AUDIO SPEAKER PROTECTION FROM UNSAFE LEVELS OF AMPLIFIER GAIN USING SMOOTH LIMITING ALGORITHMS AND FEEDBACK CONTROL

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ABSTRACT

The device we have designed and built is an audio limiting device to automatically calculate how much limiting of the small signal is needed to protect a specific speaker cabinet. The user of the device inputs their speaker impedance and speaker power rating. The audio amplifier's output then can be fed back to the limiter where the limiter can detect when the power amplifier's output wattage surpasses the power the speaker can take. The limiter can then lower the sound level of the audio signal using an algorithm to perform smooth attenuation in order to maintain the aesthetic quality of the audio signal.

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INTRODUCTION

The ever-expanding realm of technology and electronics is conquering every field imaginable in this day and the field of music and sound reproduction in general is no exception. Unfortunately, with the wide range of functions required of electronics in order to manipulate the sound in the desired fashion, there has developed a need to incorporate many different individual devices into a single signal chain from input to output. In most circumstances, the last component accounting for the audible reproduction of the manipulated sound is the speaker. As with all electronic devices, however, there are limits on the operating levels of the speaker that determine its safe operating levels. These limits cause a problem when we consider that the other components that have manipulated the audio signal up until this point are independent of the speaker and thus unaware of these safety requirements. The goal of our device is not only to incorporate a limiting device into a signal chain that would account for these safety limitations of the speaker in the presence of an amplifier, but to do so in a manner that would maintain the continuity of the volume's dynamics in an aesthetically pleasing, non-distracting manner.

A Note About Speaker Damage:

With the purpose of our device being to protect speakers, we must first understand what damages speakers in the first place. When we think of damage to a physical device, we first consider the mechanically active parts of the device and how the behavior of these parts could be put in danger. For the speaker's case, these mechanical parts are located at the point of the transduction of electrical energy in the incoming voltage signal to mechanical energy. Here, the cone of the speaker and its suspension (which is usually attached to the coil containing the incoming electrical signal; some designs allow for the permanent magnet component to be attached to the cone instead, as shown in figure 1) oscillate back and forth in its production of the desired air pressure variations in order to create the necessary sound waves. The mechanical motion of this cone is directly related to the oscillating voltage level entering the speaker. This means that the higher the peak voltage of the incoming signal, the greater the spatial displacement the speaker cone will have to undergo in order to achieve a corresponding sound wave (a higher peak in voltage signal and mechanical oscillatory motion of the cone results in a louder sound at the output of the speaker). However, as with any mechanical component, the range of motion of this cone is only designed to oscillate with amplitudes corresponding the power rating and impedance of the speaker. Any signal that surpasses these limits puts uncharacteristic strain on the cone and the suspension attaching the cone to the speaker mainframe due to the extreme accelerations, stretching, and prolonged over-vibration etc. that it experiences under such conditions. To prevent this type of damage we need to ensure a safe range of motion for the speaker cone. This however does not necessarily entail always limiting any sharp intermittent spike

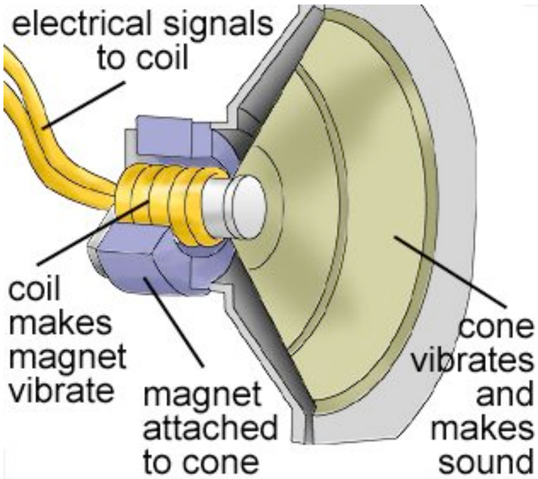

Figure 1: Speaker components
Source: http://p-hardware.blogspot.com/2008/04/speaker.html

encountered in the audio signal's amplitude. If the duration of the spike is appropriately short and the audio signal in the surrounding time window is well below dangerous operating amplitudes, it is very likely the speaker cone would be able to withstand such a spike. Most of the time one such over-excursion of the speaker cone will not cause the damage, but prolonged over-excursions leading to the mechanical fatigue will eventually result in failure of the mechanical parts, including ripping or tearing of the cone or its suspension. Another thing to note is the different types of speakers that can be used and which types are most prone to different types of speaker damage. In the case of a mechanical failure, woofer and tweeters are the usual culprits. Woofers handle the low frequency spectrum of the input and are oftentimes overdriven at loud parties, where a loud bass sound is desired. Tweeters are oftentimes overdriven mechanically when they are associated with an inadequate crossover system, since these speakers are very small, fagile, and easily damaged. In our project we will have to find a delicate balance between protecting the speaker from high-amplitude spikes that are mechanically hazardous to the system while still leaving the sound levels unaffected if it is determined that such a spike in sound will not likely be detrimental to the speaker.

The other type of damage a speaker could encounter is heat damage. If the speaker is producing a high volume of sound for an extended amount of time, the amount of power being dissipated rises to very high levels, which causes the speaker to heat up to extremely high temperatures and puts the speaker at risk for damage by overheating. At these extremely high temperatures, either the voice coil adhesive can soften and cause the voice coil to come apart, hence a mechanical failure on the part of the voice coil, or the voice coil itself or the wires leading to the voice coil can literally melt, causing oftentimes an open circuit in the wiring and thus an inability of the speakers to produce sound of any kind. The system is not put at risk of heat damage at the occurrence of sharp spikes of high amplitudes, but only at a continuously high level of sound being producedand for a reasonably long amount of time. In our device design, our goal is to protect the speaker from both mechanical and heat damage. Our protection algorithm does not involve an integration of the power dissipated by the speaker over time, however, which would be necessary for an absolute guarantee that an over-heating issue would not occur. Since we are limiting the amplitude of the signal entering the speaker, however, attenuating the signal appropriately based on both the speaker's power rating and impedance, we would expect that under normal operating conditions (where the speaker is driven with normal music for a couple hours at a time), our device would offer adequate protection from both mechanical and heat damage.

