options on the market that will do exactly what we need for less than the cost of building one ourselves. What will be needed is an amplifier that will take in a 3.5 mm headphone cable and amplify the audio from the low level signal provided by our processor and turn it into a signal powerful enough to power the front speakers of the car. This means that we will need a minimum of two channels in order to power one speaker on each side of the car. While different cars have different amounts of speakers, there are generally two main speakers in the front of a car. If we take control of these two speakers we will have adequate volume to alert the driver of its presence. Below is Table 3.10 comparing the amplifiers at our disposal.

Name	Output Channels	Output Format	Inputs	Output Power	Power Requirements	Price
Lepai LP-2020A	2 – L(+-) R(+-)	Speaker Wire	2 – RCA, 3.5mm	20 W/ch	12V DC	$27.80
Lepy 2020A	2 – L(+-) R(+-)	Speaker Wire	2 – RCA, 3.5mm	20 W/ch	12V DC	$19.86
Kinter MA170	2 – L(+-) R(+-)	Speaker Wire	1- RCA	4 W/ch	12V DC (no adapter)	$9.07
Elegiant Furmores 560	2 – L(+-) R(+-)	Speaker Wire	2 – RCA, 3.5mm	???	12V DC	$14.29
Pyle PFA300	2 – L(+-) R(+-)	Speaker Wire	2 – RCA, 3.5mm	45 W/ch	12V DC	$34.26

Table 3.10: *Amplifier Comparison*

The above requirements provide a nice framework with which to start looking for amplifiers, and with a quick search there are many that appear to fit our requirements. In the table above is a comparison of the amplifier options that we have available to us. Most of them will suit our needs, with the exception being the

Kinter Amplifier. It will not provide the power output that we are looking for. When we are ready to design our testing platform we will be able to make an informed decision on which part will work the best and which one will best fit our budget.

3.6.3 Speakers

With the issue of the receiver taken care of, the next issue to look into are the actual speakers that will be playing back both our sounds and the car audio. There is not much to be said about these, except that they must fall within a power range that can be powered by both our receiver and our external amplifier. In the interest of accurate testing, it would be the most similar to the applied environment to test on car speakers. We do not have to test on car speakers for two reasons though. The first is the fact that enclosed home audio speakers will be much more durable for our purposes and all the moving and testing that we will need to be doing. Because we are not going to be putting the system into a car, using a home audio speaker system will be the easiest way to produce playback without having to replace speakers all the time. The second reason is that there is virtually no difference between a home audio speaker and a car audio speaker. The only difference is that the car speaker is not in a housing, as the car will be the housing of the speaker. Both types of speakers are passively powered, meaning that they will be powered from an external source, and they will also both take the same wire inputs. This means that regardless of the type of the type of speaker that we end up testing our system on, we will be able to pull the EVDS out of our testing environment and add it to almost any audio system.

Currently we are looking at three sets of speakers. The first is a set of Bose Model 101 Music Monitors. These are home audio speakers, and we have up to four of them at our disposal. These will suit our needs perfectly as we are only testing on two output channels. We also have another set of speakers that are nearly identical to the Bose's, but significantly lower power. Regardless of the route we take, the RCA monitors will most likely end up being out back up speakers. Our last set is a car audio solution, a set of Kicker 3.5 inch tweeters that would identically model the car audio output that this system would eventually be inserted into. Their power has a significantly smaller range that that of the other two solutions, but they still fall within the range of our amplifier that we examined above of about 20 W max per channel.

All of these speakers would be able to accurately depict the environment that we are designing for. Speakers are not the most complicated part of this testing environment by any means, and they are also the part that is most standard across all markets. Overall, any of these speakers would work perfectly for our purposes, and we will select whichever one makes our interactions with it the easiest. Below is Table 3.11 comparing the two receivers at our disposal.

Speaker Comparison Table	Power	Impedance	Style	Cost to us
Kicker 40CS354	2-30 W	4 ohm	Car Audio	$0.0
Bose Model 101 Music Monitor	10-60 W	4 ohm	Home Audio	$0.0
RCA Home Monitors	10-60 W	4 Ohm	Home Audio	$0.0

Table 3.11: *Speaker Comparison*

3.6.4 Parts Decision: Audio Interface

After carefully examining the above research we have come to a decision on each of the parts that is required for our audio interface. Each of these parts also has a backup part that will be used in the event that the desired part is broken or ends up not fitting the desired specifications after sufficient testing. The first part under evaluation was the audio receiver. We have determined that both for its common power supply to the rest of our system and its accurate depiction of the environment that we are attempting to simulate, the Fujitsu receiver will be used as our car audio source; we will have the Yamaha as a backup in the event that we need it. For our external amplifier we will be using the Lepy 2020A. Due to the fact that we have to purchase an external amplifier and all of the ones that we looked at were extremely similar in specifications, the Lepy will do everything we need of it for the lowest price, allowing us to stick to our budget. It also runs off of the same power supply of 12 V. The speakers we have decided on are the Bose 101 Home Monitors. They are extremely convenient for their all in one capability and similarity to the car speakers that we are trying to simulate. The reason that we did not go with an automobile speaker setup, is that we do not want to have to deal with exposed speaker components and attempting to make a casing for said components. There will be additional speaker wire that needs to be acquired, but this is generic enough that any such wire will work for our purposes. The parts that we have chosen will accurately depict the environment that we are trying to simulate while still allowing us a large degree of control over the system. They are shown below in figures 3.15, 3.16, and 3.17.

Figure 3.15
Lepy 2020A

Figure 3.16
Fujitsu 86120-AA160

Figure 3.17
Bose Model 101

3.7 Power Supply

To most accurately power our system, we will need to use a car battery. This will place some constraints on us as to how much power draw we can have before we exceed the limits of the battery. On its own we will need to also have a way to charge and maintain our battery, but if and when this system would be installed into an actual car, the battery would be charged by the running of the car itself. There would be no need to have an extra charging method. Seeing as most car batteries are the same, almost any battery will do, but it would be best to pick a relatively common one so that we are working with a nice average of what the industry uses today.

In order to simulate the action of turning on a car and activating power to the system we will also need to include a power switch of some sort to the entire system. This will be equivalent to turning the key in a car and causing the battery powered electronics to activate. We do not want our system constantly running, as that would kill the battery of the car and make it so that the car would not be able to start. To do this, all of the positive leads to the battery will run into a switch that will in turn run to the positive terminal of the battery. This will allow us to turn of the system completely, halting the draw of power from the battery. We can do this with something as simple as a light switch, which will also be entirely capable of carrying the loads that we would need it to carry. We will not need multiple switched because if any part of the system is not powered on, then the entire system will be powered off as well. One switch will be entirely sufficient.

3.7.1 Parts Decision

Due to the nature of the power supply, any 12 V car battery will be able to supply us with the power for our system that we need. With this in mind we will look for a used working battery at some point in the future when the time for purchasing said battery comes into play. This will not affect the overall timeline of the project. The battery is easily obtained and fairly universal across all of the automotive industry.

4.0 Design Plan

After we have decided on all of our parts, we will move on to the overall system design. Each of the individual functions for the system has been described in the research sections above, and now we are going to lace them together. The flowchart below, Figure 4.1, depicts a brief overview of the EVDS with the major decisions listed. There are many components that will come together to achieve the final goal which are outlined in detail in the following sections. Not all of the parts described in the rest of this chapter are depicted below, but they all perform critical jobs in the overall performance of our system.

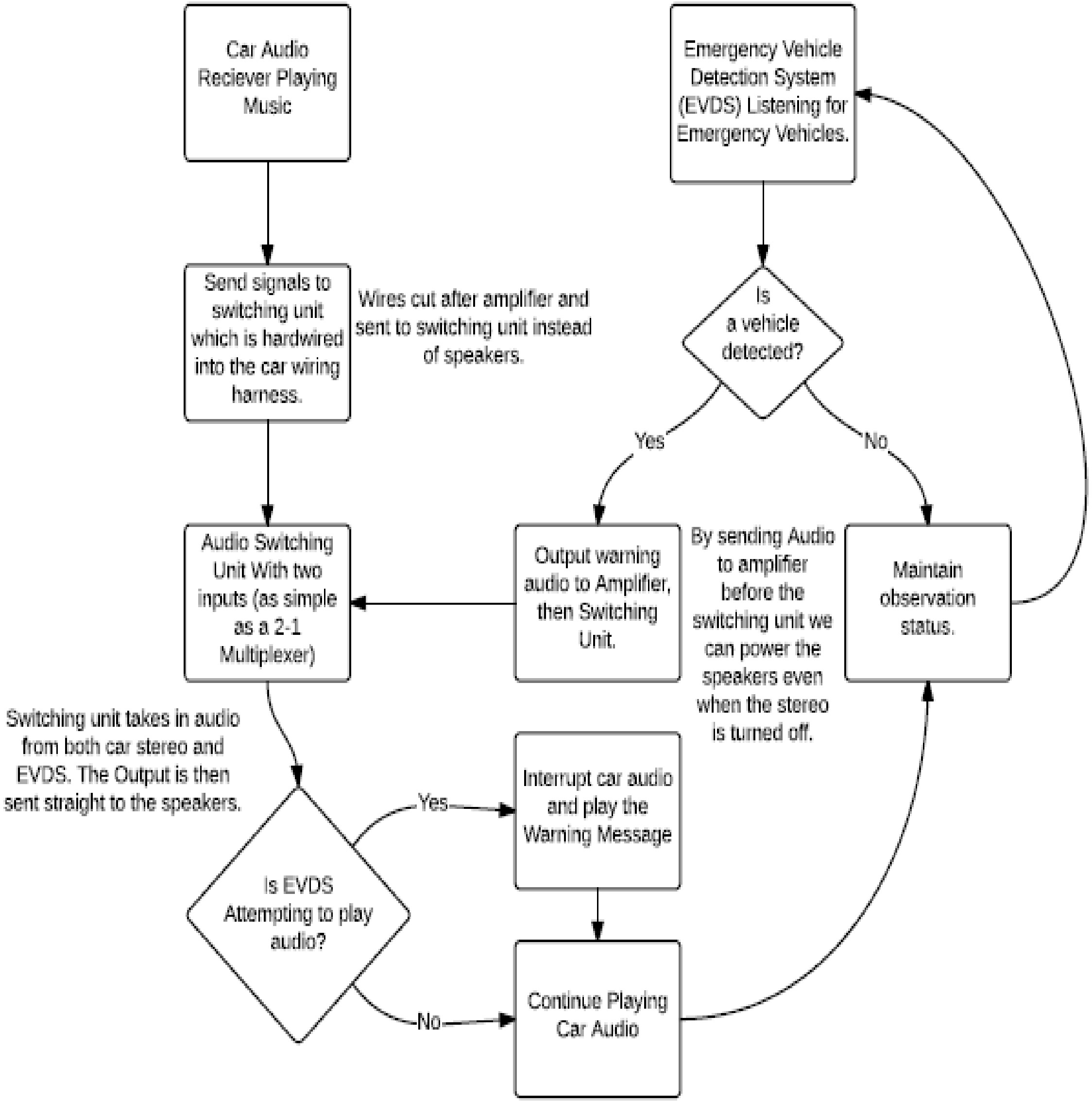


Figure 4.1: *Flowchart depicting whole system operation.*

4.1 Microphones

Four Shure SM58’s will be used in the emergency vehicle detection system. This model meets all the requirements while coming in under budget. This microphone's cardioid polar pattern and flat frequency response will be ideal for this application. These microphones will be mounted using their standard mic-holder, but this mic holder will be attached to the “unit” and not a stand. To begin testing, these microphones will be placed in a square one foot apart facing outward, parallel to the ground. The microphones will be placed so that each faces multiples of 45 degrees from the front center of the car. These measurements will be adjusted and tested until an optimal position is discovered that delivers the best results and has tuned blind spots in the detection, such as directly in front of the vehicle.

4.2 ADC Processor

The Texas Instrument ADC081S021 is the processor that will be used for the analog-to-digital conversion. It meets all the requirements that were looked for, as well as the lowest minimum sampling rate out of all the processors that were observed. Even though the minimum sampling rate of the processor is 50 KSPS, this has been tested in the digital signal processing simulation, and should not cause in issue with properly processing the signal. This sampling rate is also well above the Nyquist sampling rate.

Since the common battery power supply will be a car battery delivering 12 volts, followed by a voltage regulator to keep at 5 volts, the input power supply will be around 5 volts. This means that the power consumption of the processor will be 7.7 mW. Since the system is set up to be run on a car alternator/battery this is near negligible power consumption.

The input of the analog-to-digital conversion processor will be the output of the Butterworth filter. The reason for the Butterworth filter is because the system calls for an active filter in order to amplify the system. A fifth order Butterworth filter is used so the filter drops off at 100 dB/decade. The amplification of the Butterworth filter will be at the level so that no signal exceeds the 5 volt supply voltage.

The output stream of the ADC081S021 is straight binary and is compatible with the chosen digital signal processing processor of the system. Since the output voltage of the analog-to-digital conversion processor is 0 to the supply voltage there is no need for a digital filter between the analog to digital conversion processor and digital signal processing processor. This is because the digital signal processing processor will also be running on the same supply voltage, therefore the output of the analog-to-digital conversion processor will be in the same range as the input of the digital signal processing processor.

The digital signal processing processor that was selected has input compatibly with SPI type interface. The ADC081S021 has output compatibly with this type of interface. Below, in Figure 4.2, is a block diagram of voltage inputs and outputs for the ADC system.

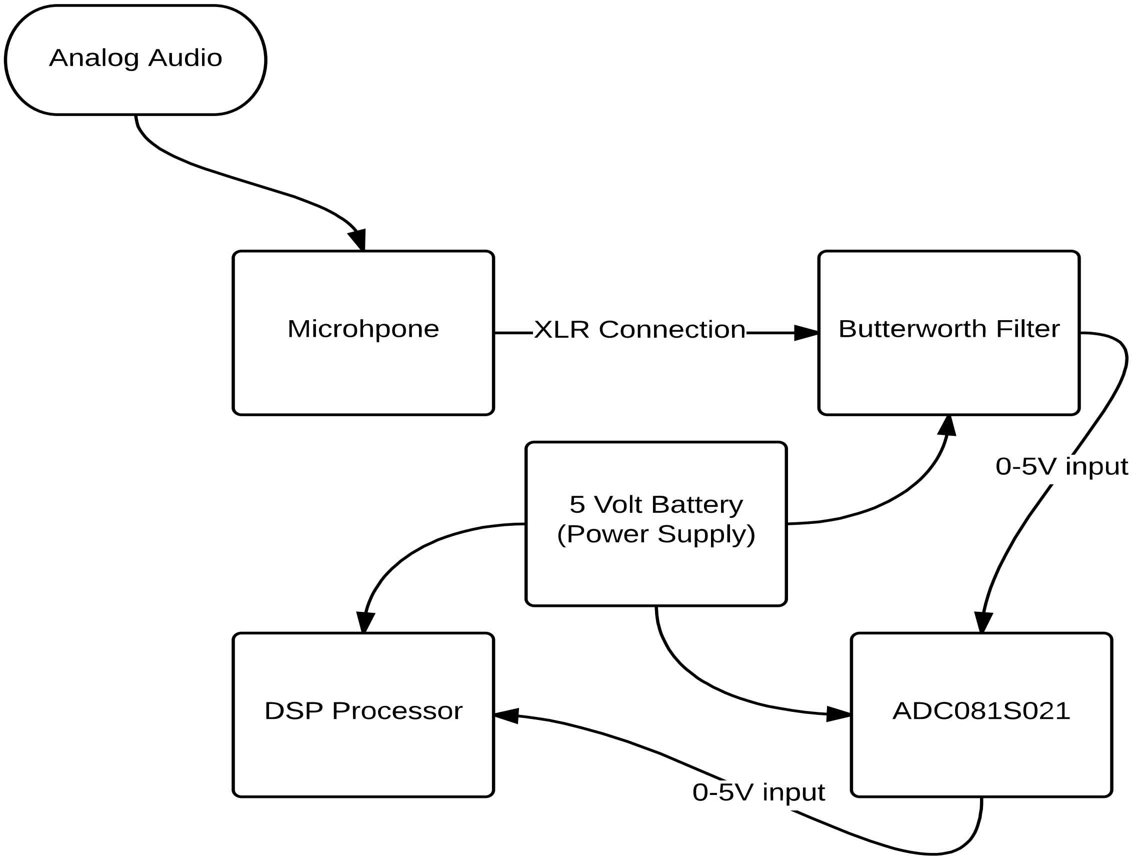


Figure 4.2: Block diagram of voltage inputs and outputs for the ADC system.

In the above Figure 4.2 the set up for the emergency vehicle detection system up until the connection to the digital signal processing unit is shown. Along with the connections, the figure also indicates the voltage levels that each unit can handle as both inputs, and the output levels that they will be delivering. The purpose of the diagram is to show that all the input and output levels for this section of the emergency vehicle detection system are compatible with each other. The five-volt supply voltage will come from a linear voltage regulated twelve-volt car battery. This way the car battery can power the entire emergency vehicle detection system on a five-volt supply to avoid damaging any components in the system.

In the below Figure 4.3 the four microphones were all selected as Shure sm58 microphone units with XLR cords running from them. The output connections of these microphones are female XLR cords. On the PCB board there will be four spots for 3.5mm headphone in. This means that each XLR female connection will have an XLR to audio 3.5 mm adaptor. The adaptors will then be plugged into the board and will be ran through an analog Butterworth filter that has band pass cut off frequencies of 300 Hz as the low pass cut off frequency, and 3000 as the high pass cut off frequency.

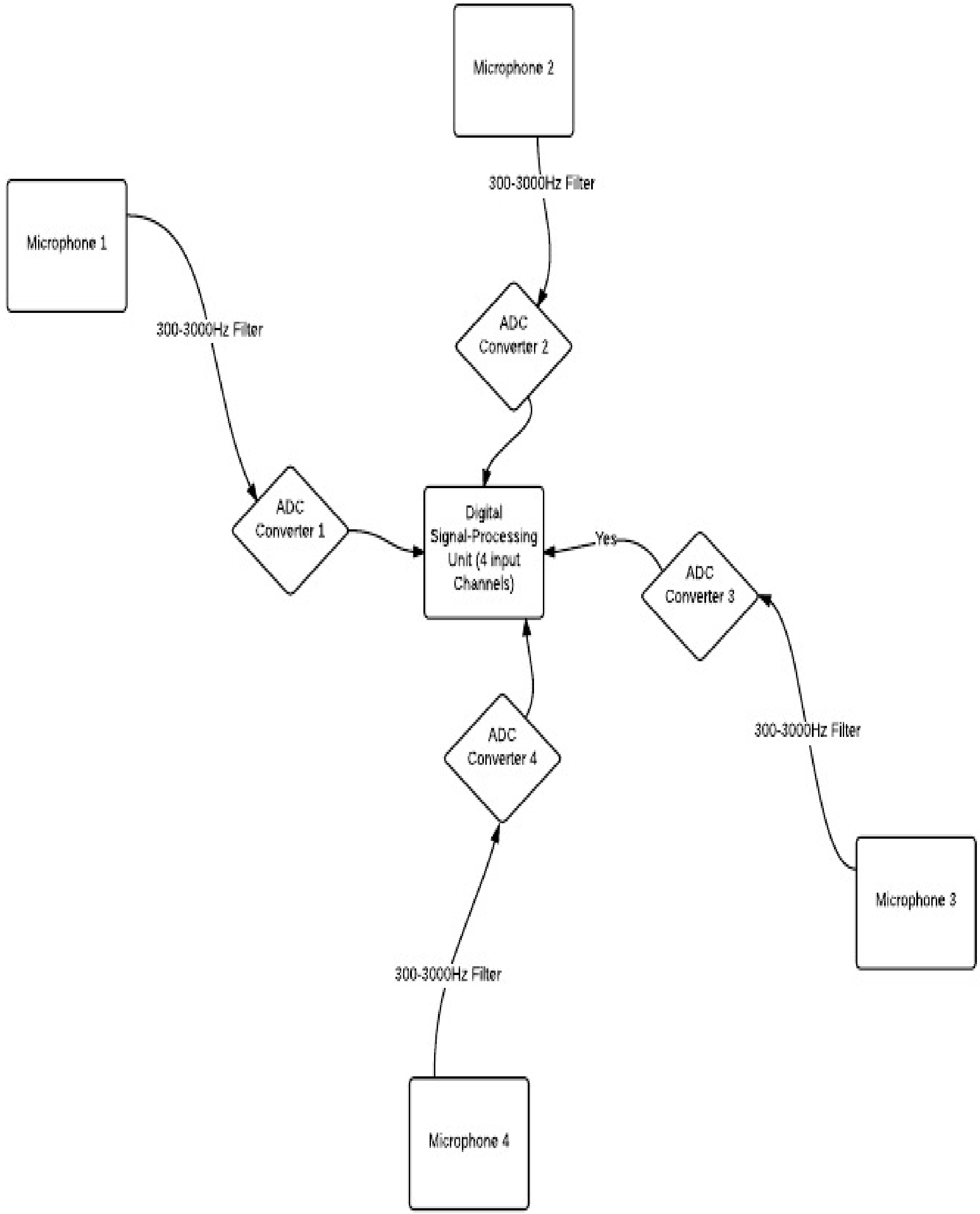


Figure 4.3: *Four-microphone system up to the DSP unit.*

At this point the Butterworth filter will amplify the signal to the levels from 0 V to Vcc+, the voltage of the power supply. These are the compatible input levels of the analog to digital conversion unit. Each fifth order filter will be comprised of a TL084 op-amp, as well as 1 percent error capacitors and resistors. The cost of each of these op-amp units is around a dollar, which makes them useful for multiple test scenarios, and no worries about soldering multiple units. Each analog to digital converter will be the ADC081S021 unit. Again, these are very cheap units, which make ordering multiple possible for different soldering test scenarios. From there the analog to digital conversion outputs continue on each to the digital signal processing unit, through SPI interface connections.