A Note About Power Ratings and Distortion:

In our attenuation calculations, we must often incorporate the power ratings of both the speaker, as input by the user, and the power rating of the amplifier we are using. We must be aware, however, that even though these power ratings appear to be on the same scale one another, they in fact are not. The power ratings of an amplifier is based on the amount of power the amplifier is capable of dissipating with a sinusoidal, monotone input into the system producing that power. This method of determining he power rating is as straightforward as the power rating on a light bulb, where the power a light bulb can dissipate is based on a steady ac power input from a wall socket. The speaker's power rating, however, is rated much differently. Since we would almost never expect to be using a speaker to output a sound that stays at a single frequency, these speakers are in stead based on typical trends found in music waveforms, where there are constantly varying amplitudes and frequencies of sound and where, more importantly, even if the signal may have an instantaneous power dissipation at a high value, this is usually followed by a period of much lower values, which essentially allows the speaker time to cool off. Because of this, an amplifier rated at a high level can oftentimes be used to drive a speaker with a much lower power rating safely. For example, if we have a speaker rated at 100 Watts, this speaker could not handle 100 Watts worth of a continuous signal (such as a monotone sinusoid) into the system. However, an amplifier rated at 400 Watts would be required to carry out the amplification of a music signal that had instantaneous power peaks of up to 400 Watts. A given music signal with an average power of 100 Watts would likely be constantly varying between say 50 and 400 Watts of power, and thus would require an amplifier of at least 400 Watts of power rating to amplify this signal without clipping. Since we know that the speaker is rated based on the varying power changes in music, then, we could conclude that a 100-Watt speaker would be able to safely run the signal, even though it is driven with an amplifier rated at 400 Watts and with a music signal containing 400-Watt peaks.

We should be aware that the previous example is dependent on the amplifier operating without distortion, that is, without the clipping of the signal peaks. When the input is clipped by the amplifier, the transistors saturate and automatically temporarily output their maximum supply DC voltage of the system to produce the square-like waveforms characteristic of clipping. When this happens, the speakers are more prone to over-heating damage than they would be if that signal would have been unclipped and reached its full amplitude peak. For this reason, it is often said that some speaker damage can actually be avoided by using a higher rated amplifier, since it would avoid this distorting clipping from occurring.

In our project, since we are not detecting if a peak value observed was from a clipped signal or an unclipped signal, in order to avoid an increased risk of heat damage due to clipping, we would need to assume that the amplifier is operating in a non-distorting range. However, since we are taking the power rating of the amplifier into account in our calculations, this should allow our device to prevent the amplifier from reaching the range where it would begin such distortion. Therefore, our signal not only functions to protect our speakers from being damaged, but also to prevent distortion in its signal output.

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KEY TO ABBREVIATIONS

ADC – analog to digital conversion: a general signal conversion method; also a register defined on the Atmega 32 microprocessor for carrying out such a conversion

CS – chip select; an input pin to the MAX 5411 digital potentiometer, active low

PCB - printed circuit board; copper platforms whereupon electronic circuits have been etched. PCBs are rugged, inexpensive, and can be highly reliable. They require much more layout effort and higher initial cost than either wire-wrapped or point-to-point constructed circuits, but are much cheaper and faster for high-volume production.

QSOP - Quarter-Size Small-Outline package; a chip package with pin spacing of 0.635 mm

SPI – serial peripheral interface; a synchronous serial data link standard of communication involving a “master” talking to its “slave” over four wires called master-in-slave-out (MISO), master-out-slave-in (MOSI), clock (CLK), and slave select (SS)

SS – slave select; acts as a chip select for SPI communication slave enabling; a dedicated pin on port B of the microprocessor, active low

CHAPTER 1
PROJECT FEATURES AND OBJECTIVES

The main distinguishing feature of our device compared to similar devices currently on the market is our capability to accommodate a wide variety of speakers, allowing the consumer to use any typical speaker sold commercially today. We accomplish this by inserting user interfaces whereby the user can specify the impedance and power specifications of the particular speaker, regardless of the speaker's make or model. The current industry standards for speaker impedance values are set at discrete values of 2, 4, or 8 Ohms and the power consumption levels of a typical speaker ranges between 50 and 1000 Watts. The user simply selects the levels of these parameters of their speakers and our device automatically processes these inputs and recalibrates the attenuation scheme of the limiting algorithms in order to allow for protection for a speaker with these specified parameters.

Another benefit of our device is that it employs feedback control, where the feedback signal is taken directly from the output of the amplifier, which is also the input signal fed directly into the speakers (see figure 2). This provides a greater safety guarantee over another common design involving predictive control, since we have direct access to the signal that is to be seen by the speaker and are not merely trying to predict this signal.

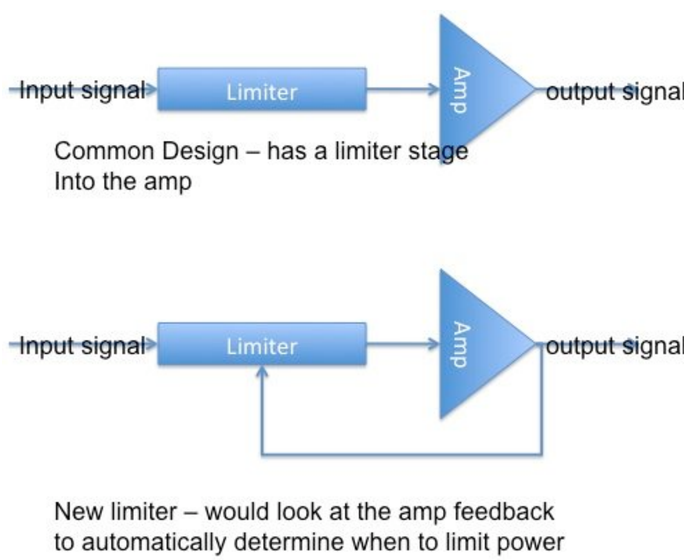

Figure 2: Feedback control

The final benefit of our device design is its attenuation scheme. Unlike other products intended to protect speakers from unsafe levels of input, our product does not simply disconnect our circuit once dangerous levels are experienced, thereby abruptly muting the signal. Instead, our system employs gradual attenuation algorithms that are continuously yet subtly limiting the system input. By beginning to slightly attenuate the signal in the upper range of the safe levels of speaker operation, once the threshold of the speaker's capability is reached the limiting function of our device has already begun and therefore moves seamlessly over this threshold.