4.3 PCB

4.3.1 Passive Filter

The passive filter was designed using Table 3.3 as a reference for all circuit elements. Figure 4.4 below shows the frequency response of the passive filter that was simulated on LT Spice. As can be seen the filter works with a band pass between the two -3dB frequency levels. The two frequency levels were found to be roughly 393 Hz for the high pass filter, and 2.9 KHz for the low pass filter. The input frequency range that is trying to be filtered is 500 Hz to 3 KHz. The 500-hertz signal will be well within the 300 Hz range of the high pass filter, however the 2.9 KHz -3dB responses may cause issues being below the intended frequency. However, it can be measured from the graph that the filtering of the passive filter at 3 KHz is -3.14dB. As seen in Figure 4.4, the response is still at a level that will not affect the decision making because the signals will still be at a large enough level for the processing.

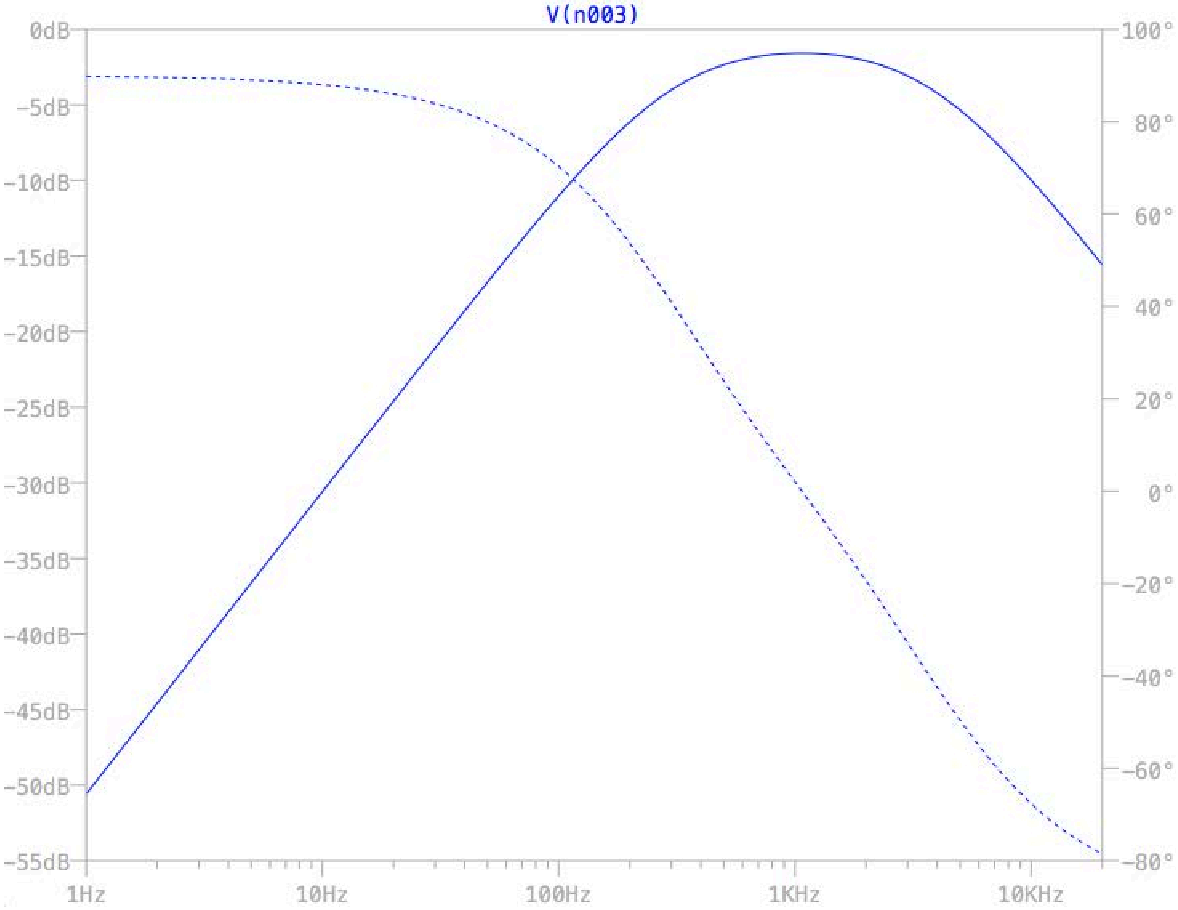


Figure 4.4: *LTSpice Frequency response of passive filter.*

4.3.2 Active Filter

The active filter was designed using Table 3.4 as a reference for all circuit elements. The above graph shows the frequency response of the active filter that was simulated on LT Spice. As can be seen the filter works with a band pass between the two -3dB frequency levels. The two frequency levels were found to be roughly 344 Hz for the high pass filter, and 3.327 KHz for the low pass filter.

The below Figure 4.5 shows the frequency response of the active filter. As can be seen from the image, the filter levels drop by 20 dB per decade past the cut-off frequencies. This is the expected level, however, since the low pass filter cut off frequency is such a low level, there is not much room for the response to drop past -20 dB for the lower levels. Because of this a fifth order Butterworth filter will be considered to steepen the drop off levels of the filter.

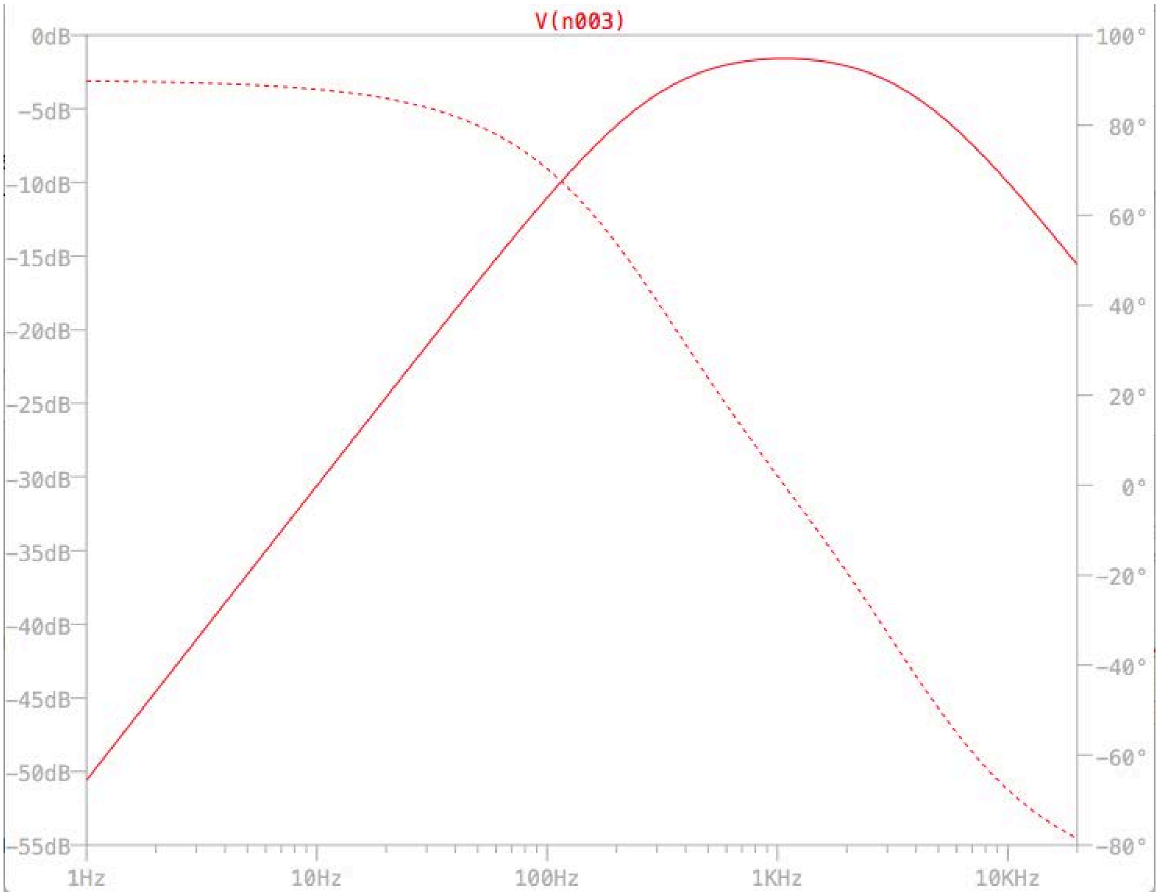


Figure 4.5: *LTSpice Frequency response of Active filter*

The input frequency range that is trying to be filtered is 500 Hz to 3 KHz. The 500-hertz signal will be well within the 300 Hz range of the high pass filter, also the 3.327 KHz -3dB of the high low pass filter provides a bandwidth that includes the entire range of frequencies that are under consideration in the system. Also, the central frequency decibel level with the op-amp is higher than that in the passive system, this of course is because of the amplification in the active system.

However, what was left out of this simulation was the added stage of gain that could be connected between the negative terminal of the op-amp and the output of the op-amp. If after testing the averaged received voltages from the microphones are at too low of levels for the analog-to-digital converter then the active stage has capabilities to add amplification in reference to the added resistors.

4.3.3 Butterworth Filter

In order to have a steeper filter response outside of the band pass section, a band pass Butterworth filter could be used. A fifth order filter will be constructed in order to have the attenuation values of the band pass filter to be 100 dB/decade. The benefit of having the steeper filter response is to cut out even higher level of noise from reaching the decisional making unit. If the frequencies outside of the desired band range, 300 Hz to 3000 Hz, they need to be considered noise. So with the 100 dB per decade drop off past the cut-off frequencies the outside frequencies will be suppressed greatly to allow the decisional logic unit to process a narrower range of signals.

The other benefit of the Butterworth filter over the passive filter is that there are multiple stages of amplification. Since it is a fight order filter, there will be 4 points of amplification for the signal throughout the filtering process. Since the filter will be a bandpass filter there will need to be two Butterworth filters in series, one low pass, one high pass. Since there will be two filters back to back, this gives the opportunity for 8 stages of amplification for the signal between the microphone output and analog to digital conversion input. Since the analog to digital conversion input accepts 0 to the power supply voltage. The system will be powered by a 5 volt power source. Because of this the butter worth amplification will hit a maximum output voltage of only 5 Volts, the rail voltage of the op-amp used. This means that the Butterworth output will be compatible with the analog-to-digital conversion input.

Since the Butterworth filter is an active filter, there will be need for an op-amp in the design. The op-amp that is chosen is the TL084 from Texas Instruments. The TL084 is a common and reliable op-amp from Texas Instruments. This model contains four op-amps on each processor, this is important because each Butterworth 5th order filter will need four op-amps. This means that each TL084 can be used for each Butterworth filter.

The TL084 has a high slew rate and works with high input impedance, which is compatible with the microphone input. The output voltages range from –Vcc terminal to + Vcc terminal. Since the entire emergency vehicle detection system will run on a voltage regulated 5 volts, the maximum output from the filter will be 5 volts. This is compatible with the input of the analog to digital processor which takes an input of 0 to + Vcc.

The other benefit of the TL084 op-amp is that it allows for operation at Unity Gain. This means that in the filtering stages of the Butterworth there can be a direct line run from the op-amp voltage terminal to the output terminal. This is important because the microphone input may not need all eight stages of amplification from the two filters. So instead of having to run through eight stages of amplifications, the filter can have sections that run at unity gain to avoid extra amplification. The below figure shows the simulated response of a fifth order low pass Butterworth Filter, results obtained from Multisim in Figure 4.6:

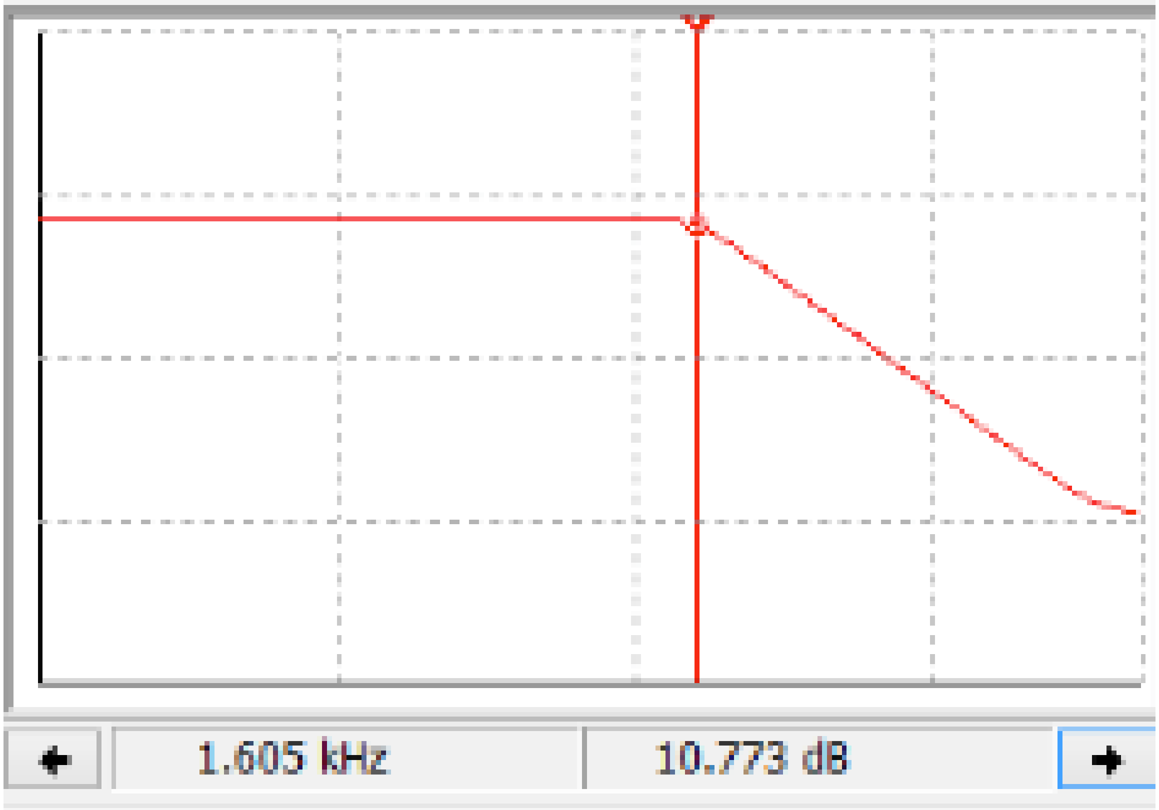


Figure 4.6: *Proof of concept for Bandpass Low pass filter.*

The above Figure 4.6 displays the response of fifth order Butterworth low pass filter with a center frequency of 10,000 Rad/S. The purpose for the constructing was to show that the proposed filter would be able to filter out the noise from the emergency vehicle detection system at a level of 100 dB/decade. Since the high pass and low pass section of the filter are identical, there is no need for the testing of both ends on Multisim.

What the filter simulation did not account for is the needed amplification for the input of the analog to digital conversion unit. The amplification cannot be greater than the supply voltage of the Butterworth filter. This is ideal because the op-amps in the filter will be run off the same supply voltage as the analog to digital conversion unit. This means that the gain will have to be tuned to make sure that all of the incoming signals in the pass band will not be pushed to a too high of voltage level, which would interfere with the conversion process.

4.3.4 Filter Performance Comparison

Each of the two filters worked as expected in the spice models. In terms of having a wider bandwidth, the active filter performed better. Its frequency range covered more than the spectrum under consideration by only a small amount. This gives the system room for error in case a siren occurs at a frequency that is a little bit above the ideal frequency. The passive filters lower end frequency was low enough to allow room for error in regards to the 500 Hz minimum frequencies that the sirens occur at. However, the upper cutoff frequency is slightly below the -3dB frequency. However, the measured loss at the upper frequency limit of 3000 Hz was only -3.14dB. This is still a close enough match to the -3dB level that it can be ignored in design constraint considerations.

Taking power consumption into consideration the passive filter has the upper hand because it has no need for any supply voltage. There is, however, a loss in power due to the components all being passive. There will be a loss in each component. From the LT spice simulation though the center frequency has only a loss of -1.2 dB of the input signal. In comparison the center frequency level of the active component had a signal loss of only -.72 dB when it was run in unity gain mode. The difference between the two is only .5 dB, which is a loss than should not have a large effect on the signal resolution. For the active filter there needs to be a voltage supply powering the op-amp. This leads to power consumption. However, for most op-amps, in this regards the TL084, has very low power consumption of only 1.4 mA/ch.

The main advantage that the active filter has over the passive filter is the stage of amplification if needs be. This is the main parameter that will affect the decision between the two. Since both have low power consumption, low signal loss and frequency ranges that expand for the intended frequency range, both options will be viable. The deciding factor will come after microphone testing and deciding if we need a stage of amplification for a higher signal level or not.

Both designs will utilize 5% circuit elements. For the op-amp that is being used for the active stage of the circuit there will be a Texas Instrument TL084 op-amp that has four input channels. This means one op-amp would be able to supply the amplification for all four filters.

4.3.5 Audio Interface

We will be dealing with multiple audio ins and outs that run from several components to our switching unit. There are the inputs from the car audio to the switching unit, the inputs from our external amplifier to our switching unit, the output from our DAC to the external amplifier, and the output of the switching unit to the speakers that will all be dealt with in this section.

Our car audio will be reaching the PCB from four wires. The easiest connectors for this input will be stereo connectors often seen on the back of stereo components. They will have leads that can be soldered directly into the board, and just like the microphone connectors above will allow for quick swapping of wires or components if needed. Like always, we could simply solder the wires directly to the board, but for ease of use we will use connectors that are made to be detached. Similarly, the input back into the switching unit from the external amplifier will be done in the exact same way. This covers all of the audio inputs to the board, with all eight going to the switching unit.

Our board will have two sets of outputs as well. The first output will be a 3.5 mm audio jack, not a microphone jack as the other four were. This will take our DAC audio which is the message to the driver from the board to the amplifier. The amplifier already receives a 3.5 mm cable so there is no issue with converters needed. The second will be a very similar to the inputs of the car audio and amplifier audio. There will be four stereo connector outputs that will run directly to the speakers. This will be the last that the audio signals will have to deal with the PCB. This also covers all of the audio inputs and outputs that the PCB will have to deal with.

4.3.6 Power Interface

Power input will not be difficult for our PCB. We are already using a DC power source so there will be no need for a transformer. We can either wire the leads directly to the board or add in a 12 V power plug receiver. As in the above cases, this will allow us to swap parts around as needed without worrying about soldering wires every time. This plug also has the advantage of containing both positive and negative power in one plug. We will only need one power plug on our circuit board.

Overall there will be many inputs and outputs on the board. We will have four 3.5 mm microphone lines in, one 3.5 mm audio line out, eight speaker terminal connectors for input and four for output, and one power plug. These will all have their own dedicated spaces for soldering directly to our PCB. Each of these parts were chosen for their ease of use and their readiness to allow part swapping very easily. Images of the three types of plugs that we will be using are included in Figures 4.7, 4.8 and 4.9. The only other inputs into the board will be soldered wires directly to the PCB.