CHAPTER 2
ANALYSIS OF COMPETITIVE PRODUCTS

2.1 General Approaches

Essentially, our device is simply a limiter programmed to accomplishing a specific goal. There are many different kinds of limiters that would be physically capable of performing the same signal attenuation as our device, but very few would be "smart" enough to automatically and continuously adjust this attenuation towards the goal of protecting a speaker of a specific speaker.

Within the realm of limiters designed to protect speakers, there are in general two different types: predictive limiters and negative feedback limiters. Predictive limiters are convenient in that they are more portable in general than feedback limiters and aren't necessarily used as dedicated speaker protectors all the time. However, their drawbacks far outweigh this slight benefit. As seen in figure 3 below, the predictive limiters merely predict the output of the amplifiers that the speakers will be exposed to and do not have any connection to the actual voltage the speakers are exposed to. As such, in order for the limiter to adequately calculate the amplifier's output from its input, the amplifier must be held at a constant value. Once the amplifier's gain has changed, the limiter must be

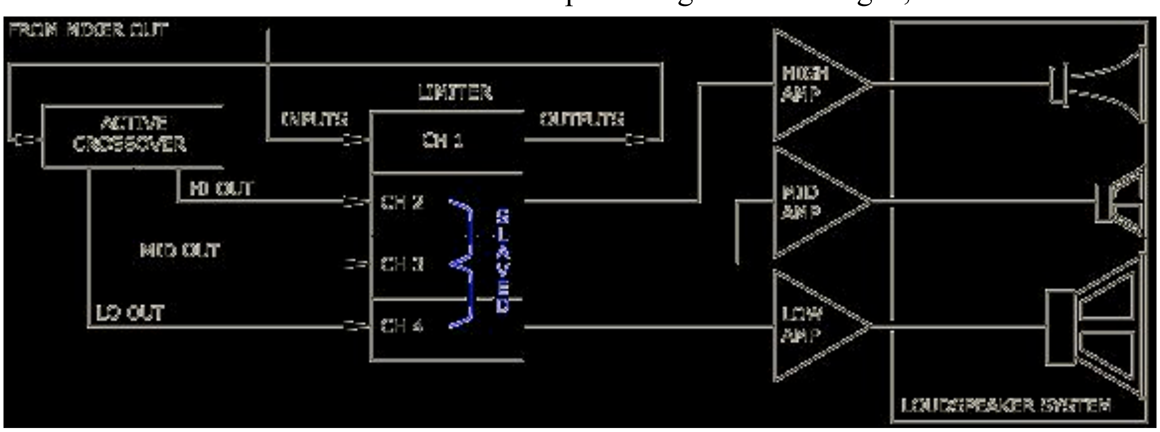

Figure 3: Predictive limiter
Source: http://www.rane.com/note127.html

recalibrated or else the speaker is in danger of being over driven.

Negative feedback limiting is the concept our design is based off of, though we were not aware of the existence of this technology upon the conception of the idea for our project. This type of limiting in regards to speaker protection has become more popular than its predictive counterpart for its many benefits. In general, this limiting technique can guarantee safer, more reliable limiting since it is receiving its feedback directly from the signal line that is being directly fed into the speaker itself (see figure 4 in section 2.2). Because of this, that signal information can be used directly without having to back calculate the gain of the amplifier, hence the limiter does not require recalibration upon each readjustment of amplifier gain. Also, in this set-up knowledge of the parameters regarding the amplifier, such as impedance and power rating, does not have to be as exact as it does for the predictive set-up. Since we are presented with the output of the amplifier, the main parameters we are concerned with in this situation becomes the parameters of the speakers, so that slight variations or discrepancies from ideal amplifier parameters are not as crucial to the functionality of the limiting device in this case.

2.2 Specific Examples

United States Patent #4173740[1] is a device that uses feedback control in an attempt to protect speakers from being overdriven, but instead of actually attenuating the signal as it approaches dangerous levels this device simply "cuts off the supply voltage to a power amplifier circuit in disconnecting a loudspeaker from the output terminal of the power amplifier circuit when an overvoltage is developed at the output of the power amplifier circuit." Our device has a significant advantage over this approach to speaker protection in that we allow the audio signal to continue to play uninterrupted even once the input has reached unacceptable levels because of our limiting scheme.

A low frequency loudspeaker processor called the LSP-1 (see figure 4) is built with a high signal input line to monitor exactly what the gain of the power amplifier is. However, for every different model of loud speakers this device requires a PCB card. The only pre-made cards available are for a few different types of speakers manufactured by their own specific company. If a user wanted to use this device for another speaker not manufactured by the Australian company, they would have to submit a request for a custom card to be made in accordance with their speaker type. With the introduction of user inputs of their speaker specifications, our speaker processor allows the user full customization for this device to work with any traditional loudspeaker model without the need of any kind of chip to be custom made for the device to function properly.