Figure 4.7
12V Power plug

Figure 4.8
3.5 mm plug

Figure 4.9
Speaker Connectors

4.3.7 PCB Inputs and Outputs

In Order to connect our PCB with all of our components there will need to be numerous inputs and outputs on board. Each component will have its own needs and requirements based on its own interaction with the PCB, and we have taken that into account. The external component wiring diagram is shown below in Figure 4.10. Any component that is not directly contained on the circuit board is show in this diagram. This is how we will hook up each of the parts that we will be using. The diagram also contains information regarding the types of connectors needed and the quantity of each connector. This information will be further explained in the following sections.

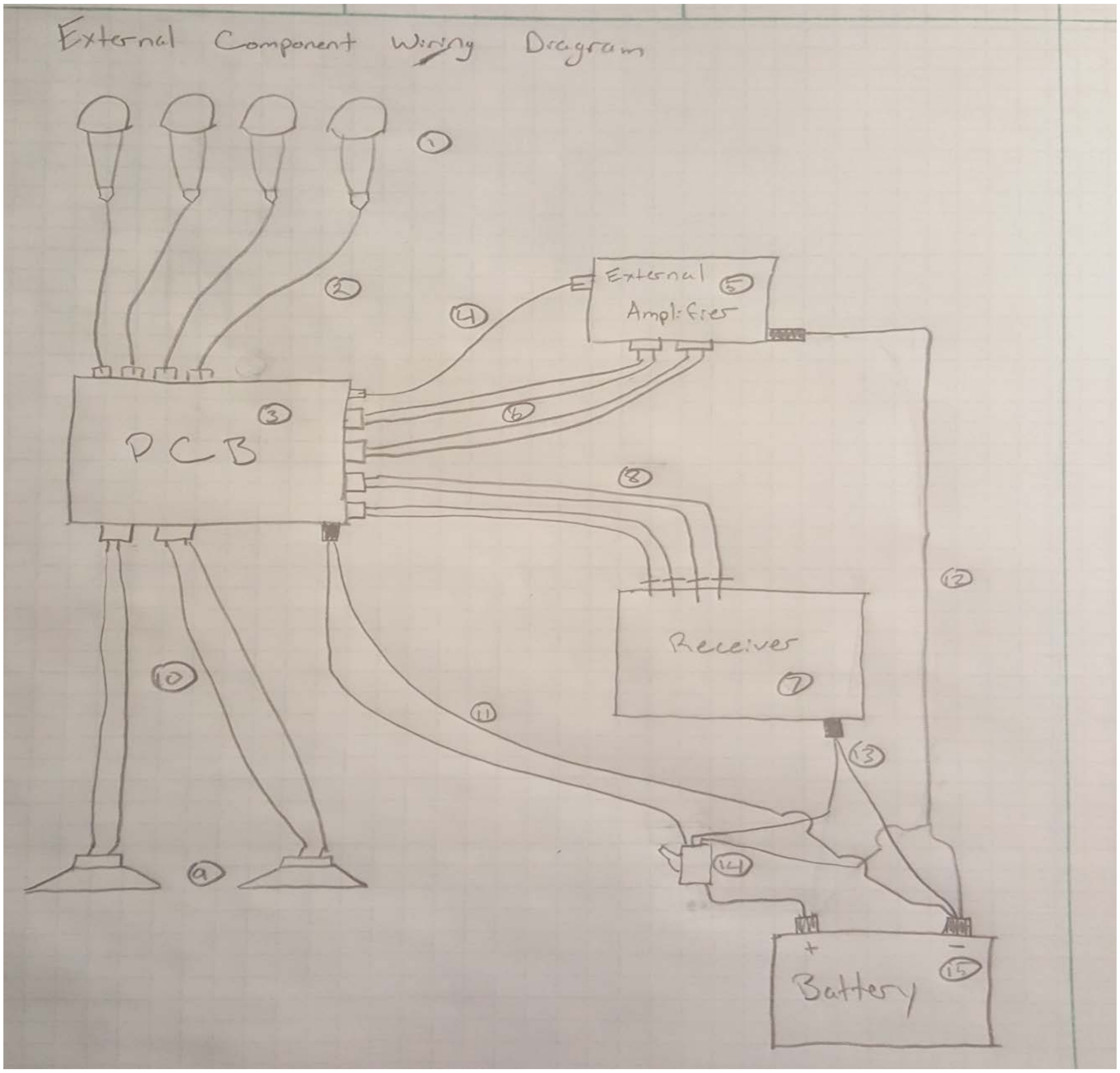


Figure 4.10: *External Component Wiring Diagram: 1. Microphones (4) 2. EXL to 3.5 mm cables (4) 3. PCB 4. 3.5 mm to 3.5 mm Aux cable (1) 5. External Amplifier 6. Speaker wire (4) 7. Receiver 8. Speaker wire to individual pin out grabbers (4) 9. Speakers 10. Speaker Wire (4) 11. 12V DC plug to +- (1) 12. 12V DC plug to +- (1) 13. 12V DC plug to +- (1) 14. Power Switch 15. Battery*

4.4 DSP

The Texas instrument OMAP 3535 has been chosen to be the main digital signal processing unit in the emergency vehicle detection system. With its multiple input ports and quad core design, the information from each microphone will be processed simultaneously for the fastest results.

Each core will be responsible for taking the bit stream for each ADC unit and breaking it into sample lengths of .0375 seconds. Each of these samples will be taken and transformed using a discrete Fourier transform in order to isolate the fundamental frequency of the sound wave. This process will first be done using built in function calls specific to the part, however, a unique algorithm may be written to optimize speed and functionality.

The pin array on the OMAP 3535 will require a unique socket for mounting it to the PCB. This will need to be ordered with the part. This socket will allow the analog digital converter to be connected to the SPI type interface that is needed. This interface will need a voltage regulator to keep the supply voltage to the OMAP processor logic at 1.35 volts and the OMAP core logic supply at 1.15 volts. The clock, CMD, and memory stick I/Os can operate at 1.8 volts or 3.0 volts, which will also need to be regulated.

Once the signal has been transformed into the frequency domain and the fundamental frequency of the sound wave has been found, the information will be passed to the OMAP 3525's internal ARM processor. There the decision logic will decide if the frequency detected matches the known patterns of the sirens. An output of the processor will be connect to the switching unit through the PCB to allow for the playback of the saved audio file.

For the software portion of the digital signal processing stage in the emergency vehicle detection system, the main goal is to extrapolate the frequency of the sound being heard and determine what microphone the sound was most intense. Figure 4.10 below is a flow chart laying out the digital signal processing stage of the emergency vehicle detection system. To begin, the bit steam coming from the analog to digital converters will need to be split into small samples. These samples will be roughly .0375 seconds in length. This length may need to be adjusted, most likely to a shorter time frame, to ensure all of the necessary data is found. These samples will be processed on one of the four cores in the OMAP 3525. The multi core proccing will help in deciding what microphone had the highest amplitude of the sound. With the blind spot built into the microphone array, ideally, one microphone will have no input. This will make the bit stream into that core all 0's. From that point on, the system will know the general location of the sound with 180 degrees of accuracy. Further inspection of the quantization levels in the bit stream will narrow the location of the sound down to a quadrant. Figure 4.11 below is a flow chart laying out the digital signal processing stage of the emergency vehicle detection system.

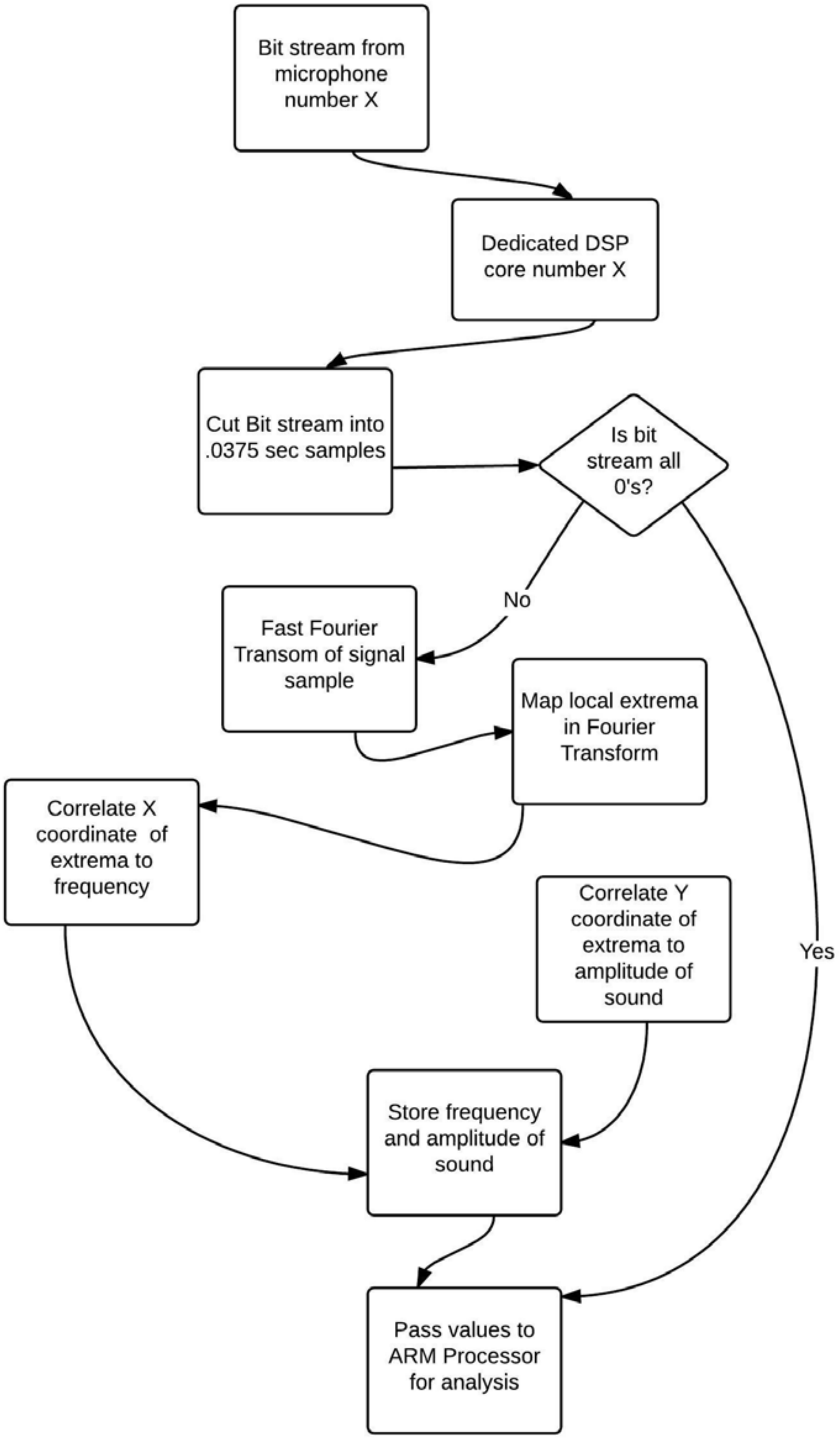


Figure 4.11: *Flow chart of DSP section of software design.*

Once the bit stream has been sampled and sent to the core dedicated to that microphone, the sample will be put through a conversion loop to convert the 8-bit binary stream into the Fourier frequency domain. Once this discrete Fourier transform has been performed, the fundamental frequency of that section will need to be found. This can be done by mapping the array holding the fast Fourier transform of the signal and finding the local extrema. These local extrema will correlate the sample number to the fundamental frequency of the tone. While the amplitude of the sound will correlate to the amplitude of the extrema.

Once this fundamental frequency is determined, the next sample of .0375 seconds will be processes. These frequencies will be passed onto the decision logic section of the system, residing in the dedicated ARM processor in the OMAP 3525. This is where the system will determine the location of the sound.

4.5 Switching Unit

The switching unit is a critical part of the design that has potential to destroy the audio quality and low interference that we are trying to achieve. There are two ways of designing the switching unit discussed above, and testing these two designs later will determine which of them will have less interference with our audio signals. The first method is a purely digital method of switching. It will rely on a processor that takes in both audio lines and then proceeds to run a code that determines which method of switching will be used. This method can either use the processor that we have on board or use its own processor to deal with the small bit of code that we are dealing with. For the purposes of speed, it makes the most sense to use a separate processor so as to not clog up the cores that are being used for EV identification. The second method of switching would be to build a physical representation of the multiplexor. This method would not require any processing power or coding. All of the components for the second method could be set directly into the PCB.

There are positives and negatives for each of these design methods. The first method allows us to simply write code and flash the processor. This would allow us to modify the code if we needed to down the line, or add in more functionality if we wanted. It also limits the strength of the signal that we would be able to send in to the system as it might be high of a voltage for the processor that we choose. The processor option might also cause disruptions in the sound from both sources as it repeatedly cycles. This is something that we would want to avoid. In the physical option we could avoid the cycling mess. Our circuit would not be running any code so there would be need instantaneous switching of the two sources. This method is also more robust in that it only uses parts that are necessary to the operation of the switching unit, there would be no extra power available. The merits of each of these units will be thoroughly tested when there is a system built ready to test. The testing will be outlined in a later section.

Regardless of the process used to perform the switch, the logical wiring of the circuit will be the same. In Figure 4.12 below, the logical wiring of the switching function can be seen through the use of multiplexors. Whether this is a digital or physical multiplexor has yet to be determined, but the code will be the same regardless. In the image below it can be seen that there are four selector line spots, yet they are all drawn from the same source. This is due to the possibility of variation in the lines causing the multiplexors to behave differently. Right now we have them as being wired to the front left positive speaker wire in order to cause a similar behavior from each multiplexor, but we will potentially wire them all to a constant voltage selector line straight from the processor, so that we do not get any variation in the signals. This will be determined once the PCB layout has been completed sometime next semester.

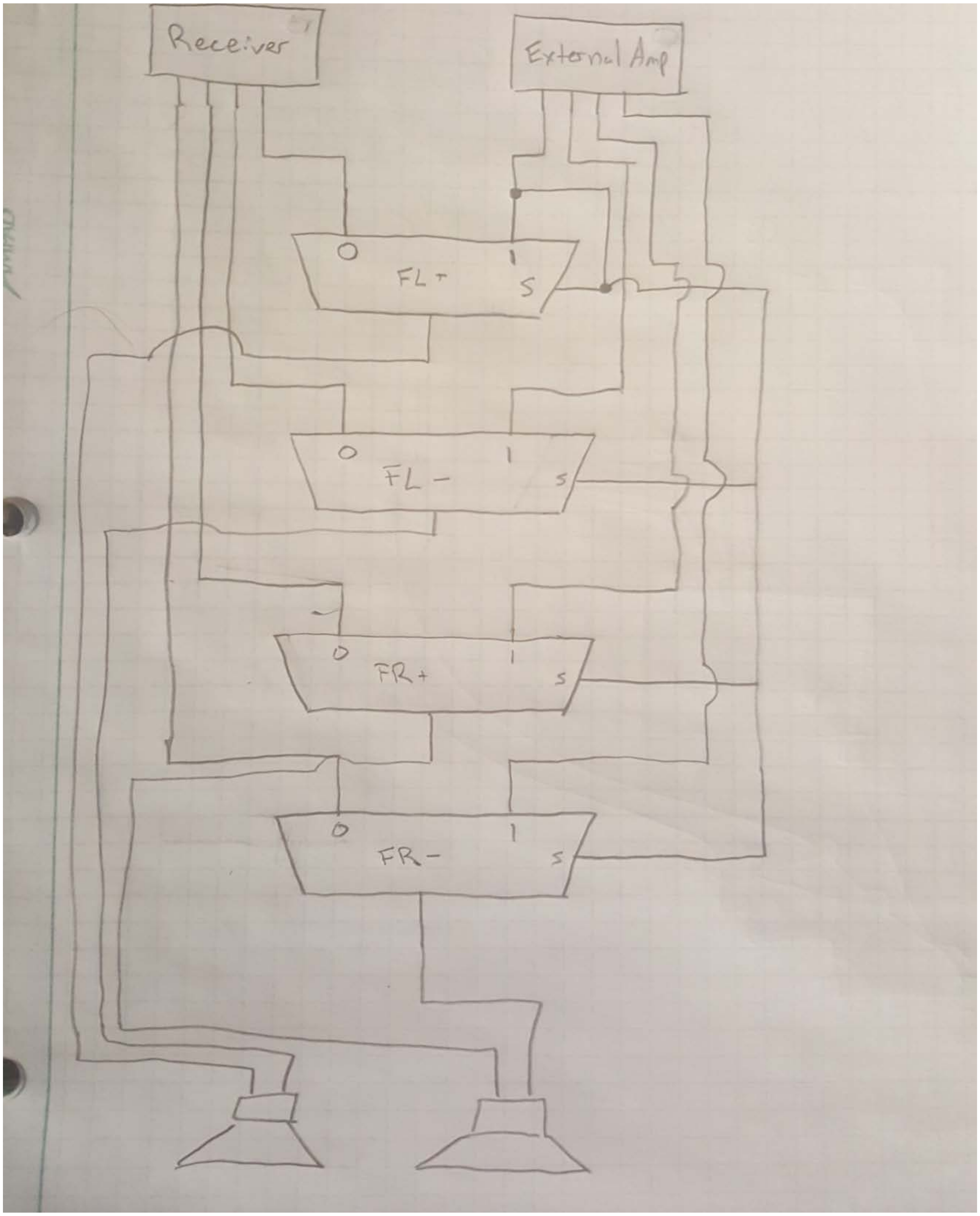


Figure 4.12: *Switching Function Wiring Diagram. The interface between the PCB and the external units is not shown here. This is shown in Figure 1.6. The External Amplifier, Receiver and speakers depicted at the top and bottom of the figure will be on the circuit board, but everything else will be.*

4.6 Audio Interface

4.6.1 Receiver Design

Due to our decision to go with a car audio receiver, we are not going to be able to use the simple audio in and out plugs that would have come with a home audio receiver. This means that we will need to work with the pion outs that are on the back of the receiver. There are two methods by which we can gain access to the pins; one is to directly wire to the pin outs with pin connectors, and the other is to find the plug that are needed for each of the connectors and then use the actual wiring that was intended. If this option is possible then we will pursue it, but after a moderately intensive search on the internet, it does not appear that the parts necessary are readily available. Not many people need the end of the connector that I am looking for. Our other option for obtaining this part is to look for a scrapyard that has a car with the appropriate connectors in it. This might be our best bet when it comes to wiring the receiver.

The receiver will be hooked up just at it would be in a car. There would be a 12V power in running to its respective port. This power in will be connected to the positive terminal on the battery. The ground of the receiver will be connected to the negative terminal of the battery. The rest of the interface will be done through direct connection to the receiver. Because we are using the car receiver there is already a radio and a CD player that we will be using for our car generated audio when testing. There will be no need to attach an external audio source in order to play music. This will greatly simplify the wiring required.

The output of the receiver will be run almost identically to how it would be run when installed in a car. There will still be positive and negative wire outs for each speaker in the car. Each of these wire outs will be run into parallel multiplexors that will perform the audio switching that is required of them. Because the wires are already amplified when they leave the receiver, there will be no need for these audio signals to pass through an external amplifier after going through the switching unit, unless we determine through testing that the switching unit provides distortion on the higher voltage input signals that would not be present on a lower voltage signal that is then amplified, but this will have to be determined through testing.

As stated before there will be wire outs for the positive and negative terminals of each speaker used. Our EVDS can do this one of two ways. The first way would be to have the EVDS take control of all the speakers in the car. This method would be inefficient because the system would not be as readily transferable from one car to the other. Different manufacturers have different numbers of speakers present in their cars and having our system take over each of them would be a much more complicated matter. Instead we are simply going to take over the front channel of speakers and allow the rear speakers to continue with their operation as normal. The back speakers of a car are already set to a lower volume than the front speakers and with the absence of the front speakers entirely there will be a drastic change in the volume of the car audio that will allow our warning message to be heard loud and clear.

This receiver interface should allow for adequate testing of our system in its design phase, as well as easy transplant ability into cars in potential future production phases. It will accurately represent the interface of a car while still allowing us the flexibility to easily access the wiring in order to splice in our EVDS. Also by using a receiver that is extremely common on the car market today, our product can be extrapolated to work on a wide variety of platforms, not solely the test platform that we have designed for it.