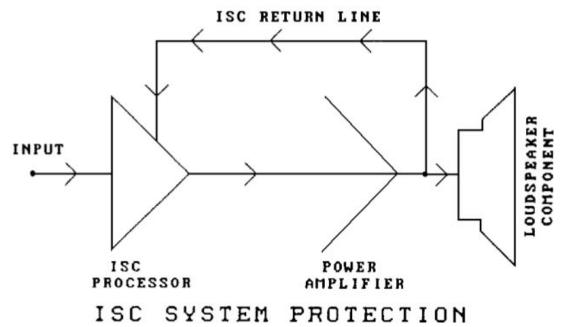

Figure 4: Example of feedback limiting
Source:
http://www.arx.com.au/pdf/old/LSP1manual.pdf

One speaker protection design that is currently on the market, called TruPower™ Limiting by the Meyer Sound Company, however, appears to be much more sophisticated than our design, taking into account other details regarding speaker operations that our design does not consider. However, we must first take into consideration that this design is merely a technology that has been directly incorporated into a company's speakers they themselves manufacture. Since the design is literally embedded within the speaker cabinet itself, there leaves no room for this device to be used on any speaker/amplifier combination the user wishes and therefore does not offer the same freedom of speaker choice to the user as our device does.

TruPower™ is literally designed to calculate the "true power" that is being delivered to the speaker. For example, though in our device we assume that the impedance of the speaker (as specified by the user as an input to our system) remains a constant, this impedance actually varies slightly both over the frequency range of the signal and over the operation time of the speaker as the speaker coils warm up and experience a slight increase in impedance. The impedance increase in the speaker coils causes a slight drop in the overall dynamic range of the speaker system over time and this TruPower™ technology seeks to correct this problem. On the opposite end of the spectrum, the frequency dependency of the speaker impedance can cause the speaker impedance to unknowingly decrease at certain frequencies and thus open the speaker up to potential damage occurring at signal levels that under normal impedance would have passed as harmless. Though this frequency vs. impedance relationship is likely hard to determine directly, this technology seeks to prevent any speaker overload due to this phenomenon by automatically attenuating the signal by a couple extra dB after a long period of operation time to prevent overheating of the speaker.

CHAPTER 3
CONCEPT/TECHNOLOGY SELECTION

3.1 The Attenuator and Analog vs. Digital Considerations

We knew from the start that the three major functions that needed to be performed by our device were the collection of user input and feedback loop data, the processing of that data to form the appropriate attenuation algorithms, and, finally, the actual performance of the attenuation on our input signal. We knew that in order to interpret and process information regarding analog data, we would likely find it best to perform our processing in digital form. Our initial idea was to first perform analog to digital conversion from our feedback signal to determine the amount of attenuation needed, and then to go back and attenuate a digitized form of the input signal by that amount before sending the analog version of the newly attenuated input signal through the amplifier. Under this design, we thought it would be best to use either a digital signal processor (DSP) or a microprocessor to carry out the attenuation of the signal in its digital form. However, the problem with using the microprocessor for this purpose was a fear of severe distortion of the audio signal, as a microprocessor would likely not be able to produce a high quality of sound once the signal had been converted from analog to digital form and then back again. The option of using a DSP chip would be beneficial in that not only is it commonly used in manipulating audio signals and reproducing a high quality of sound, but also it would be capable of some very complex limiting algorithms and would even give us the option of using frequency distinction in our attenuation scheme. However, neither of us had very strong knowledge of digital signal processing nor any practical experience in the use of a DSP chip. We were both enrolled in the Electrical

Engineering Department's Introduction to Digital Signal Processing class at the time, but feared we would not learn enough about the practical functionality of such a device in order to successfully incorporate this into our project with only a month or two of lessons.

Our next idea was to employ the use of a potentiometer to control the attenuation of our input signal in its original analog form, avoiding the possible distortions and complications of digitization. However, a traditional analog potentiometer would need to be controlled by our system's "brain" and the only way to control such a potentiometer is by physically turning a knob. The only way to accomplish this with electrical signals would be through control of a motor, which would involve adding another component to our device that would make matters unnecessarily complicated. To circumvent the problems presented us by the traditional analog potentiometer, we decided on using a digital potentiometer that could be controlled directly by the "brain" of our system. The digital potentiometer would contain physical wipers and resistors so that the attenuation of our input signal should preserve the quality of our audio signal. The drawback to this is that the digital potentiometer, unlike the analog potentiometer, has discrete positions for the wiper to be located at, whereas the analog potentiometer's wiper position can continuously span between the high and low terminals of the potentiometer. The benefit of being able to directly control the wiper's position by the same device that will perform the attenuation algorithm calculations and in a very accurate and precise manner, however, was seen as a benefit well worth the loss of a continuous spectrum of attenuation from our potentiometer, and thus the digital potentiometer was chosen over its analog counterpart.

3.2 The "Brain"

We then needed to select a device that would act as the "brain" of our device. This would need to first take in our user inputs and our feedback loop information; from this it would not only determine the attenuation algorithms necessary to protect the speaker in question but then send commands to perform that attenuation. Once we had decided that using a digital potentiometer to perform the attenuation would be best, it was an obvious choice to use a microprocessor for this purpose. Once this was decided, the two main considerations for settling on a final device were the type of microcontroller to use and what size of a microcontroller was needed. Our choice of an Atmel brand microcontroller over a PIC (Peripheral Interface Controller) controller was made on the basis of code familiarity. Bethany was in charge of the software programming of the microprocessor, so she chose the c programming coding language of the Atmel controller over the basic coding language of the PIC due to her experience with the related C++ programming language. Regarding size considerations, with our experience with the Atmega 32 microcontroller and our general estimation of the number of pins and the types on functions that would be needed (analog to digital conversion, communication with the digital potentiometer, dedicated user inputs, and potential LCD output pins) we decided that this device would be sufficient to meet the needs of our project.

3.3 Digital Potentiometer Model Selection

To begin with, we had very little knowledge regarding not only the types and different features of digital potentiometers but also regarding which of these features would be most suitable and beneficial to our device. What we did know, however, was that we were to be attenuating an audio signal and that detecting changes in audio signals in our brains is calibrated in terms of a logarithmic, or decibel (dB), scale. Therefore, if we were to use a linear-based digital potentiometer and were to start attenuating our signal in linear correspondence with the increasing input signal, it would appear at first that there was a huge decrease in the sound being produced by the speaker, and then as further attenuation occurred as the signal got louder and louder there would seem to be almost no attenuation occurring at all in the signal. We were able to circumvent this problem fortunately by finding digital potentiometers designed specifically to deal with audio signal attenuation with wiper positions at taps positioned along the resistor to allow resistance changes to occur on a decibel scale instead of on a linear scale.