4.6.2 External Amplifier Design

The external amplifier will be used in order to make the audio from our processor strong enough to power the speakers that we are using in the system. It will only need to power the signal from our processor and not from the receiver. This is due to the fact that the receiver has an amplifier internally that it will be using. This amplifier will be wired to the same battery that our EVDS and our receiver will be wired to. In reality they will all be powered by the car battery when installed in a car, so we will show that by having them all powered by a car battery throughout our testing. The amplification of our EVDS audio signals will come before the audio signals are sent to the switching unit. This will allow us to only amplify the audio from our system and not from the car audio. This is also contingent on our testing of the system. If it happens that our switching unit causes distortion of amplified signals, then we will have to send up amplified signals to the switching unit and then to the amplifier. This will not complicate our wiring any, in fact it might simplify it so that there is only one amplifier running on the system, thus reducing strain on the battery.

Our external amplifier will take in a single stereo in input, and output the same. This will be represented by a standard 3.5 mm headphone cable. Though in production there would be different ways to accomplish this, for our purposes there is not an easy way for us to create our own interfaces between products that would allow seamless interchangeability. Because the audio signals that we will be outputting will be in a mono-channel format there will be little reason to have multiple channels running to the amplifier and then to the switching unit. A single channel of sound that will go to each speaker available will suffice and provide the most consistent resultant playback for our purposes.

With the interface being a 3.5 mm headphone cable, one female headphone port will need to be on our PCB, and the input for our amplifier will need to also be a 3.5 mm port or something that is easily convertible. We could also use stereo plugs, but the size of a 3.5 mm cable is preferable to the multiple plugs of a 3.5 mm cable. After reaching the amplifier in this format, the signal will be output in standard speaker format, with a positive and a negative for each channel we need.

For our purposes we only need one channel to run to multiple speakers. Though this might cause a little loss in power, we will account for that when adjusting out amplifier output.

In our first method of amplification we address the issue of a powered off audio system in the car that is running our EVDS. If someone is driving with their receiver off, which is a rather common thing to do, by having our audio powered externally there is nothing stopping our device from playing sound through the speakers. Because the speakers are passively power, meaning that they have no built in power supply as is with all car speakers, they rely solely on the power of the incoming audio signal. Even if the car receiver is off, our natively powered audio from the EVDS will have enough power to make the speaker's playback the audio we want. Due to the process of returning to car audio after every playback in our switching unit, this will return the car to its silent off mode whenever the EVDS is not attempting to playback audio. This will not change the normal operation of the car at all.

Our primary method of amplification will be the lease invasive on a car when installed. If we had to switch to the secondary method of amplification after the signal has gone through the switching process then there might be more wiring required in order to allow for the car audio to function properly with regards to volume. Both methods can be made to work. But the second method will only be used if, after testing, the first method does not work as expected.

4.6.3 Speakers Design

Due to the way that we have our speakers integrated into the system, there is very little that needs to be changed in their standard wiring configuration. Each speaker will still get the same amount of wires coming to it, the main difference being the starting point of said wires. Instead of coming directly from the receiver, they wires will come from the switching unit built into our PCB. This will allow us to use the standard wiring configuration rather than running some parallel system in order to allow our audio switching to happen.

The speakers we are using are very similar to car audio quality to begin with, so there should be no interference problems with the way that we are using them. They are built to be amplifier powered and that is how we will be using them. Though the speakers we have decided on are going to be home audio music monitors, speakers are so similar across industries, that as long as we are careful in our selection, the speakers will perform the same way. The speakers we have chosen, though home audio, match the specs of a the majority of automobile speakers, and across the automobile industry almost everybody uses the same format for their speakers, and that is not set to change any time soon, except for a few erroneous outliers. This makes our design all the better because it can be applied across the board to any car and still be made to work.

The final design of our system will have the front speakers directly wired to the switching unit, and all of the channels aligned to the respective channels of both the receiver channels and our processor's channels. The sound will already be amplified by either the receiver or the external amplifier in the case of our generated audio. There should be no playback problems with this design. This will all be extensively tested for distortion as we are trying to minimize impact on sound quality throughout the process of our intrusion into the audio system of the car.

4.7 Power Supply

The design of our power supply will be fairly straight forward. Because it is going to simply be a 12V car battery there will be no need for an AC to DC transformer. There are two reasons for keeping it as a battery rather than a plug in device. Reason one, mobility; by having the device be battery powered we will be able to test it in the field without worrying about extension cords and other restraints that plug in devices natively carry. Number two is the real life application of the project. It will eventually be in a car, and the only power source in a car is a 12V battery nearly identical to the one that we are going to be using to power our system. This will eliminate the possibility of our using more power than is available for use in a car.

There will be very few wires actually running to the battery. In a car, there are two large wires that carry all the power the car's systems require to the various systems that require it. Our system will run in much the same way. There will be one wire leaving the positive terminal of the battery that will split into three ends. These ends will connect to the receiver, the external amplifier, and the EVDS PCB itself. The last of which will power the microphones, LED, processor, and any other systems that are required. Then running from the ground of each of these devices, several wires will run back into one that will return to the ground terminal on the battery. We will have the necessary components to maintain a proper charge in our battery so that we can recharge it after testing, and even have it plugged in while testing.

Once the power supply is applied to the system there will need to be a voltage regulator applied in order to protect some of our components. The receiver and the amplifier both utilize 12V DC power inputs, but out processors and circuit board will not have such a high voltage tolerance. This means that there will need to be a voltage regulator built into the circuit board. This is a problem that is easily rectified in our PCB design.

This design of the power supply will be the most effective in transferring over to a real life application of our EVDS. It closely models that of an actual car and like many other aspects of our design focuses on transplant ability. If used effectively there is no reason why this unit cannot be picked up and placed into any car and be powered by its battery.

4.8 Case

We have spent a lot of time researching our case design and make, and have come to one solid conclusion; it does not matter. We were originally going to design and 3D print a case that would hold each of the parts that we needed in exactly the right way. After some discussion, we determined that it would not matter how we packaged it, because in production, this would just be built into the vehicle that it would be used in. The circuit board would be added into the many already inside of the car and there is almost no chance that our final design would ever end up in a car. So rather than spend our time designing and printing a case to hold everything we determines that simply using the mounting solutions from the parts themselves would work perfectly. The job of our case is primarily to hold the microphones in place with reference to each other, and secondarily to hold the other parts in an easy to access configuration. The microphones that we are using come with microphone holders that screw onto standard tripod screws, which are 5/8-27 screws. These are easily acquired through most hardware stores and will serve our purposes well. Our plan is to simply attach these screws to a square piece of wood and mount microphone stands to it. Then we can fix them in position and we have our mounting platform for our microphones. Underneath the microphones the receiver will be mounted using the car mounting hardware that it came with. In between the microphones is a space that will be plenty large for our PCB. The last two components that will need to be accounted for are the battery and the external amplifier. The amplifier will be mounted beneath just as the receiver is, and the battery will most likely be on its own. In Figure 4.13 and Figure 4.14, a rough sketch of the layout of our mounting board can be seen.

There are several distinct advantages to this style of mounting. The first being its robust nature. A wood and metal mount will last us much longer than a 3D printed mount and we do not want to be redoing our mount halfway through the project due to a broken mount. This is critical because with our testing methods as mobile as they are, we will be moving our system around a lot, potentially causing harm to the case and the components inside. Secondly, the size that we are looking at would be very intensive on a 3D printer and take up a lot of print space. This is something that we want to minimize, so this handmade mount makes a lot more sense. The size factor is something that we could have done without. If we had elected for a smaller size we could have made the project work, but in our case, the further apart the microphones are, the better for our accuracy. It also allows us to build an all in one unit rather than multiple units, as was our original design. With these considerations taken, it does not make any sense for us to 3D print a case when a more effective one is very easily built by hand.

In a realistic application of this device, everything could be done as we have done it, but there would be other things that must be taken into consideration. First, this is being designed so that someday the entire unit can be built into the car without an external unit. The only changes this will require, are modifications to the location calculating geometry based on the new placement of the microphones. The receiver would also not be mounted to the same board as the PCB and external amplifier. Everything would be contained behind the dash, and the receiver and speakers would be installed into their locations in the car. The design discussed above is purely used for our design and coding process, so that we have a standard way to measure everything.

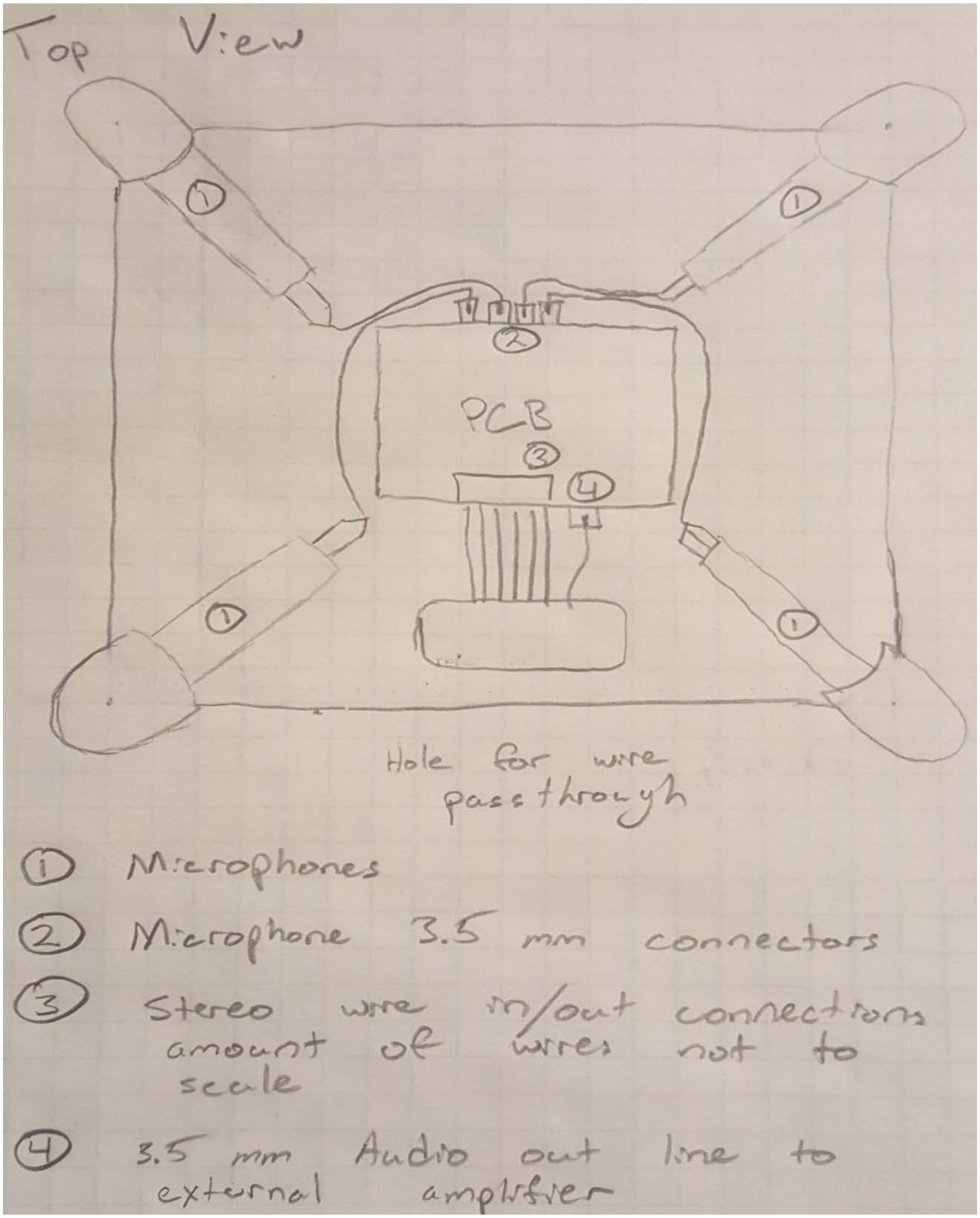


Figure 4.13: *Top view of the mounting board.*

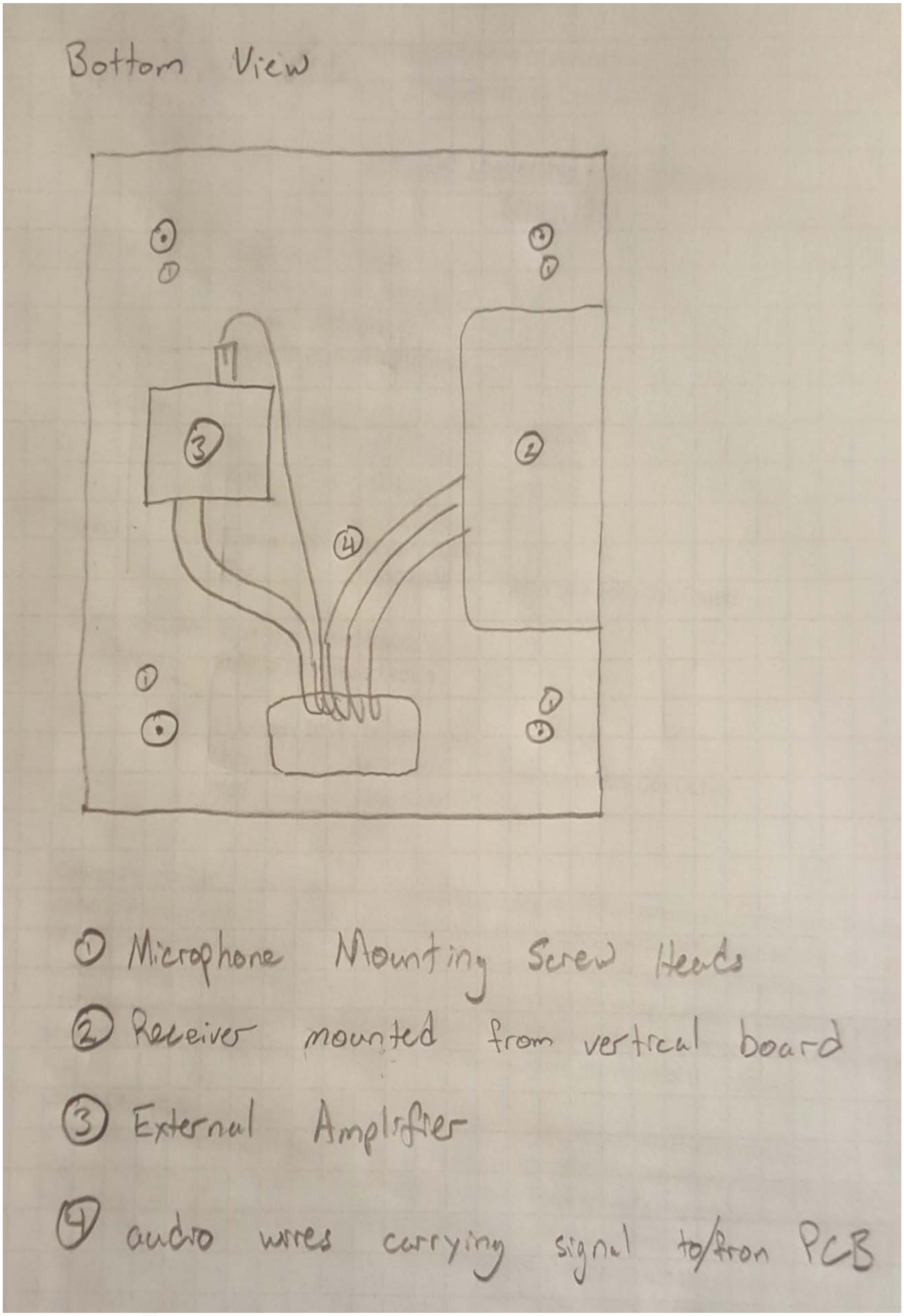


Figure 4.14: *Bottom view of the Mounting Boar*

5.0 System Prototypes

Over the design cycle of this system there will be a total of three prototypes that will represent the three different stages of the system. The prototypes will represent functioning systems, with unfinished parts and casing decisions.

Prototype 1: The first design prototype will be the fully functioning emergency vehicle detection system, except without soldered boards or PCB. For this stage of the prototype, the digital signal-processing unit will be used through a test socket for the ball grid array. This test socket is being ordered specific for the type of ball grid array found at the bottom of the digital signal-processing unit. The Butterworth filter will be comprised of 1 percent error capacitors and resistors plugged into a basic white breadboard. The op-amp that will be used for the fifth order Butterworth filter will be the TL-084.

This model contains four op-amps on a single processor, because of this a single unit can handle the entire low pass or high pass fifth order Butterworth. Since the desired filter is a band pass, this means there will be two TL-084 units used for each complete Butterworth band pass filter. Since each No wires will be soldered at this point. The analog to digital conversion units will be soldered with wires to make the connections to breadboards easier. Because of the relatively low unit price of these units, they are expendable to solder on due to cheap replacement.

Each analog to digital conversion processor will be connected to one of the four inputs on the digital signal-processing unit. For the first prototype this will be possible through the use of a test socket for the ball grid. The digital signal processing unit’s output will then be connected to a digital to analog converter, in order to properly announce the direction of the incoming emergency vehicle through the speaker. At this point the output connection will be through another 3.5mm audio input that will run through the 2-1x mux, which will be deciding, between the detection system audio, and normal radio audio.

The radio in the case of the first prototype will not be an actual car, but instead just an old car radio. This way the wires to the output can be spliced to include the addition of the mux to choose between the detection system and current radio. This prototype will be tested in the confines of a lab initially. Once the prototype shows it can determine correctly the direction of an approaching emergency vehicle up to a certain level of accuracy from the set distance in an artificial environment, and then the prototype will be tested in a real environmental setting.

For the real environment test the device will be placed inside a car and driven near a fire station. Then the device will wait there until a vehicle leaves the station with sirens. This will allow confirmation whether the system works accurately in a semi controlled setting, which knowing the direction the emergency vehicle will be leaving from before time. For this point in the testing the emergency vehicle detection system will not be spliced directly to the car audio, it will still be outputted through a make shift amplifier and speaker. All the parts for the first prototype are listed in Table 5.1 below.

Part	# of Parts
Tl-084 (Butterworth Filter)	8
ADC081S021 (Analog to Digital Conversion)	4
One Percent resistors (Butterworth Filter)	Varies with needed gain per stage
One Percent capacitors (Butterworth Filter)	Varies with needed gain per stage
OMAP3525 (Digital Signal Processor)	1
Ball Test Socket (Digital Signal Processor)	1
Shure sm58 Microphone	4
XLR female- 3.5 mm audio jack adaptor	4
Wires for breadboard connections	Varies
Output speakers	2
Amplifier	1
Receiver	1
DAC unit	1
Power Supply (12 Volt Car battery)	1
Lm2576 Voltage Regulator (5Volt)	1

Table 5.1: *List of components for initial prototype.*

Prototype 2: After the first prototype passes all the tests for efficiency and accuracy the second prototype will be constructed. The difference in this prototype will be come from more secure connections. The PCB board will be ordered at this point, so there will be no more loose wire breadboard connections. The PCB board will have five 3.5 mm headphone input located on it. Four of these will have connections to the band pass filter analog to digital conversion path that leads to the inputs of the digital signal-processing unit. The fifth 3.5 mm headphone jack will serve as the output for the decisional logic unit sends as a message to the driver. This message will relay to the driver that an emergency vehicle is approaching from a specific direction.

Now all the connections from the digital processing and analog to digital conversion devices will be secure. However, this will still not be the final prototype because the output speakers, as well as the input microphones will not be in the final casing. At this prototyping stage the input microphones will be loosely kept together on a piece of plywood at the same distances apart that the final mounting of the system will have. The output speaker and simplifier system will still be the makeshift speaker interface, the wire outgoing from the emergency vehicle detection system will be spliced by a 2 to 1 mux that will choose between the speaker interface sound, and the signal from the emergency vehicle detection system.

For the testing of this model of the prototype a similar approach will be taken as in the first model. The system will be first be set up and tested in an lab setting, to make sure that the desired goal is achieved in a lab setting. Next, the system will be brought outside to a quiet area. 50 yards away from where the emergency vehicle detection system is there will be a large speaker that will emit the sound of one of the emergency vehicle sirens. If the emergency vehicle detection system can successful notify when a siren sound is being emitted, as well as not being activated by any other types of sounds, then the system will be set up in a car to test outside the fire station again. The reason for the set up close to the fire station is to ensure that an emergency vehicle will be leaving, and the direction of the emergency vehicle will be known. Once a certain number of successes are seen from the system the final casing for the final prototype will begin being assembled.

Differences from prototype 1:

- PCB board has been implemented into the system with slots for five 3.5 mm audio inputs.
- Input microphones will be set up on a test mount at the same set up angles as the final design will implement
- Testing will take place in three different stages, including an outside environment that is not inside of a car

Prototype 3: The third iteration of the emergency vehicle detection system prototype will be the final displayed system. Like the prototype 2, the PCB board will have all the connections between the input ports, output ports, filters, and all three processors. At this point, the final TL-084 op-amps, the ADC081S021 analog to digital conversion unit, the OMAP3525 digital processing unit, and the digital to analog conversion unit will all be soldered onto the PCB board. There will be no loose connections on the board. The PCB board will have the five 3.5 mm inputs, which will securely hold the connection coming from the input microphones, and output microphone.

The input microphones will be in there final mounting placements. They will be secured onto a piece of sturdy wood. In the middle of the microphones will be the PCB board inside of its final casing. The output of the system will run to the amplifier and speaker set up.

For the final prototype, the system still will not be installed onto a car. This is because the system is designed to be directly integrated into the car wiring system, not just be an added on feature. Because of this the final system will be shown in a cart type set up, the wood with four-microphone set up will be mounted on the top of the cart. The speaker and amplification system will be set up inside the card. The testing will be done at all the specified distances to check for accuracy at certain distances.

Differences from Prototype 2:

- Input microphones set up in final mounting situation
- All processors will be soldered onto the PCB board to ensure no loose connections.
- PCB will be located inside casing
- Final speaker set up in cart

6.0 Test Plan

6.1 Microphone Testing

To begin all testing, a small test microphone was used on a breadboard to test the filter designs, microphone capabilities, and the basic concepts of the system. The SparkFun Sound Detector was ideal choice for the testing because of its small design and cheap cost.

The sound detector was connected to the breadboard to test the filter capabilities. Those results are discussed below. However, at that time, the testing of a microphone's pickup capabilities were tested. This was done by playing a tone at different distances and observing the response of the microphone on the oscilloscope. The microphone was able to discern between the noise of people talking in the room and the tone being played for testing even when the tone was being played over three feet away. One issue that was observed was that the angle of incidence between the sound wave and the microphone needed to be almost 90 degrees. When the tone was moved from directly in front of the microphone to roughly 60 degrees of center, the response of the microphone was decreased.

These results help to show proof of concept on a small scale. Given the unsophisticated microphone and the fact that the sound was being played out of a cell phone speaker during testing, the response of the microphone with a sound three feet away was more than acceptable to move on to the next phase of testing.

The next phase in testing will be to perform the same tests as before, except this time the microphone being used will be the Shure SM58. This allows for the opportunity to learn the wiring of the microphone, as well as its response capabilities. This microphone should be able to pick up the sound a much further distance than the SparkFun Sound Detector. The distance will need to be tested to find the minimum dB level desired. This level will be implemented in the filter discussed below.

Next, the roll off of the microphone needs to be tested as well. To do this, the sound will be played directly in front of the microphone, then the sound will be played from an angle greater than 60 degrees but less than 90 degrees from center. These results will be compared to find the dB loss with respect to angle of incidence. This information will be used to write the decision logic code when the amplitude of each microphone is analyzed.

Now that is has been proven the Shure SM58 will suffice for the emergency vehicle detection system, it is time to test the directional capabilities of the design. The directional capabilities of the system will be determined by the physical placement of the microphones and the decision logic taken in the software, both of which are equally important. These two pieces will come together and depend on one another to accurately determine the location of the emergency vehicle. The physical location of the microphones on the car are discussed here and the software side of this testing is explained below.

The distance between each microphone and the angle each microphone points out of the center are the two variables that will define the physical placement of the microphones. To begin testing, the ideal scenario will be created. This system was designed with the end goal of each microphone being built into the four corners of car during manufacturing. Meaning that the microphone will be anywhere from 1 meter to 3 meters apart. Based on an educated guess, and to create a starting point for testing, the microphones will aim outward facing 45 degrees right and left of the centerline of the car. This design will be tested to determine accuracy and the blind spots of the system.

If this design suffices, it will be implemented. However, if it does not work, further testing will be done. These tests will vary the distance between each microphone first, and then the angles they point. This two-step process will allow for single variable testing. The first distance to be tested will be with the microphones almost touching each other. This will act as another extreme scenario for testing. With the microphones this close, the amount of overlap in the pickup patterns will be maximized. As the microphones are moved, the amount and range of “blind spots” changes as well. These “blind spots” are created by the roll off of each microphone's pickup pattern when the sound is created more than 60 degrees from center. Referencing Figure 3.2, moving the microphones from being centered on the car to being on the edge, changes the area of the blind spot drastically. This area can also be moved to different spots around the vehicle by changing the angle each microphone is pointing. For testing, the distance between each microphone can be increased until the desired result is recorded.

Once the distance between the microphones have been determined, the angle of direction will be tested. These angles will change the software in the decision logic. Any angle will technically work, however, testing will be done to find the ideal angle for this application. Because the driver will be looking forward, and the main goal of the emergency vehicle detection system is to enhance the driver's awareness, more emphasis may be put on the detection of sounds in the rear and to the side of the vehicle. By varying the angle of each microphone, the system can be tuned to pick up sounds better in the rear and sides than in the front of the vehicle.

6.2 ADC

In order to test the analog to digital conversion unit in the emergency vehicle detection system is to create Mat lab codes to show proof of concept. In this paper the analog to digital conversion will be considered successfully tested once the output of the analog to digital conversion unit successfully resemble a converted sine wave. This is because the microphone will receive and process the sound as sine waves, which will then be converted to a digital signal. So if the processor can show that it can accurately convert a specific frequency sine wave, then it can be ensured that the right signal is being delivered to the decision making unit, as well as the digital signal processing unit.

In the first step a mat lab code will construct that will take in a sine wave input and convert it into a digital signal. The sampling rate of the mat lab code will be 50 ksps to ensure symmetry with the actual analog to digital conversion unit sampling levels. The Bit resolution will be set to 8 bits, and the maximum voltage level will be 5 volts (3 volts will also be tested in case a lower supply voltage level is needed). Since there are four microphones, there will be a separate analog to digital conversion unit on each microphones path. However, since all input signals are the same noise, the same unit type and code will be used for each analog to digital conversion unit.

Once the mat lab code proved that the bit resolution and sampling rate of the analog to digital conversion section of the system will work for the desired levels, the next step was to compare the results with a known analog to digital conversion signal. This test was run in the lab similarly to the filter test. The SparkFun sound detector was plugged into the breadboard. The output of the sound detector was then run to the input of the ADC081S021 device. The sampling of the device was set for 50 ksps and the bit resolution set at 8 bit. Next different tones were exerted for the sound detector to pick up. The tones would stay at the same frequencies for long periods of time. Because they stayed at the same frequency, we were able to record the output of the ADC081S021 and view the results on the oscilloscope. The results of the output were captured and compared to that of known converted sine waves at specific frequencies.

If at the end of this test all of the frequencies are converted to the proper digital output signals are found to match the desired conversion signals, the next step is to test a copy of the siren signal through the analog to digital conversion unit. The ideal conversion will be found through the use of a mat lab code. The sirens all will have a digital signal. This signal will be run through the mat lab test code and display the correct converted signal in mat lab. We will know the signal is correct because the mat lab code will already have properly converted an arbitrary signal correctly against a template.

Once mat lab proves that the correct signal can e reproduced from the selected siren signal, the digital copy will then be played through a microphone. The same set up will be sued with the microphone receiving the stereo output, with a direct connection to the input of the analog to digital converter. The analog to digital converter output will then be connected to the input of the digital oscilloscope. The siren sound will be played through the output microphone, most likely from a phone, and then on the oscilloscope it will be determined whether or not the converted signal properly matches the correct waveform.

After the analog to digital conversion unit has been proven to work efficiently, not only on its own, but also in interfacing with the microphone and digital signal processing unit, then the PCB board will be designed to leave space for the ADC081S021 unit. Once the final board is received, the processor will then be rechecked with the new connection on the finish PCB unit to make sure no wires were misconnected in PCB fabrication.

The final testing of the analog to digital conversion unit will come with the testing of the overall emergency vehicle detection system. The system as a whole will be tested to make sure that all parts work together correctly. In this stage it will be ultimately determined whether or not all units in the system are operating at frequencies that lead to the ideal determination logic. If it is found that the sampling rate of 50 ksps is too low, or too high, for proper decision magic, then the ADC081S021 can be tuned to a different sampling frequency. Once it is determined that the processor is compatible with all other units in the system then testing of the unit is complete.

6.3 Filter

The testing of the Butterworth filter will be done in three stages. The first testing stage will be by hand and simulation. In order to determine the correct components for the required pass band the output levels of the filter will be mathematically computed. Following this, simulations will be run on LT Spice in order to ensure that all components will work as expected. Once the LT spice simulations ensure that the Butterworth has an out of band filtering level of 100 dB per decade the first op-amps will be ordered.

Once the first op-amps are available each stage of the Butterworth filter will be individually checked. If the stages each perform at the correct cut off frequency the fifth order filter will be assembled for both the low pass and high pass stage of the band pass filter. Each low pass stage and the high pass stage will use its own TL084 unit. Each unit contains four op-amps so it will be able to handle all stages of the fifth order filter. After the testing shows that both the low and high pass stages have a filtering of 100 dB/decade the will be combined in series to complete the band pass filter.

The first band pass filter prototype will be comprised of loose wires connected to a common breadboard. A DC supply voltage will provide the necessary positive and negative voltage source. The filter will then be connected to the positive terminal of a test microphone, the output of the band pass filter will run to an oscilloscope which will be set in the Fourier domain to better visualize the sound frequency amplitudes. Using a tone creator played through a phone speaker, different frequency tones will be played. First frequencies within the band pass of 300 to 3000 Hz will be played. Next, frequencies of 3 Hz and 20,000Hz will be played following. At each of these frequencies the amplitude of the frequency spike in the frequency domain will be checked to ensure that they are down 100 dB (or the proper ratio value if not on a precise decade) in comparison to the levels within the band pass

Once the filtering levels of the Butterworth filter have been confirmed to be at 100 dB per decade, the next step is to ensure that the output voltage levels will be within the constraints of the analog to digital conversion units voltage input levels. Since the analog to digital conversion unit has input levels that range from 0- Vcc+, the output voltage of the Butterworth will never be above the maximum input level, this is because the rail of the op-amp that is the active component of the Butterworth filter has a maximum level of Vcc+ that is connected to its positive terminal. The next concern is to make sure that the signal is not amplified to a level to high that will cause an issue discerning the difference between the different amplitudes of the received sound. The tests for the amplitude will be run with the same set up as checking for the filtering levels. However, there will be no need to check each stage of the filters amplification. The only amplification that will need to be checked is the level at the output of the band pass filter. If it is found that all microphone signals are being amplified to the rail of the op-amp so that it causes issues with decision-making then the amplification stages inside the Butterworth filter will be properly reduced to lower the output filter levels.

Once the filter in the test set up is found to meet all the standards that are expected out of the filter the next step is to design the filter section on the PCB board. Once the board is received the filter will be tested to ensure proper levels after construction.

6.4 DSP

In order to test the digital signal processing of this system, the first step was to write some simple codes in Matlab in order to have proof of concept. There are three parts to what this paper is referring to as the digital signal processing portion of the code: bit stream conversion, frequency analysis, and passing of the values. Each of these parts has to be tested.

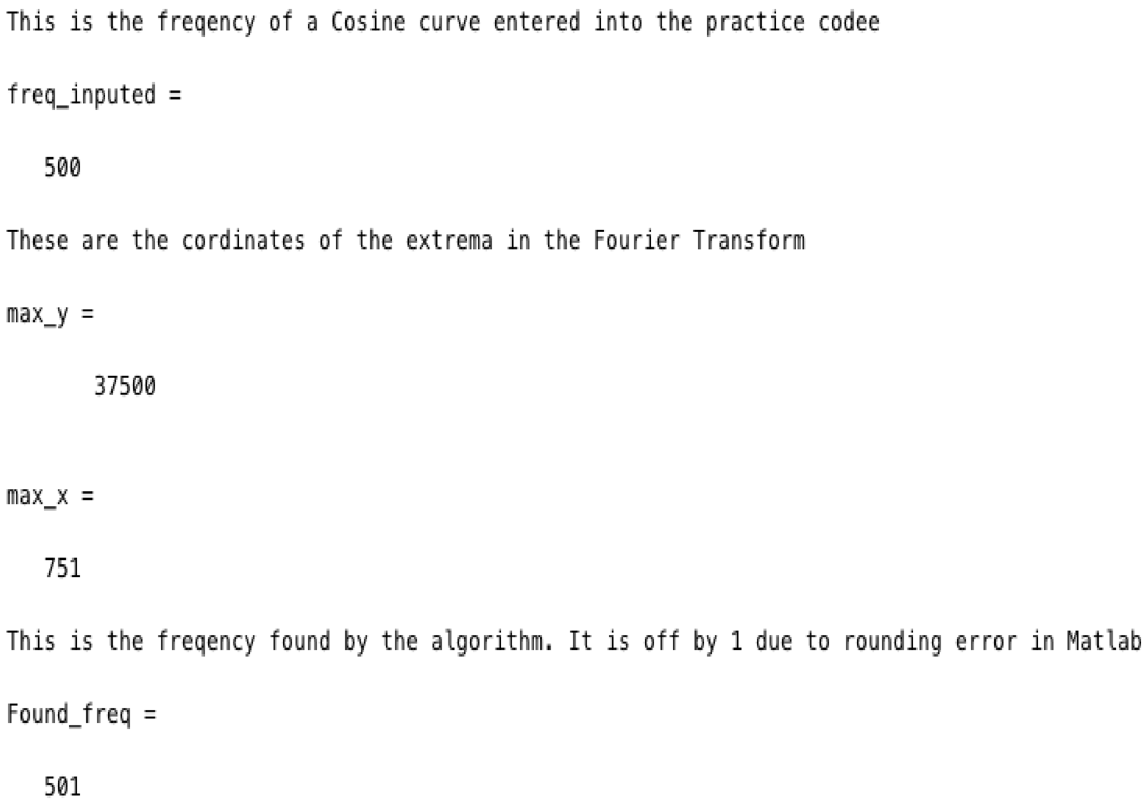
The first step in the digital signal processing process is to reconstruct the analog signal from the bit stream created by the analog to digital conversion process. This will be done by taking allotted samples of data and running the sample stream through a loop to convert the binary representation to quantization levels. Each sample will have eight bits, meaning that the loop will need to take eight digits, convert them, store that value, then continue until all the samples in the interval have been converted. At this point the digital signal is able to be read and interpreted.

The length of the interval sample will need to be tested. Taking the cycle times of the sirens into account, the sample time will need to be shorter than .0375 seconds. This number came from taking the fastest cycling siren, the piercer, and dividing it by two. By dividing it in half, the program should extrapolate a frequency of 725 Hz for the first half of the cycle, then the 1600 Hz for the second half. Because the piercer is the fastest cycling siren, this technique can be used on all the other sirens without any information loss. If a longer cycle timed siren is being heard, the process will be continued until a change in frequency is observed.

A sample code will need to be written to convert the bit stream in the cut sample into data that can be interpreted. This will be simulated by entering the bit stream of a known cosine curve into a conversion loop. This loop will take the matrix signal and break it up into individual sample points, 8 bits in length. This sample point will be converted into a decimal value that corresponds to the quantization interval it was assigned in the ADC conversion. This value is then plotted with the sample number as the x axis and the decimal value as the y axis. These plots will then be compared to the cosine curve that was put into the matrix to find any errors in the algorithm.

Once the sample of the signal has been recreated using the conversion loop, the processor will need to do a Fourier transformation of the signal in order to extract the information needed. Once transformed, the signal will then be run through a loop to find the local extrema. These extrema of the Fourier transform will correlate to the amplitude the frequency of the signal.

The coordinates of these extrema will be related to frequency and amplitude, but in order to determine that relationship some steps must be taken. Because the program was written in Matlab, the coordinates found by the extrema loop will correspond to the sample number taken not the frequency of oscillation. To find the relation between bin number and frequency, the program must take the sampling frequency of the ADC into account. The sample frequency will determine the number of bins and the relation between each bin. In the figure below, a cosine curve with a frequency of 500 Hz was sampled at 50kHz. This produced 75000 “bins” of information. This means that a value in bin 75000 would correspond to a frequency of 50kHz. Using this relation and simple proportions, the program can extrapolate the frequency at which the cosine curve is oscillating. Figure 6.1 shows the input and the output of the sample code.



```
This is the freqency of a Cosine curve entered into the practice codee

freq_inputed =

   500

These are the cordinates of the extrema in the Fourier Transform

max_y =

       37500


max_x =

   751

This is the freqency found by the algorithm. It is off by 1 due to rounding error in Matlab

Found_freq =

   501
```

Figure 6.1: *The output of a sample Matlab code produced to prove the idea of using Fourier transforms to find the fundamental frequency of a cosine curve.*

With these simple “hello world” type codes complete, the concept of using the Fourier transform to analyze the signal has been shown to work. The next step is to connect each of these pieces together to create one piece of code that takes the inputted binary stream and outputs the frequency of the oscillation forever sampled section. Once this code is completed, the final step will be to write the code to handle the continuous input that is required by the system.

After the code is completed, the system as a whole is ready to be tested. To see if the design meets the specifications, there will be a three part test. The first test will be to determine if the system can differentiate between sirens and the noises of the road. To do this, several sounds will be played before a siren to see if the system produces any false positives or negatives. The system will need to handle horns, screeching noises, and even music. These three sounds will be played and the system should not respond. Once the sound of any siren is played, the system should go into action. One final test will be performed with the siren overlaid with noise to ensure the system can filter noise properly.

Once these system has shown it can differentiate sounds, the next step is to test the range of the system. To do this, the previous test with the different sounds will be performed at different distances. The starting distance will be 50 yards, and will be extended by 10 yards until the maximum range of the system has been found.

The final stage of testing involves the directional capabilities of the system. To test this feature, the sirens and noises will be played at different points around the systems. Starting at the front, and moving clockwise every 30 degrees, the sound will be played and the system will need to identify the location of the sound. This will be done at a distance between 10 and 50 yards for testing purposes.

6.5 Decision Logic Testing

In order to test our decision logic we will have to do it in two parts. We will not have all of the required components completed when the code needs to be tested to test it in our system. First we will have to run it in a simulated environment where we feed it values and determine if the function works in distinguishing the sounds from each other. This test will need to be repeated at full speed with natural inputs to completely verify that it works. Because the decision logic both identifies and locates the sound, we will need to run each of these tests in at least two times so as to check each component individually. Each of the below tests will be conducted in a laboratory setting. A loudspeaker set to the appropriate volume setting will simulate our “siren”. The microphones will be used and connected through our DSP processor that we have designed and coded above. Once we have determined that this works in a laboratory setting we can assemble and repeat the tests in a non-laboratory setting using actual sirens.

The first version of the test will involve the code identifying a siren. This first iteration will only use one loop of the code, versus the four that will be in the final iteration, in order to test the individual identification ability of each microphone. Each loop will correspond to a different microphone, so cutting the code down to one loop will effectively test one microphone. In order to test the first loop we will start by flashing the processor with the code that is to be tested. Once it is up and running it should allow us to input a frequency value for the microphone input. Because we are purely testing the decision logic we will either hand input values, or have the computer auto-input values for us. Our test values will fall into several different categories to accurately test the decision logic coding. The first set of values will be random noise values that do not align with any siren. After multiple inputs in a row we should still be seeing the same thing and have the program asking us for new values, not sending out any signals indicating that there is a siren heard. If we can keep doing this indefinitely then we know that the device will not signal a false positive for siren identification. The next step will be to input a series of values that indicate a siren has been heard. These values will use the frequency analysis, yet to be obtained, to determine which values to input into the system. The values entered will be consistent with one of the four siren patterns that we are testing and should after a number of matching values in a row, trigger a positive siren response. This is critical because this is the main functionality of the decision logic, to determine if it hears a siren. We will need this function to work for each of the four sirens before we can call this test successful. The code will also have to be able to go back to a non-positive response when the values switch back to random noise. Once we can achieve accurate behavior for all five cases of the code, we know that our code works for at least one microphone to identify a siren.