Our initial approach was to order a wide variety of audio-tapered digital potentiometers and to investigate which one we felt would be best suited for use in our project. We ordered the following devices from the Maxim/Dallas Semiconductor company: DS1801, DS1802, DS1808, DS1881, DS1882, MAX5411, and MAX5486 and researched the features of each device. Our initial decision was to use the DS1881 model that provided 63 wiper steps of 1 dB per step, a power supply range within the 5 V limit of the microprocessor we were using, zero-crossing detection to eliminate the clicking and popping sounds that tend to accompany wiper position changes in audio chains, and an option to store wiper position information in an EEPROM upon power off. However, we were forced to abandon using this potentiometer once we discovered that the difficulty of using the Atmega 32 microprocessor for I2C (Inter-integrated Circuit) communication far surpassed that of using the SPI (serial peripheral interface) communication required for the MAX5411 potentiometer.

The functionality for carrying out SPI communication was built right into the hardware configurations of the Atmega 32 microprocessor as accessed by the dedicated pins at port B. The Atmega processor already had registers set up in order to allow the parameters of the SPI communication to be adjusted to suit the needs of the given SPI communication application and some simple sample code was already presented in the data sheet to initiate SPI communication and to begin sending information via SPI communication. The only major drawback to using this new potentiometer model was the fact that this model only allowed 32 wiper positions of 2 dB per step, which would result in less resolution in our attenuation accuracy and a slight loss of smoothness as the wiper transitions took place. It was determined that the ease in communication format and microprocessor coding was well worth this slight sacrifice in attenuation quality.

3.4 Feedback Signal Incorporation

Our next task was to find a way to tap into the signal coming out of the amplifier and feed back a version of this signal into the microprocessor where it could easily and safely be measured by the microprocessor in determining the attenuation algorithms necessary. The main problem here was that once the small signal input was fed through the amplifier, it was at voltage levels that were unsafe to be directly fed into the microprocessor, which could only safely handle voltages roughly in the range of 0-5.5 Volts. Therefore, we needed to design a circuit that would step down the voltages from the amplifier from their current values to lie within the 0-5.5 Volts range. We also needed to only submit positive voltage values to the microprocessor, so we decided to add step down circuitry that probes the voltage output fed to the speaker. Since this output is a 100-volt peak to peak swing an op-amp steps the voltage down by 1/10. Then op-amp circuitry rectifies the negative voltages and then an active low pass filter smoothes it out. This is explained further in the hardware section.

3.5 A Temporary Sidetrack

Early on in our project's development we encountered an alternate route that seemed like another adequate way we could solve our problem of speaker protection. The website for the THAT Corporation's THAT 4301 Analog Engine (see figure 5) included an application note on how this device could be used as a "Signal Limiter for Power Amplifiers." This chip sounded promising and included many components of the design we had already envisioned all on one chip, with only the addition of a couple smaller components such as capacitors, resistors, and diodes needed. Another benefit, besides convenience, that this new solution had to offer was its use of an analog voltage-controlled amplifier, which would allow the attenuation to occur through a continuous set of values. This would provide smoother-sounding volume transitions over the discrete wiper position changes that would be necessary in our use of a digital potentiometer. Also, the device not only was designed to operate by taking in the feedback signal from the amplifier's

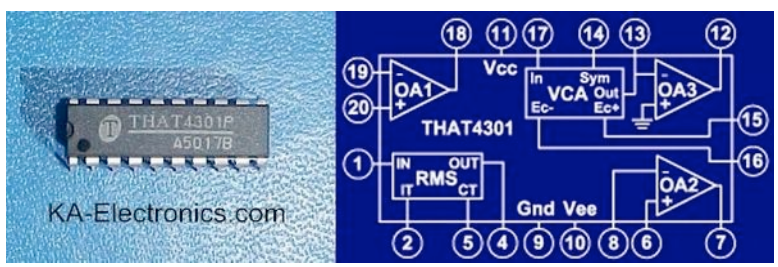

Figure 5: THAT 4301 Analog Engine
Source: http://www.ka-electronics.com/that4301/that4301.htm

output, similar to our original design, but it additionally contained an RMS detector to enable that feedback signal's strength to be more readily detectable by the device. This would eliminate the need of our originally intended "step-down circuitry."

However, there was one crucial drawback to this design that made it implausible to use: it had no real user-defined flexibility. One of the main distinguishing features of our device is its capability to be used with any speaker the user wishes, where the user only has to select two simple parameters of their speakers and our device will automatically recalibrate itself to fit their speaker's needs. This new design was meant to operate around a threshold value and automatically attenuate the audio signal past that value, but this threshold value was determined by a relationship of resistance values used in the circuit. In order for such values to be extracted from the user's input and then a determination to be made about how the attenuating algorithms operate according to these values, we would require the use of a "smart" devise to take in such inputs and carry out the necessary calculations. The only real logical solution for this is to incorporate the use of a microprocessor, for there is no functionality in the Analog Engine device that would allow for such calculations to be made. An option would then be to use a microprocessor along with the circuitry recommended for the "Signal Limiter for Power Amplifiers," but then the problem would become both an unnecessary increase in complexity between the microprocessor to Signal Limiter circuitry and the inflexibility to completely control all features of the Signal Limiter circuit due to their dependence on set resistance values that would be hard-wired into the circuitry. The Signal Limiter circuitry does allow some flexibility in setting the limiting threshold and compression ratio by incorporating 3 different variable resistors, but even so many of its parameters are set using a combination of these variable potentiometers and set resistors. Even if we could set up our system to use digital potentiometers in place of these variable resistors so that we could control the threshold and limiting parameters with our microprocessor, this would seem to be at least three times as complicated as our original design involving communication with only one digital potentiometer and without the worry of having to hard-wire some parameters of the limiting into our circuit through the use of set resistor values. Therefore, we decided that our original design would be the simplest and most efficient way to accomplish our goal of speaker protection.

CHAPTER 4
PROJECT ARCHITECTURE

4.1 From Input to Amplifier

A very high-level system diagram can be seen in figure 6. The very first function our device serves upon system power up is the input of the user's speaker parameters. This is accomplished through the use of toggle switches. The user can select speaker impedances (2, 4 or 8 ohms) and speaker power ratings (100, 200… 1000 watts). With this information the microprocessor calculates what the threshold voltage is to start attenuation (the details of such calculation will be explained in section 4.3). The audio signal itself originates from some electronic instrument, microphone, etc. as a very low-voltage signal representing the pressure variations in the air produced by the music desired to be amplified out to the general public. This signal travels through the power amplifier where it experiences an amount of voltage gain depending on the position of the "Volume" knob on the amplifier. We will assume for our project that this gain can be varied at any point in time and do not require it to be set at a constant level for any amount of time. Here is where the signal branches off: we have this amplified signal both connected to the speaker, where it will act as a transducer converting electrical

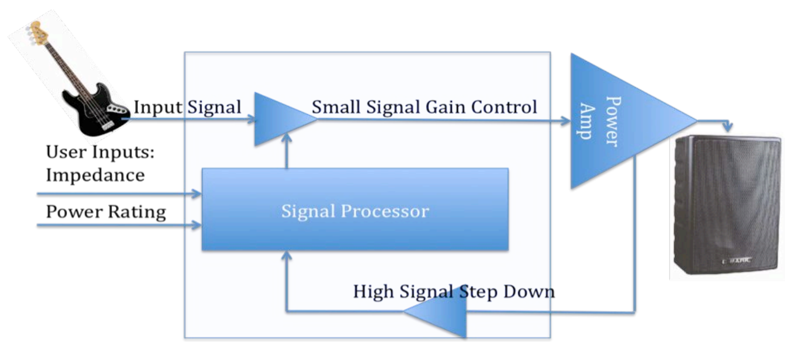

Figure 6: High level system diagram

waves into pressure waves in the air to produce sound, and we also run this signal through a step-down circuit that will attenuate and manipulate the signal in preparation for its entrance into our Atmega 32 microprocessor. The current step-down circuit attenuates the voltage signal by .08x and then rectifies it and smoothes it out with a low pass filter. The hardware architecture for this circuit is further explained in chapter 5, the hardware section.

4.2 Inside the Microcontroller: Analog to Digital Conversion

This small signal version of the amplifier's output is fed into a pin on our microcontroller from port A (see figure 7 for the software architecture of the microcontroller). (Please note that all the code for our Atmega 32 processor is included in the Appendix.) This port's pins are linked to dedicated hardware responsible for controlling the microprocessor's analog to digital conversion functionality. We utilize the microcontroller's ADC (analog to digital conversion) registers to configure and carry out such an operation, converting our small 0-5.5 V signal from the feedback loop into a set of discrete integer numbers valued between 0 and 1023 (corresponding to a 10-bit digital result). Here we have the difficulty of an analog signal at our port A pin that is not only is not sitting at a constant voltage level due to the changes in the amplifier's gain, but also fluctuates in a half-rectified fashion between 0 Volts and its peak voltage even while the amplifier gain remains constant due to the sinusoidal characteristic of musical frequencies. To be able to accurately interpret the signal level we first acquire sets of 100 samples of ADC readings and then perform statistical analyses (namely, calculating the average value and the maximum value) on each set of data in order to be able to make a better decision on how much attenuation is needed based on each sample period. The goal here is to, in general, set up our attenuation to cater to the peaks experienced in order to allow for maximum protection of the speakers, but to ignore this peak value and instead use the average value in the case that a very short-lived spike in voltage is experienced that is well above the rest of the values of that sample (see code in the Appendix for details regarding this selection criteria). If we were to only observe the peaks as a rule, we would always be responding to every little voltage peak in the system, which would lead to both a general sense of over-attenuation and a significant drop in the dynamic range experienced by the listener at the speaker output and lead to many abrupt and dramatic attenuation leaps between load and soft as any intermittent spike was encountered.