The next part of the testing will determine if the code can perform its second functionality, locating the EV. This will be accomplished through comparing decibel levels and comparing them between microphones. In order to test this we will need to have created a decibel level vs. angle chart at some constant distance, and a decibel level vs. distance chart at some constant angle. These two charts will allow us to determine location and distance once all the microphones are added to the code. By entering one value for the siren, being the magnitude of the sound, we should be able to either determine the direction, or distance. Once all of the microphones are added in, we will be able to determine direction and distance by comparing all of the values read. This test will make sure that we are able to either get direction or distance on one microphone, we will have to set one value in the equation to a constant in order to determine if the code is working. For the first part of the test we will set the distance and check the direction that we receive, and for the second part we will set the direction and work with the distance that we receive. When the code is written properly we will be able to do either of these tasks with the individual microphones and both tasks with all of the microphones.

After we have tested the microphones by themselves, we will shift to testing them as a whole. This will require the code running four times in parallel. Because of the nature of the evaluations that we are running, this will be taken care of in the design process and should be fairly straightforward. Just like the first part of the testing process we will first test the ability of the microphones to determine the presence of a siren, then we will move on to testing their ability to determine the distance and direction of said siren. The first set of tests will require running the software on the processor using a computer as the primary interface with the processor. Just like in the first iteration we will be testing the code with manual inputs before we switch over to the microphone inputs. If we can successfully identify a siren with each loop of code getting slightly different values due to the Doppler Effect and the different directions of the microphones, then we will know that our code is written properly. We will have to test this for each of the four sirens and random noise to make sure that we do not get a false positive result for any of the test samples. After we have tested this with manual inputs for the four microphones, then we will have to test the system with microphone inputs. This should not do anything different, just like the first test, simply making our processor run at full speed. This is the final determination that our system can identify a siren via the four microphones that are on board.

The last part of the test will determine if we can accurately determine direction and distance through combination of the four microphone inputs. As seen in the above test, we are able to determine either the direction or distance based on one individual microphone, and we just need to use the power of all four and their respective directions to determine both the direction and the distance. This will require testing just like in all of the above tests, first starting out with a manual input test for each of the four values. If we can successfully determine distance and direction through this method, then we will move on to the full speed test with microphone inputs instead of manual inputs. We will still need to test each of the four sirens as well as random road noise. After we are able to correctly identify the location of each of the four sirens through the microphone inputs, we know that we have a working code, and can move on to full system testing, which will consist of adding our DAC and audio interface to the end of this configuration.

6.6 Switching Unit

In order to test our two methods of switching we will need to run a series of scenarios that will be extremely similar to the environment that the switching unit will be run in. To set up for the test we will need the switching unit assembled and/or coded. For the processor method we will need to flash the processor and attach it to a breadboard if there is no PCB on hand. Four inputs will be the EVDS generated audio, and four will be the car generated audio. These inputs can be represented by any two powered audio sources. There will be an additional input that will run in parallel with the EVDS output audio. This will also run into the processor as the control line. Coming out of the processor there will be four output lines that will run to the speakers. To test this system we will have to run two scenarios, the first will be with the car audio source turned on to something making a noise. Then we will need to turn on the EVDS audio and make sure that the switch switches. If we hear the EVDS audio then we know that at least the first part of the switching unit works. The second part of the test will be switching back from the EVDS audio. By hitting pause on the audio source simulating the EVDS we will be simulating a complete message from the EVDS. If the switching unit again switches back to the car audio then we know that this method of switching works. The other test that is being conducted simultaneously to this one is a sound quality test. We will be listening to the audio that is being played at all times and judging it on its quality. After completing the second version of the test, testing the analog switching unit, we will compare our results and determine which method preserves sound quality better.

The second test will require a more intensive setup. There will still be eight inputs, four from each of the audio sources with an additional control line from the EVDS audio, but we will also have to construct the switch out of physical parts rather than just coding it to the processor. Once designed, we will be able to quickly build the design that we are aiming for on a breadboard and simulate our car environment again. The testing process will be extremely similar to the one outlined above. We will continue to listen to sound quality in order to determine which method will provide less distortion in our audio. After the least intrusive method has been established, our choice of method has been made for us and we will accommodate that into our building process. Both methods should theoretically work and if neither provides a significant difference then we will simply proceed with the least expensive method.

6.7 Audio Interface

In order to make sure that we have a working test bench to test our system on we will need to adequately test each part of the audio interface to make sure that it accurately represents the environment that we are trying to design our system for. We will need to run two sets of tests. The first will be on a car and its audio system, and the second will be the same set of tests on our own system so that we can make sure that our system closely models an actual system.

6.7.1 Receiver

The first part of the system that we will need to test is our receiver. There are two sets of test that we will need to run on our receiver. The first is a test to check if the Fujitsu receiver actually works, or if we will need to switch to the Yamaha option. The second test is going to be the one mentioned above that determines the level of similarity between our test system and that of an actual car. Both of these tests will help us determine if our test platform will be sufficient in testing our product throughout the design process.

The first test will test multiple aspects of the receiver, testing everything from the input power to the output signals. First we will have to adapt the system to our own plugs and wires directly from the adapter ports in the back of the receiver. The pins shown in Figure 6.2 of the receiver design section will need to be wired directly to our 12 V DC power in, the negative pin to the –12 V in, the ground pin to ground and the audio out pins wired directly to the speakers that we are trying to test. We will accomplish this custom plug configuration by using male to female pin selector cables. Once we have active power to our system we can begin to test if it works. The first part of this test is simple enough, we just hit the power button. If we have correctly attached the wires to our system then the receiver should power on, with the lights on the front of the system being an accurate indicator of our success.

There are two main inputs that we are interested in testing on the receiver that we have chosen, the CD input and the radio input. We do not have an antenna for the system so we are not incredibly optimistic about the radio working right now, but if it does, that will save us having to acquire another antenna to make it work. Our criteria for making the radio work are relatively simple, can we receive signal from at least one station that we can use for our testing car audio. If we can get at least one station in then we will be satisfied. The other input that we are testing is the CD player. This will be our second input, which we need to show that our system will work regardless of the input that the car receiver is set to. We have several CD’s that we can insert into the CD drive, if they will play then we are confident that our Receiver can perform all the input functions that we need it to.

The last part of our receiver test will be to see if the audio channels to the speakers still work. This can be done in two ways, the first is the simpler and preferred method. It will also confirm that our speakers work, and eliminate a later part of our testing methodology. If we can hit play on the receiver when it is connected to the speakers and the desired audio is played from the speakers then we know that both the speakers and the receiver perform to our expectations. If we do not achieve sound from the speakers, then we will need to test the second method. This will eliminate the possibility of broken speakers from our equation. We will simply hook the speaker leads up to a multi-meter and check for a change in voltage between when we have the system playing and when it is dormant. This will indicate a signal being sent to the speakers. We will not be able to confirm the accuracy of the signal until we have working speakers, but we will at least have a signal to work with. Once we obtain working speakers then we will be able to check the accuracy of the signal.

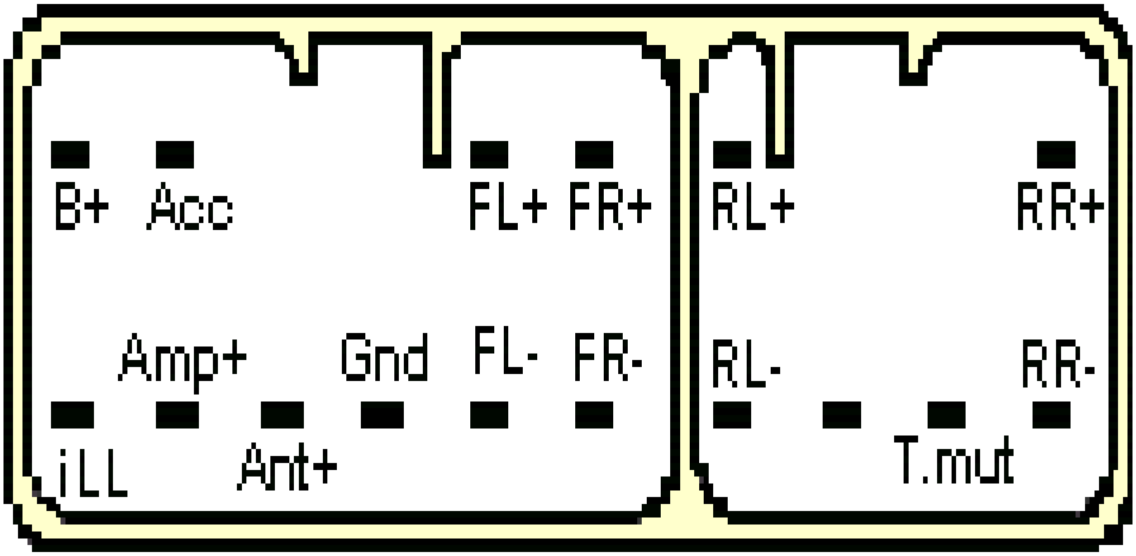


Figure 6.2: *The required pin out diagram for the back of the Fujitsu receiver. The pins that we will need are B+ (+battery), Gnd (-battery), FL+ (+front left speaker), FL- (-front left speaker), FR+ (+front right speaker), FR- (- Front Right Speaker). The rest of the pins deal with components that we will not be interfering with in the standard operation of the receiver, thus we do not need to even use them during our test process as they will remain unaffected.*

If any of these parts fails, then we will be forced to switch to the Yamaha receiver. Because this receiver is already operating in a home audio setup, we know that all of these tests will pass and that it will satisfy our requirements. Our main hesitation at selecting it for use is its size. We would much prefer using the smaller receiver that works than the much larger one, but if turns out that the smaller one does not work, then the larger one will work just as well. As stated above, this system is already tested and verified so we know that it will serve as a perfect backup system at no extra cost to our budget.

6.7.2 External Amplifier

The next portion of our audio playback system that we will need to test is our external amplifier. Like most parts of the audio system there are two methods by which we need to vet the amplifier. The first will check to make sure that eth receiver will work in the orientation that it was built for. The second test will verify that the receiver works in the orientation that we need it for. The reason for the two tests is that if it does not work in the intended orientation, then it will not work in the orientation that we have planned for it. If the amplifier we have chosen fails the first test, then we will need to return it and obtain a new amplifier until we are able to get one that works. Once we have an amplifier that passes both of the tests outlined below to our satisfaction then we know that it will work for our testing purposes.

The first test will require some audio device such as a cell phone and an aux cord. We will simply be using the amplifier to power the speakers that we are going to be using. If it plays the sound that we have input, then we know that it will work. To set up the test we will plug the phone into the amplifier via the aux cord. We will hook the speakers up to the amplifier through the channels on the back of the amplifier. For this test the amplifier will be using the included power supply plugged into a standard wall outlet. Once we have the system wired properly we will hit play on some audio file on the phone, making sure that the volume on the phone is turned up all the way and the volume on the amplifier is turned down all the way. After the audio is playing, we will continually raise the volume on the amplifier until we can her the audio that is being played. As long as we achieve an audible level of playback, then the amplifier will have passed the first test. If we do not get any playback, then there are two possibilities for this failure. The first is that the amplifier was dead on arrival, meaning that we need to send it back and get a new one. The second is that the speakers we are using are no good. We can check this result by applying a multi-meter to the speaker leads and looking for a change in voltage as we adjust the output power. If there is no change in voltage, then we know that the amplifier is dead, otherwise the speakers are our problem. This test checks the functionality of the amplifier and that it works in the way that it was intended.

The second test will make sure that the amplifier works how we need it to. There is not much difference between the two tests. The difference will come from the audio and power sources. The audio source will still be attached to the amplifier through a 3.5 mm aux cord, but instead of a cell phone, it will be plugged into a 3.5 mm port on our PCB. The audio recordings that we will be trying to play are the voice recordings that will be alerting the driver to the location of the emergency vehicle. We need to make sure that the output audio signal will work with our amplifier. The other change, the power supply, will require attaching the amplifier to our battery rather than to a wall outlet. The amplifier used a 12 V plug in adapter, so we will have to cut the wire for the power adapter and use those leads to go to our battery. Both of these tests will have to be conducted independently first, then together after we have determined that they work on their own. The test of the PCB audio with the amplifier plugged into the wall needs to be conducted first because we will have to cut the power supply for the second part of the test. To test them we will play a recording from our PCB and have the amplifier turned down all the way, when testing the receiver from the battery by itself we will still be using a cell phone to provide our audio, which we already know works. Then we will turn the amplifier volume up gradually until we can hear a playback. If each of these tests works on their own then we will combine them and test them together, this will be essentially how this part will be hooked up in our final orientation, the only difference being the switching unit that the speaker wires will run to rather than directly to the speakers. This will not affect the output of our system.

Once we get an amplifier to pass all of the above tests, we will be able to confirm that it will serve all the purposes that we will need it to. We can then move on to the next part of our testing process.

6.7.3 Speaker

The test of the speakers has been discussed in passing over the course of the last couple of tests. The presence of working speakers is rather critical to the testing of each of the other parts. There are ways of testing the other parts without having working speakers, but they are much less reliable. Our speaker test can be conducted multiple ways. Any of our receiver or amplifiers can be used to test them as long as they have already been proven to be working. The last part is the most troubling of our test process. Only working parts can test the speakers, parts which are determined to be working through the use of the speakers. The only way to correctly identify the broken part in our system is to introduce a part that we already knows works. This role is filled by the Yamaha receiver that we already have. Using this as our audio source, we can now test any speakers with confidence, knowing that if they do not produce any sound, they are broken.

To test this we will plug the speakers into the receiver audio out channels and some audio source into the input channels of the receiver. Turning the receiver down to a very low volume we will hit play on the audio input device that we have chosen, most likely a cell phone, like in the above tests. We will then turn the receiver up until we can hear sound from the speakers, and as long as it is the sound we are expecting to hear, the speakers have passed our test.

After the speakers have passed our test then they can be used to test each of the other parts in our audio playback system to make sure that those parts work as well. I do not believe that we will have much problem with broken speakers as they are rather robust devices that take a lot of abuse in order to break them. Additionally, the speakers that we are using are home audio speakers that are enclosed in a housing, further protecting them from any abuse that might happen. With all of these factors being considered and the speakers tested, we will be able to determine that the speakers we have selected are going to work for our system.

6.5.4 Overall Audio

After each individual part of the system has been determined to work, the next step for us is to configure them in the orientation that we want and test the entire system to make sure that it performs as we want it to. Based on the testing that we have already done, everything should work as planned. This will verify that, starting with the audio out from the DAC and ending with the speakers making noise. For this test there will be no simulated parts, as we want the most accurate representation of our system possible. As discussed earlier we want to be using the Fujitsu receiver primarily, only switching if it ends up breaking, but that will be determined after the first set of tests is complete. We will also start with the Bose speakers and the Lepy 2020A external amplifier. All of these components will be connected via a breadboard in order to test our concept first. After we have achieved successful test in all of our parts, we will finalize our PCB design and get that on order, then we will repeat the test with the PCB in order to verify that the PCB works as expected.

The method of this testing is very similar to how each of the individual components were tested. First we will hook everything up power to the system. By starting with one component, the switching unit wired to a breadboard and building outward we can consecutively test each component so that if the system fails we know which component caused it. Because we will need speakers to test the switching unit, the Bose speakers will already be hooked up. After repeating our test in the above section and verifying a working switch we will add in a different source for the car audio, our receiver. Again we will repeat the test on the switching unit. If it still performs as expected then we will change the source of the EVDS Simulated audio to some audio source that runs through the external amplifier, but still not the DAC signal from our board. Once again repeating the above tests we will confirm that the switch works before moving on to the last test. Using all of our actual components we will run the test one more time. This is that last iteration of the test and will determine the overall functionality of the system. If the switch operates as intended, then we know that our audio interface system works and is ready to be connected to the other components of the system.

7.0 Administrative Content

To begin the project, the group came together for several brainstorming sessions in order to determine all of the steps necessary to complete the emergency vehicle detection system. The initial milestones and schedule below were the results of the team's first couple meetings and basic research. As the semester went on, more research was done and the milestones and schedule were changed accordingly. Several of the milestones were either eliminated or pushed pack in the schedule due to time or money constraints.

7.1 Initial Milestones and Schedule

1. All Research Done (Code and Parts) Digital signal processing Code ADC Code Switcher Code LED Code Frequency Samples (digital file from FD) 2. Build Audio System Amp Speakers Receiver Switching Unit 3. Microphones Purchased and Tested 4. MCU Purchased 5. Printed Casing Design Print (In for Printing) 6. Schematic On Paper 7. Audio Recordings Made Deciding what recordings are going to be there 8. Printed PCB Ordered 9. Finish Paper Complete work Format Print and bind	18 September • Microphone Researched (Ryan) • FD Siren Discussion (John) • Board/MCU Sample (DJ) • In House Parts List (All) • Start Audio System (John) • Soar Application (All) • List of Possible Sponsors (All) 25 September • Finalize Microphone (Numbers) (Ryan) • Report Back on Coding Research (All) • Have MCU Narrowed Down (DJ) • Sponsor List (All) 2 October • Code Research (All) • Microphone Research (Ryan) 30 October (Halfway to Semester End) • MCU Purchased (DJ) • Microphones Ordered (Ryan) • Start Coding (All) • digital signal processing (Ryan) • ADC (DJ) • LED and Switcher (John) • Filter Design Started (DJ) • PCB Design Started (All) • Start Casing Design (All) • PRINT ASAP 13 November • CAD Drawing of Casing Done (All) • Filter Design Done (DJ) • PCB Design almost done (All) 4 December • Audio Recordings Made (All) • PCB Ordered (All) • Have paper ready for formatting

7.2 Final Milestones and Schedule

1. All Research Done (Code and Parts) Digital signal processing Code ADC Code Switcher Code LED Code Frequency Samples (digital file from FD) 2. Build Audio System Amp Speakers Receiver Switching Unit 3. Microphones Purchased and Tested 4. MCU Purchased 5. Schematic On Paper 6. Audio Recordings Made Deciding what recordings are going to be there	18 September • Microphone Researched (Ryan) • FD Siren Discussion (John) • Board/MCU Sample (DJ) • In House Parts List (All) • Start Audio System (John) • Soar Application (All) • List of Possible Sponsors (All) 25 September • Finalize Microphone (Numbers) (Ryan) • Report Back on Coding Research (All) • Have MCU Narrowed Down (DJ) • Sponsor List (All) 10 October • Code Research (All) • Microphone Research (Ryan) 4 September • MCU Purchased (DJ) • Microphones Ordered (Ryan) • Start Coding (All) • digital signal processing (Ryan) • ADC (DJ) • LED and Switcher (John) • Filter Design Started (DJ) • PCB Design Started (All) 13 November • Filter Design Done (DJ) • PCB Design almost done (All) 4 December • Have paper ready for formatting

After reviewing the schedule and milestones, it can be seen that the group had very high expectations at the beginning of the semester. However, time did not permit the group to complete all that was planned. Overall, the group was able to stay on schedule and complete almost all of the tasks laid out.

7.3 Division of Labor

Since there are three members in our design group, along with three major components of the project, the division of labor was centered around these lines. The analog to digital conversion portion of the emergency vehicle detection system was distributed to Daniel Christiano, along with the filter design and digital to analog conversion section. Also, Daniel was in charge of compatibility of voltage levels, and connection types between processors. The digital-signal processing unit and microphones were sectioned off to Ryan Chappell. John Fick was assigned the decision logic once the analog to digital signal was converted to frequencies and amplitudes from Ryan. Also, John was assigned the output audio system. This consisted of the amplification, switching logic, and output speaker compatibility.

7.4 Initial Budget

Below are description and break downs of the initial, middle and final budget plans for the emergency vehicle detection system. The initial budget was created with minimal research into parts, and was more about making a parts list than defining the budget. The middle budget is a more refined version of the initial budget, but is still incomplete. This middle budget more accurately depicts the cost of proposed parts.