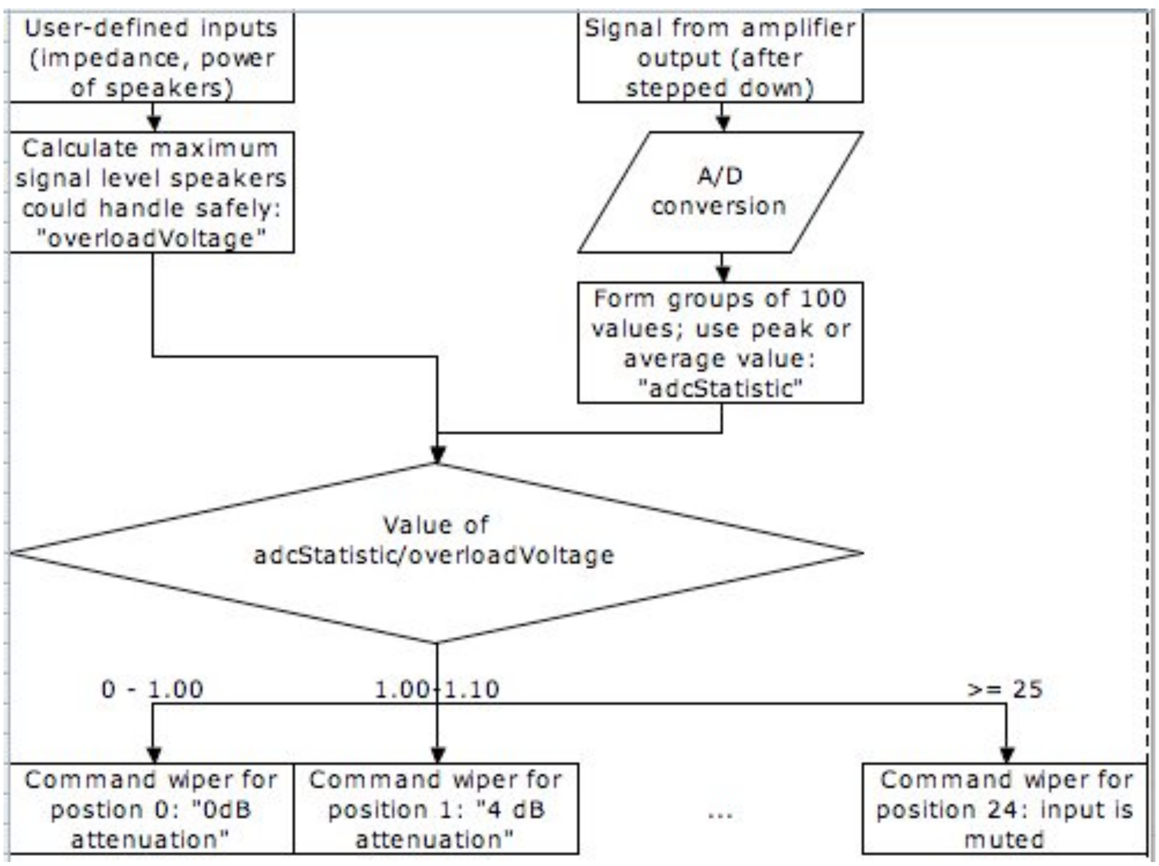

Figure 7: Software architecture

4.3 Inside the Microcontroller: Attenuation Determination and Implementation

Once we have the peak or average information from the ADC results, the user's inputs that were read into the system upon the device's power up come into play. These user inputs of speaker resistance and power rating are used to calculate the software variable called the overloadVoltage, which is the threshold determining safe operation of the user's speaker.

> As a short side note: we are using the term "threshold" more loosely for our actual attenuation implementation where we allow the signal to float close to and even slightly above this threshold level and consider this safe operation for the speaker. This is due to the lax definition on how the actual speaker rating is calculated, as mentioned in the Introduction, where the threshold is assigned to an average level over time and allows for slight up and down fluctuations in the music around this level.

The overloadVoltage equation is: $overloadVoltage = (\sqrt{wattage\_rating \times impedance} \times .08 \times 1023)/5$. This comes from the fact that: $Power = \frac{voltage^2}{resistance}$, so then $Power \times resistance = voltage^2$. The step down circuitry attenuates a signal to .08 of its peaks, so $voltage = \sqrt{Power \times resistance} \times .08$. Voltage is the overloadVoltage, power is the power a speaker can take, and resistance is the speaker's impedance, which is commonly 2, 4 or 8 ohms. Below is a chart of common speaker impedances and power ratings that the user can set with their corresponding overloadVoltage values:

overloadVoltage [decimal] 0~1023	overloadVoltage [volts]	Impedance [Ohms]	Speaker Power Rating [Watts]
231.4784759	1.13137085	2	100
327.36	1.6	4	100
462.9569518	2.2627417	8	100
327.36	1.6	2	200
462.9569518	2.2627417	4	200
654.72	3.2	8	200
400.9324811	1.959591794	2	300
567.0041524	2.771281292	4	300
801.8649622	3.919183588	8	300
462.9569518	2.2627417	2	400
654.72	3.2	4	400
925.9139036	4.5254834	8	400
517.6016074	2.529822128	2	500
731.9992131	3.577708764	4	500
1023	5	8	500
567.0041524	2.771281292	2	600
801.8649622	3.919183588	4	600
1023	5	8	600
612.4344811	2.993325909	2	700
866.1131492	4.233202098	4	700
1023	5	8	700
654.72	3.2	2	800
925.9139036	4.5254834	4	800
1023	5	8	800
694.4354277	3.39411255	2	900
982.08	4.8	4	900
1023	5	8	900
731.9992131	3.577708764	2	1000
1023	5	4	1000
1023	5	8	1000

Table 1: overloadVoltage Calculations given all possible user input combinations

Once this voltage limit value has been scaled down by the same proportional amount used on the voltage during the step down process (0.08) in order to put it in the 0-5 Volt range, that value is then multiplied by a factor of (1023/5) in order to put the parameter into a value on the same scale as that of the data from the ADC process, as a value between 0 and 1023, so that the two values can be compared accurately. This newly calculated threshold value is saved by the software as the variable "overloadVoltage" and is carried, along with either the peak or average parameter from each ADC set, as an argument into a function establishing attenuation levels based on a comparison between the ADC set statistic and the threshold voltage for safe operation. For example, if the peak voltage from a set of ADC conversions is equal or less than less the "overloadVoltage" value, the first attenuation bracket bracketNumber = 0) would send a command to the digital potentiometer to position the wiper to allow for 0 dB attenuation. With increasing ADC peak or average values in relation to the "overloadVoltage" value, the wiper position is commanded to position itself to allow for increasing amounts of attenuation. Also incorporated into this same function is a calculation of how drastic of a change in position a wiper must undergo in order to move from its current location to the newly calculated location and based on this calculation send the wiper through an appropriate amount of intermediate steps in order to more smoothly transition from one attenuation level to another. For example, if we had a wiper in a position corresponding to bracket 0 (for no attenuation) and we wanted to move this wiper to the position corresponding to bracket 12 (to reduce the signal to about 1/5 of its current value), we would command the wiper to move through two other intermediate positions on its way from the bracket 0 position to the bracket 12 position.