Description	Amount
Microphones (2-4)	$137.50
MSP 430	$20.00
Pcb board (Custom)	$100.00
3d printed casing	$100.00
Wiring for speaker	$20.00
Battery	$50.00
Siren	$50.00
Car audio controller	$100.00
Speaker	$50.00
Amp for audio controller	$100.00
Bolts/screws for mounting	$10.00
“Car"	$10.00

Total	$747.50

7.5 Middle Budget

Description	Amount
Microphones (2-4)	$548.00
ARM Processor	$20.00
PCB board (Custom)	$100.00
3d Printed Casing	$100.00
Wiring for Speaker	$20.00
Battery	$50.00
Siren	$50.00
Car Audio Controller	$100.00
Speaker	$50.00
Amp for Audio Controller	$100.00
Bolts/Screws for mounting	$10.00
“Car"	$10.00
	$1,148.00
Total	$1,158.00

7.5 Final Budget

Description	Amount
Microphones (2-4)	$548.00
DSP + ARM Processor	$50.00
PCB board (Custom)	$100.00
Wiring for Speaker	$20.00
Battery	$50.00
Siren	$50.00
Car Audio Controller	$100.00
Speaker	$50.00
Amp for Audio Controller	$100.00
Bolts/Screws for mounting	$10.00
“Car"	$10.00
ADC chips (4)	$20.00
Total	$1108.00

This final budget is the most accurate version based on the research and design done this semester.

8.0 Conclusion

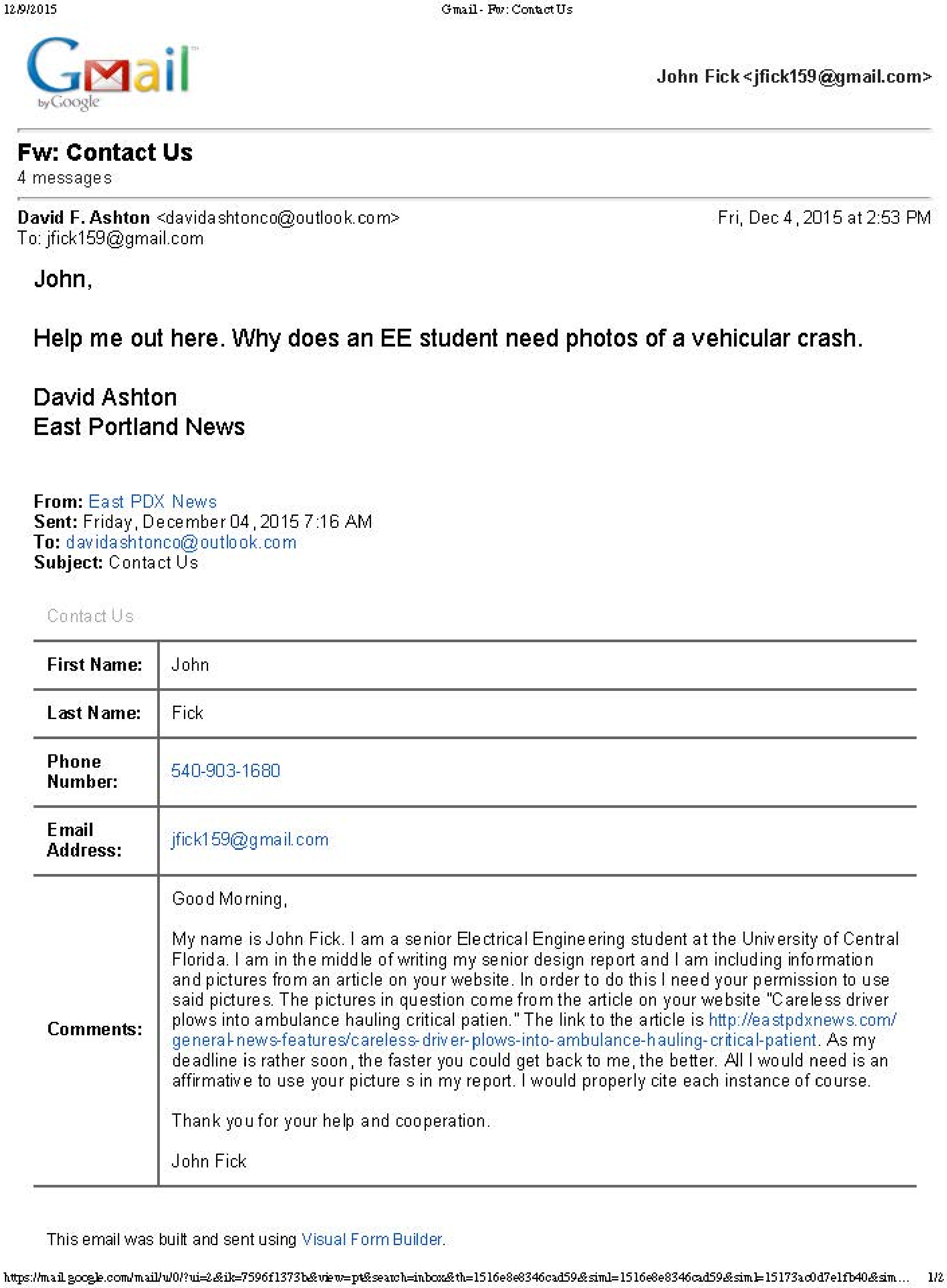
At the start of our project we set out to improve automobile safety in one very specific area, Emergency Vehicle collisions. Through personal experiences and research, we determined that the main cause for accidents of this nature are drivers caught unware. Due to the increasingly soundproof nature of modern automobile cabins, and their ability to almost completely block out the outside world, drivers are no longer able to hear an approaching emergency vehicle. In order to combat this problem we decided that another set of ears for the driver was in order. Through our implementation of microphones, we have allowed select sounds outside the car to make their way inside the car. Though the driver will not be treated to a jarring mirror of a standard siren, a soothing message recording indicating an approaching EV will be played when one is heard. By doing this we can make sure that every driver with our EVDS installed in their car will be alert to the approaching vehicle, hopefully without causing so much shock or alarm as to cause an additional accident.

We have spent the semester designing our system and have reached the point where we are ready to begin acquiring and assembling the parts of our system. Each piece of code that we will be utilizing has been laid out and is ready be written. We have on hand many of the parts required for testing the system and will be starting testing our system ASAP. We have outlined above the various methods by which we will be testing our system, both as components and as a whole. Through the utilization of the plans outlined above we will be able to construct our system with maximum efficiency, achieving all the goals that we have set before us.

Through our research and design, we have delved deeply into several digital fields as well as the analog ones that will interact with the driver. We will be using segments of ADC, DSP, and Decision Logic to process the information that is running through our system. These are three major fields of Electrical and Computer Engineering that we have explored and will be covering in much greater detail next semester. Overall this project has broadened our horizons in many aspects of Electrical and Computer Engineering, and will on continue to do so as we progress forward next semester.

Appendices

i. Permissions



12/9/2015 Gmail - Fw: Contact Us

John Fick <jfick159@gmail.com>

Fw: Contact Us
4 messages

David F. Ashton <davidashtonco@outlook.com> Fri, Dec 4, 2015 at 2:53 PM
To: jfick159@gmail.com

John,

Help me out here. Why does an EE student need photos of a vehicular crash.

David Ashton
East Portland News

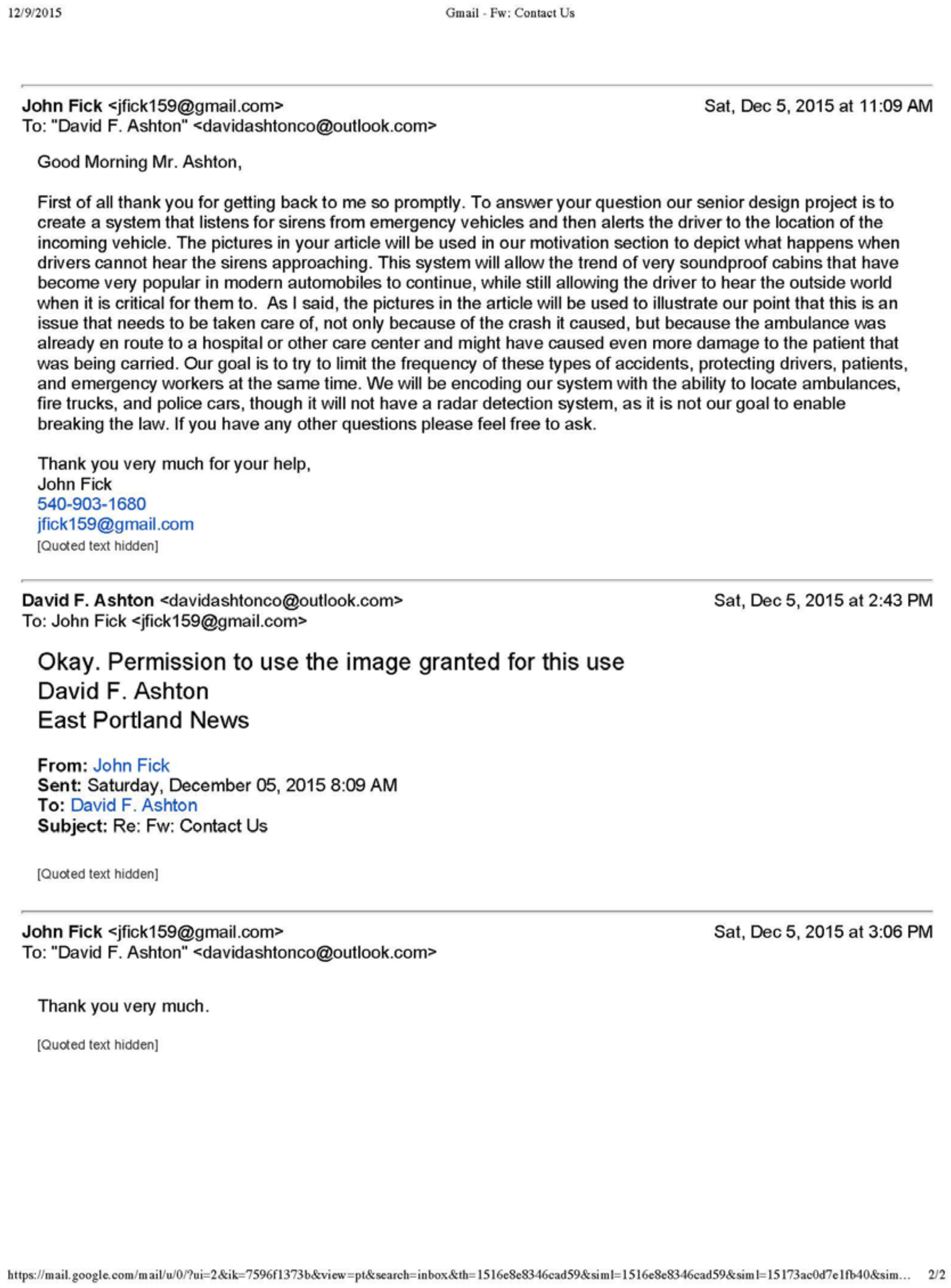
From: East PDX News
Sent: Friday, December 04, 2015 7:16 AM
To: davidashtonco@outlook.com
Subject: Contact Us

Contact Us

First Name:	John
Last Name:	Fick
Phone Number:	540-903-1680
Email Address:	jfick159@gmail.com
Comments:	Good Morning, My name is John Fick. I am a senior Electrical Engineering student at the University of Central Florida. I am in the middle of writing my senior design report and I am including information and pictures from an article on your website. In order to do this I need your permission to use said pictures. The pictures in question come from the article on your website "Careless driver plows into ambulance hauling critical patien." The link to the article is http://eastpdxnews.com/general-news-features/careless-driver-plows-into-ambulance-hauling-critical-patient. As my deadline is rather soon, the faster you could get back to me, the better. All I would need is an affirmative to use your picture s in my report. I would properly cite each instance of course. Thank you for your help and cooperation. John Fick

This email was built and sent using Visual Form Builder.

https://mail.google.com/mail/u/0/?ui=2&ik=7596f1373b&view=pt&search=inbox&th=1516e8e8346cad59&siml=1516e8e8346cad59&siml=15173ac0d7e1fb40&sim… 1/2



John Fick <jfick159@gmail.com> Sat, Dec 5, 2015 at 11:09 AM
To: "David F. Ashton" <davidashtonco@outlook.com>

Good Morning Mr. Ashton,

First of all thank you for getting back to me so promptly. To answer your question our senior design project is to create a system that listens for sirens from emergency vehicles and then alerts the driver to the location of the incoming vehicle. The pictures in your article will be used in our motivation section to depict what happens when drivers cannot hear the sirens approaching. This system will allow the trend of very soundproof cabins that have become very popular in modern automobiles to continue, while still allowing the driver to hear the outside world when it is critical for them to. As I said, the pictures in the article will be used to illustrate our point that this is an issue that needs to be taken care of, not only because of the crash it caused, but because the ambulance was already en route to a hospital or other care center and might have caused even more damage to the patient that was being carried. Our goal is to try to limit the frequency of these types of accidents, protecting drivers, patients, and emergency workers at the same time. We will be encoding our system with the ability to locate ambulances, fire trucks, and police cars, though it will not have a radar detection system, as it is not our goal to enable breaking the law. If you have any other questions please feel free to ask.

Thank you very much for your help,
John Fick
540-903-1680
jfick159@gmail.com
[Quoted text hidden]

David F. Ashton <davidashtonco@outlook.com> Sat, Dec 5, 2015 at 2:43 PM
To: John Fick <jfick159@gmail.com>

Okay. Permission to use the image granted for this use
David F. Ashton
East Portland News

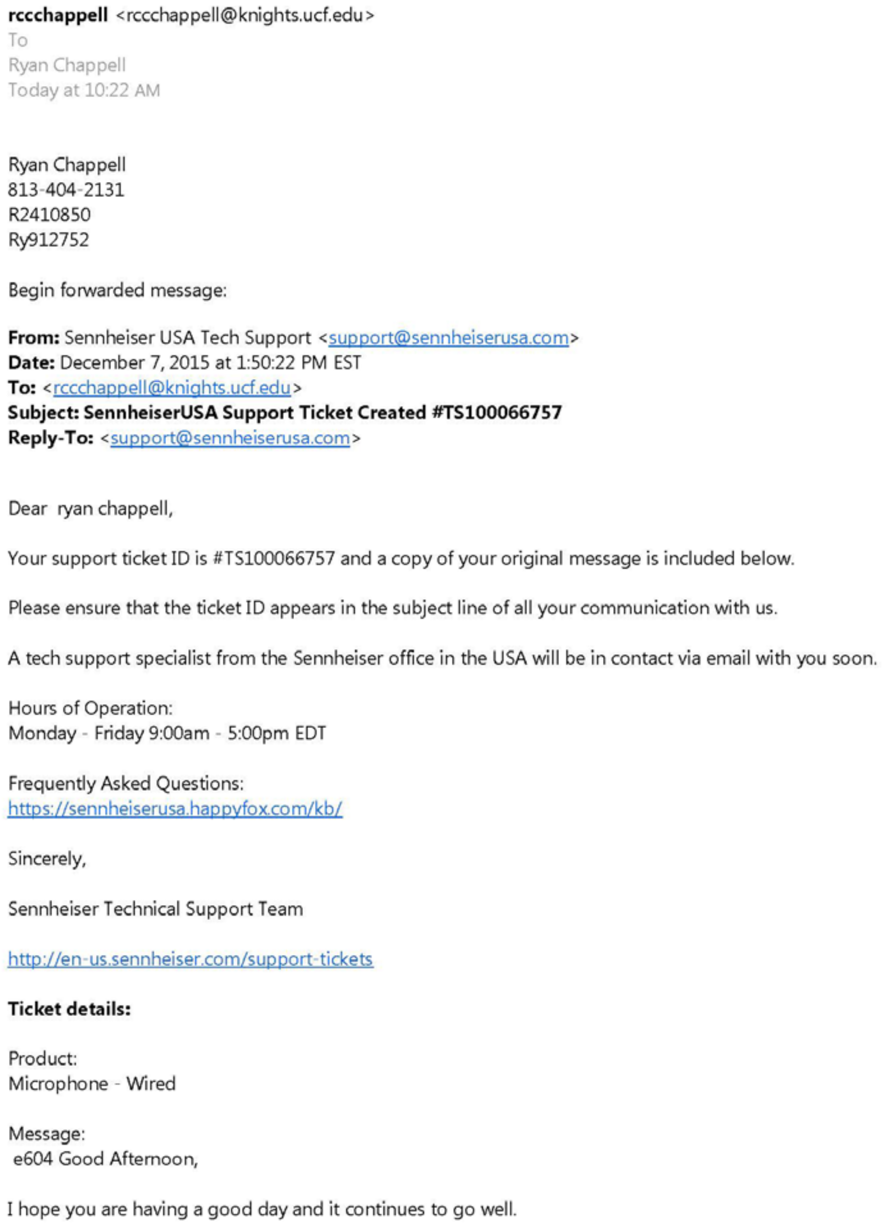
From: John Fick
Sent: Saturday, December 05, 2015 8:09 AM
To: David F. Ashton
Subject: Re: Fw: Contact Us

[Quoted text hidden]

John Fick <jfick159@gmail.com> Sat, Dec 5, 2015 at 3:06 PM
To: "David F. Ashton" <davidashtonco@outlook.com>

Thank you very much.

[Quoted text hidden]



rccchappell <rccchappell@knights.ucf.edu>
To
Ryan Chappell
Today at 10:22 AM

Ryan Chappell
813-404-2131
R2410850
Ry912752

Begin forwarded message:

From: Sennheiser USA Tech Support <support@sennheiserusa.com>
Date: December 7, 2015 at 1:50:22 PM EST
To: <rccchappell@knights.ucf.edu>
Subject: SennheiserUSA Support Ticket Created #TS100066757
Reply-To: <support@sennheiserusa.com>

Dear ryan chappell,

Your support ticket ID is #TS100066757 and a copy of your original message is included below.

Please ensure that the ticket ID appears in the subject line of all your communication with us.

A tech support specialist from the Sennheiser office in the USA will be in contact via email with you soon.

Hours of Operation:
Monday - Friday 9:00am - 5:00pm EDT

Frequently Asked Questions:
https://sennheiserusa.happyfox.com/kb/

Sincerely,

Sennheiser Technical Support Team

http://en-us.sennheiser.com/support-tickets

Ticket details:

Product:
Microphone - Wired

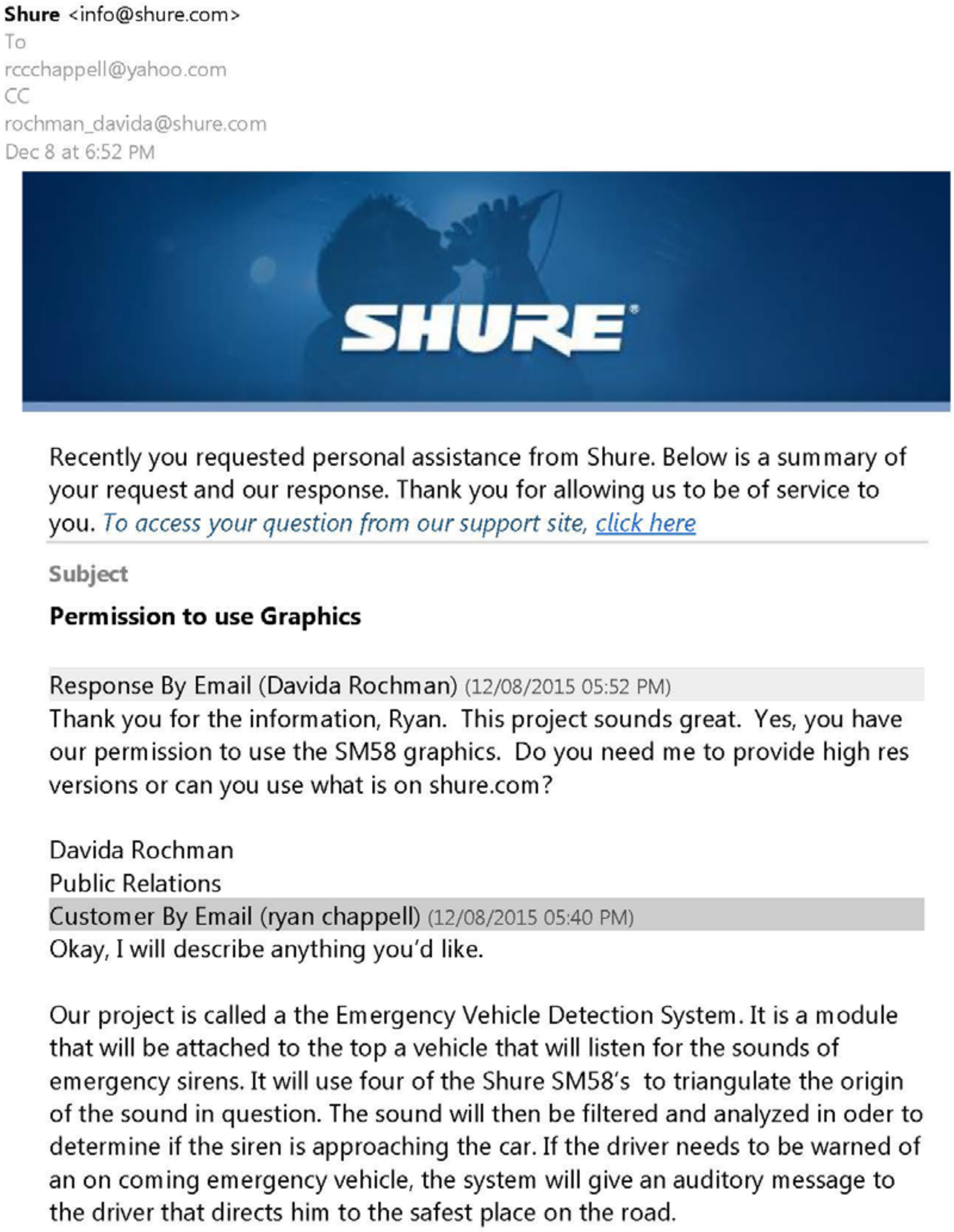
Message:
e604 Good Afternoon,

I hope you are having a good day and it continues to go well.

I am apart of a electrical engineering senior design group at the University of Central Florida. My group is going to be utilizing the e604 in our design. We need written permission to allow my group to use some images from the data sheet in the final paper . These images include the polar pattern and frequency response of the microphone.

Thank you for your help and quick response,

Ryan Chappell
8134042131



Shure <info@shure.com>
To
rccchappell@yahoo.com
CC
rochman_davida@shure.com
Dec 8 at 6:52 PM

Recently you requested personal assistance from Shure. Below is a summary of your request and our response. Thank you for allowing us to be of service to you. *To access your question from our support site, click here*

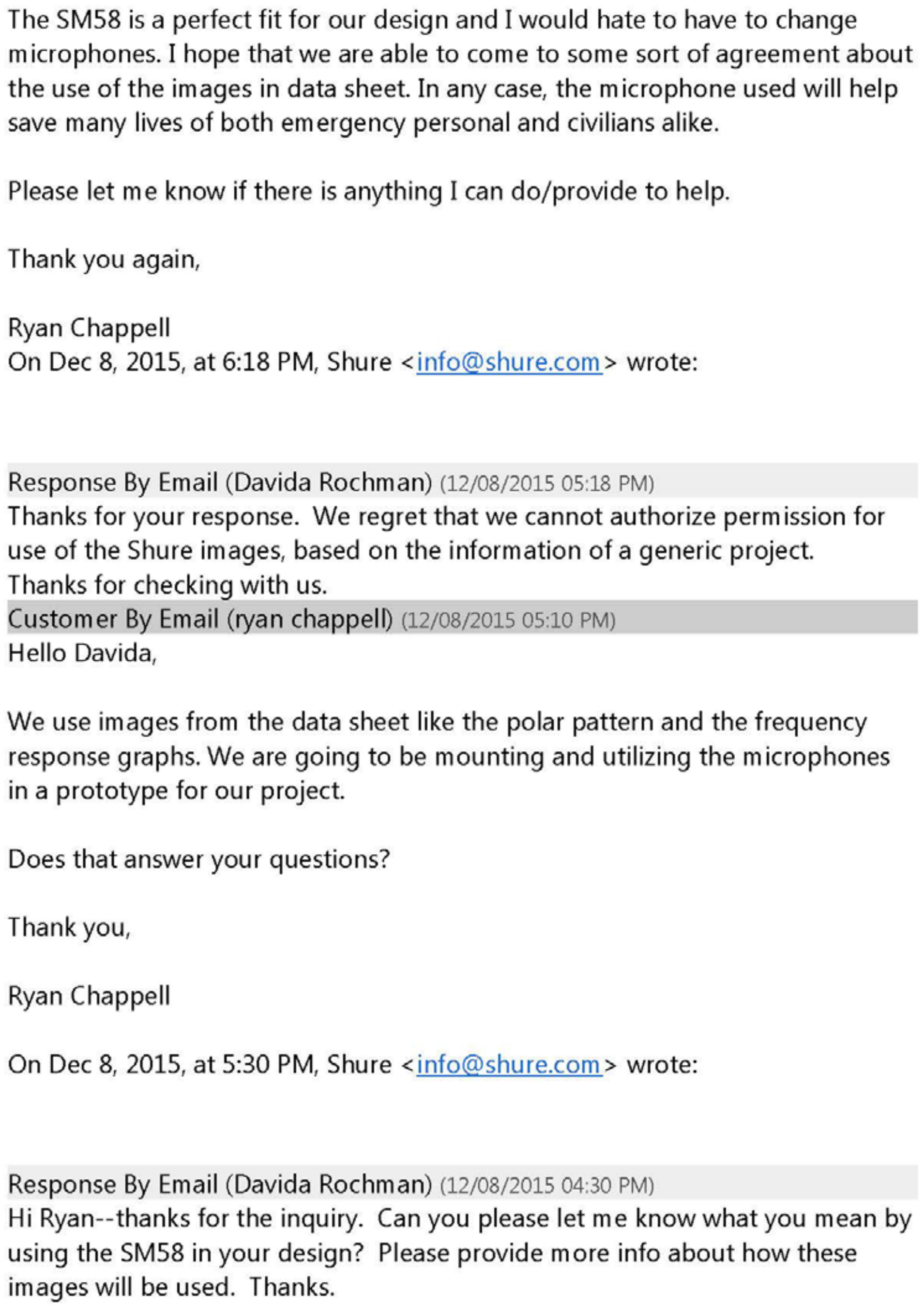
Subject

Permission to use Graphics

Response By Email (Davida Rochman) (12/08/2015 05:52 PM)
Thank you for the information, Ryan. This project sounds great. Yes, you have our permission to use the SM58 graphics. Do you need me to provide high res versions or can you use what is on shure.com?

Davida Rochman
Public Relations
Customer By Email (ryan chappell) (12/08/2015 05:40 PM)
Okay, I will describe anything you'd like.

Our project is called a the Emergency Vehicle Detection System. It is a module that will be attached to the top a vehicle that will listen for the sounds of emergency sirens. It will use four of the Shure SM58's to triangulate the origin of the sound in question. The sound will then be filtered and analyzed in oder to determine if the siren is approaching the car. If the driver needs to be warned of an on coming emergency vehicle, the system will give an auditory message to the driver that directs him to the safest place on the road.



The SM58 is a perfect fit for our design and I would hate to have to change microphones. I hope that we are able to come to some sort of agreement about the use of the images in data sheet. In any case, the microphone used will help save many lives of both emergency personal and civilians alike.

Please let me know if there is anything I can do/provide to help.

Thank you again,

Ryan Chappell
On Dec 8, 2015, at 6:18 PM, Shure <info@shure.com> wrote:

Response By Email (Davida Rochman) (12/08/2015 05:18 PM)
Thanks for your response. We regret that we cannot authorize permission for use of the Shure images, based on the information of a generic project. Thanks for checking with us.
Customer By Email (ryan chappell) (12/08/2015 05:10 PM)
Hello Davida,

We use images from the data sheet like the polar pattern and the frequency response graphs. We are going to be mounting and utilizing the microphones in a prototype for our project.

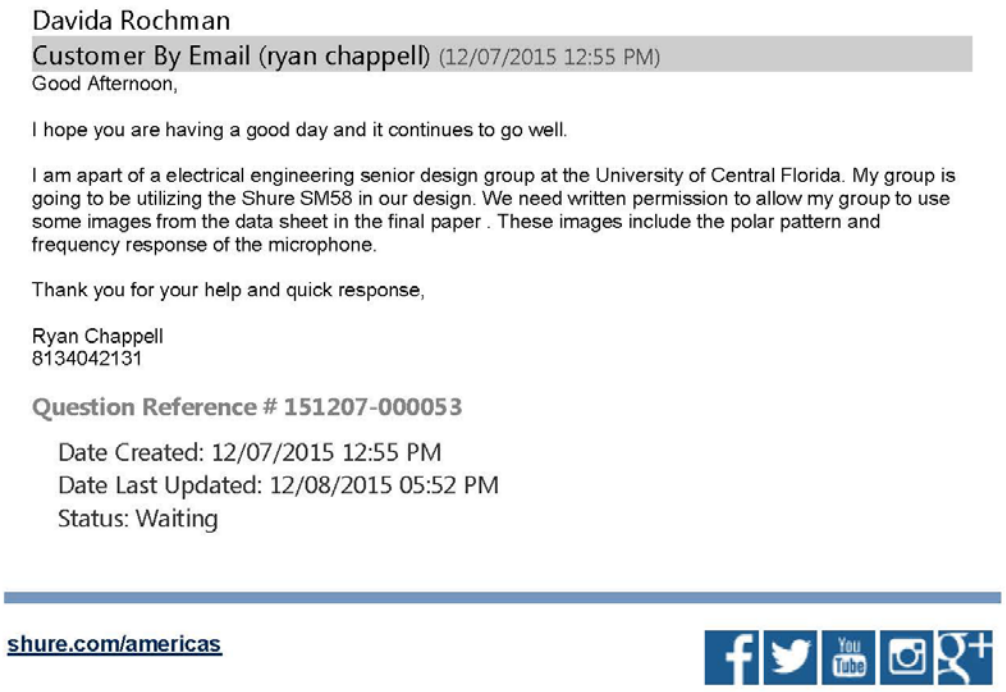
Does that answer your questions?

Thank you,

Ryan Chappell

On Dec 8, 2015, at 5:30 PM, Shure <info@shure.com> wrote:

Response By Email (Davida Rochman) (12/08/2015 04:30 PM)
Hi Ryan--thanks for the inquiry. Can you please let me know what you mean by using the SM58 in your design? Please provide more info about how these images will be used. Thanks.



Davida Rochman

Customer By Email (ryan chappell) (12/07/2015 12:55 PM)

Good Afternoon,

I hope you are having a good day and it continues to go well.

I am apart of a electrical engineering senior design group at the University of Central Florida. My group is going to be utilizing the Shure SM58 in our design. We need written permission to allow my group to use some images from the data sheet in the final paper . These images include the polar pattern and frequency response of the microphone.

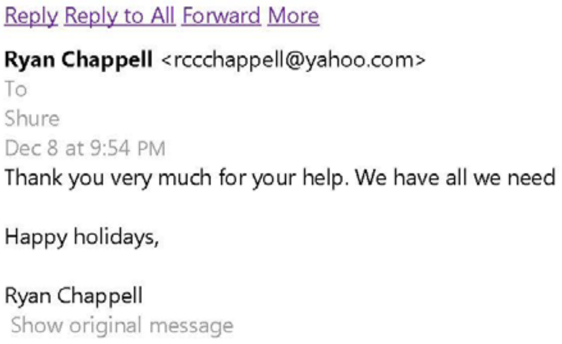
Thank you for your help and quick response,

Ryan Chappell
8134042131

Question Reference # 151207-000053

Date Created: 12/07/2015 12:55 PM
Date Last Updated: 12/08/2015 05:52 PM
Status: Waiting

shure.com/americas

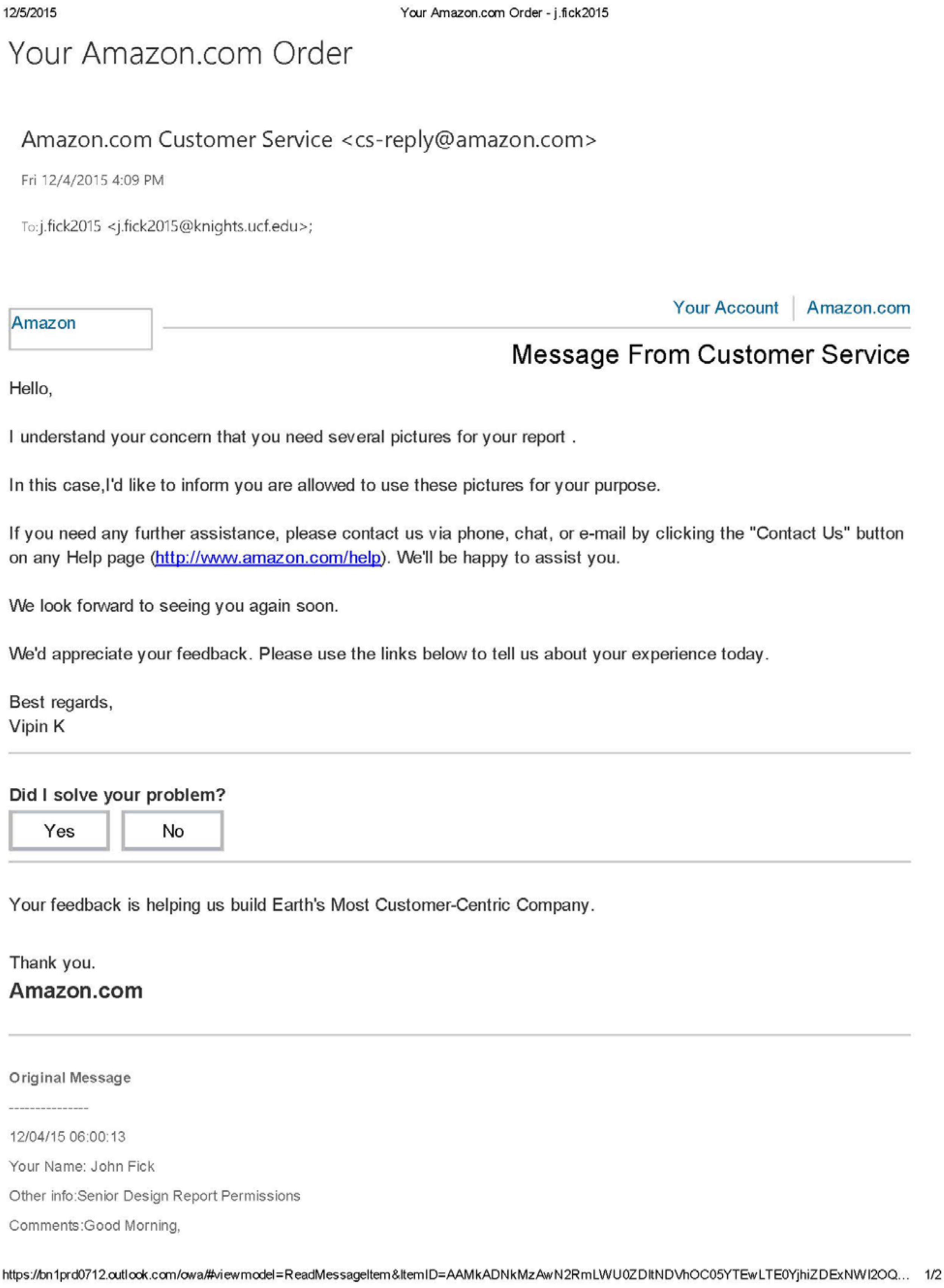


Ryan Chappell <rccchappell@yahoo.com>
To
Shure
Dec 8 at 9:54 PM

Thank you very much for your help. We have all we need

Happy holidays,

Ryan Chappell



Your Amazon.com Order

Amazon.com Customer Service <cs-reply@amazon.com>

Fri 12/4/2015 4:09 PM

To:j.fick2015 <j.fick2015@knights.ucf.edu>;

Your Account | Amazon.com

Amazon

Message From Customer Service

Hello,

I understand your concern that you need several pictures for your report .

In this case,I'd like to inform you are allowed to use these pictures for your purpose.

If you need any further assistance, please contact us via phone, chat, or e-mail by clicking the "Contact Us" button on any Help page (http://www.amazon.com/help). We'll be happy to assist you.

We look forward to seeing you again soon.

We'd appreciate your feedback. Please use the links below to tell us about your experience today.

Best regards,
Vipin K

Did I solve your problem?

Yes No

Your feedback is helping us build Earth's Most Customer-Centric Company.

Thank you.
Amazon.com

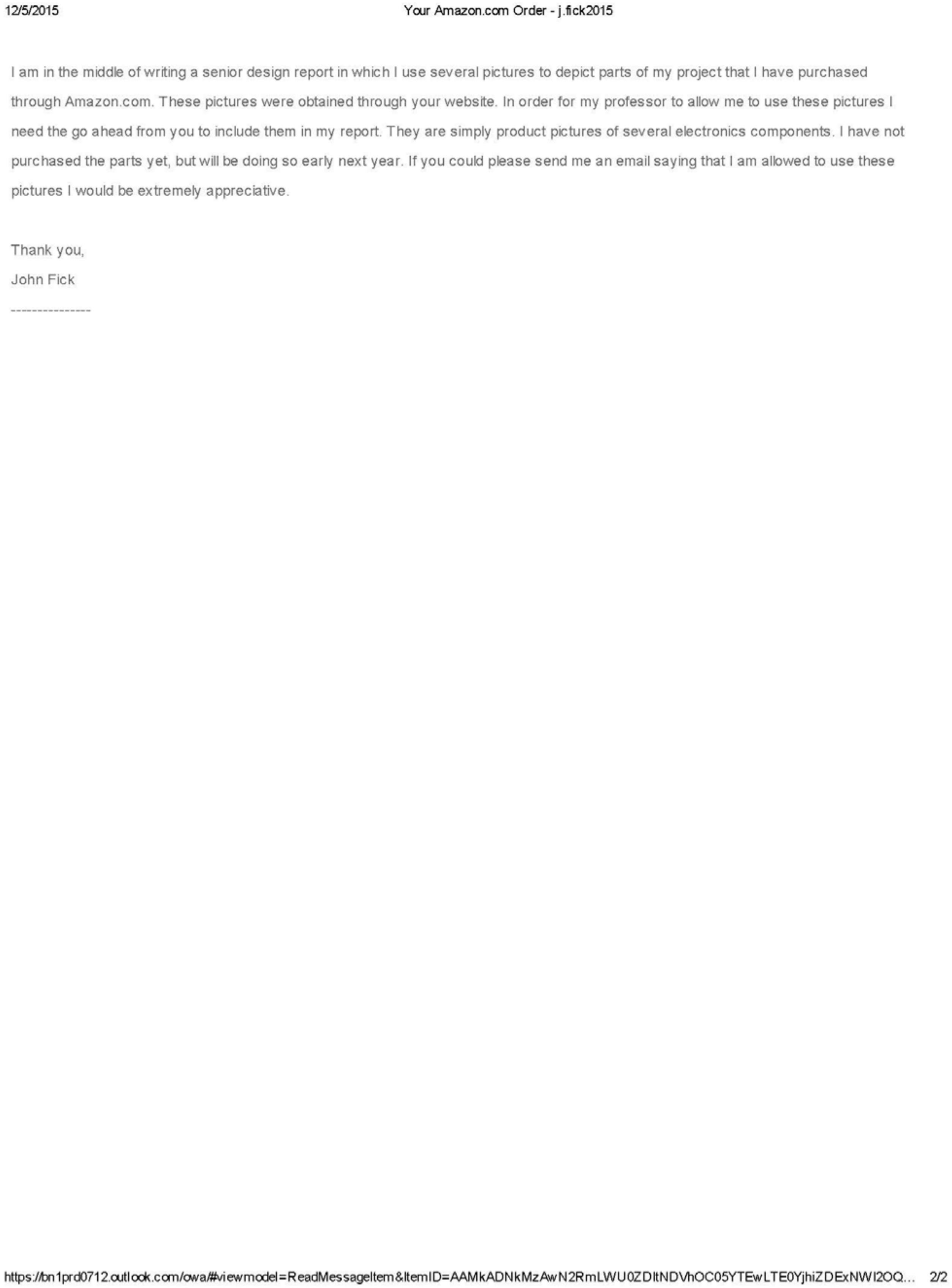
Original Message

12/04/15 06:00:13

Your Name: John Fick

Other info:Senior Design Report Permissions

Comments:Good Morning,



I am in the middle of writing a senior design report in which I use several pictures to depict parts of my project that I have purchased through Amazon.com. These pictures were obtained through your website. In order for my professor to allow me to use these pictures I need the go ahead from you to include them in my report. They are simply product pictures of several electronics components. I have not purchased the parts yet, but will be doing so early next year. If you could please send me an email saying that I am allowed to use these pictures I would be extremely appreciative.

Thank you,
John Fick
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ii. Figures

iii. Tables

iv. Data Sheets

For printing purposes the data sheets for the relevant parts have been provided electronically