4.4 Practical Adjustments

One of the lessons learned from this project is that no matter how much thought, analysis, and calculation you put into a design, once you test that design there is a great likelihood that you will encounter some issues that never occurred to you from a theoretical standpoint. This was the case with the attenuation algorithms that had been coded in the software portion of our system. During the final testing stages of our device our code worked perfectly, but that fact ended up being our only problem. During testing, when the amplifier was causing the feedback signal to fall in the upper ranges of attenuation, we would see the code step through the steps needed to reach the destination attenuation bracket that had been specified for it in our code and then once it got there, as a verification that the attenuation levels had been indeed calculated correctly, we saw the ADC readings to go from the high values of the amplified signal to low values that were well within the overloadVoltage safety threshold value. The problem, then, was that the next command in our code was to run the function again to calculate an appropriate attenuation bracket for the feedback signal. Since this function had already run once and had done its job of bringing the signal into the range where it would not need further attenuation, this second time around the code put the attenuated signal into bracket 0, where no attenuation was required and the potentiometer's wiper would be moved accordingly. We must realize here, though, that by lowering the level of our small signal audio input when we first commanded the attenuation, we did not change the high gain setting present on the amplifier, so that once we bring this attenuation back to 0 dB, the high amplifier gain is still present. Therefore, essentially the current code was causing the attenuation for a high feedback signal to toggle between a given attenuation level and 0 dB of attenuation: once through the attenuation function and the attenuation would be set, a second time through and the level would be in the safe range again so the attenuation would be unset. This behavior both made the sound output from our system to be distracting, unnatural, and aesthetically unpleasing and caused the output signal going to our speaker to oftentimes reach very high levels that would very likely allow for damage to the speaker to occur.

In order to change this, we only had to modify a few lines of our code, but this drastically changed the behavior of the system in practice. We decided that instead of calculating a destination attenuation bracket the potentiometer should reach based on its current level (which would lead often to the toggling 0 dB bracket once the attenuation did its job of bringing the signal back to safe levels), we should have the system only change its attenuation bracket one at a time. To do this, we used the framework of the existing attenuation bracket definitions to assign the current level to a bracket number (the bracket numbers 0 – 23 correspond to wiper settings 0 – 46 dB, with a multiplication of 2 factor to convert between the two) and then compared this bracket number with the last bracket the wiper was assigned to. If the new bracket was lower than the previous one, that meant that less attenuation was needed than was previously set, so our system commands that the wiper position be moved +2dB from its previous setting. If, on the other hand, the new bracket was higher than the previous one, that meant that less attenuation was needed than was previously set, so our system commands that the wiper position be moved -2dB from its previous setting. After making only this slight 2dB change in the wiper position within our attenuation function, the code loops back to again take a new ADC reading and calculate a new bracket number, where it can be determined if the wiper should be moved for more or less attenuation. Also it should be noted that if the signal has not changed levels enough to associate it with a different bracket number, then the wiper is simply commanded to stay at its current position. By employing this new algorithm, the safety threshold is reached a bit more slowly, but once it is reached it stays within a reasonable limit of the threshold (per our "lax interpretation of the threshold" explained in section 4.3), which is the main point of the speaker protection function of our device. See figure 8 for the new software flowchart.

Under this new algorithm, the initial purposes for the bracket definitions are nearly lost. Originally, these brackets were meant to calculate out where a wiper should be, based on the signal level presented to the ADC sampler, for that signal to lie within the threshold value. However, since we are not sending the wiper automatically to this pre-calculated level but instead changing the attenuation only step-by-step and checking each time to see if the level has yet reached safe values, these levels are only used as a means of comparison. The function they now serve is to tell how far away the new ADC voltage level has to be from the old ADC level in order to command the 2 dB change. In this function, then, the bracket definitions are still serving a useful purpose, so that the wiper's position will not be changed every single cycle of the function. Also, built into the attenuation bracket calculations is the natural dB characteristic of the potentiometer's wiper, so that signals requiring only a low level of attenuation will be more sensitive to voltage changes

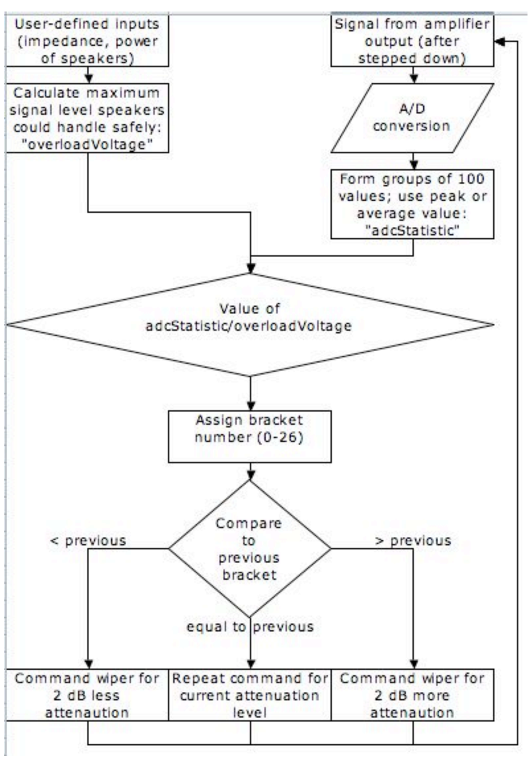

Figure 8: Revised software flowchart

whereas it will take more drastic changes in ADC voltage level for changes to occur at higher levels of attenuation.

4.5 The Analog Attenuation

The digital potentiometer changes resistance and is input into the non-inverting terminal of a TLC2272 op-amp. As the potentiometer's resistance goes down, the resistance lowers and the signal attenuates. The output of the op-amp is fed back into the input terminal, making it a simple voltage follower. The low terminal of the digital potentiometer receives a 2.5 Volt DC signal. The high terminal of the potentiometer receives the music with a Vcc/2 = 2.5 Volt bias. To make sure no DC component is present in the original signal, coupling capacitors were used on the input to the resistors and on the output of the TLC2272 op-amp.

CHAPTER 5
HARDWARE

The premise of the hardware for the limiter and speaker protector was using a potentiometer to attenuate the small signal of an iPod or guitar before the signal reaches the input of the guitar amplifier. The potentiometer would be digitally controlled by an atmega32 microprocessor using SPI protocol serial communication. The processor made the decisions of controlling the digital potentiometer to change the music volume based on a signal fed back from the amplifier.

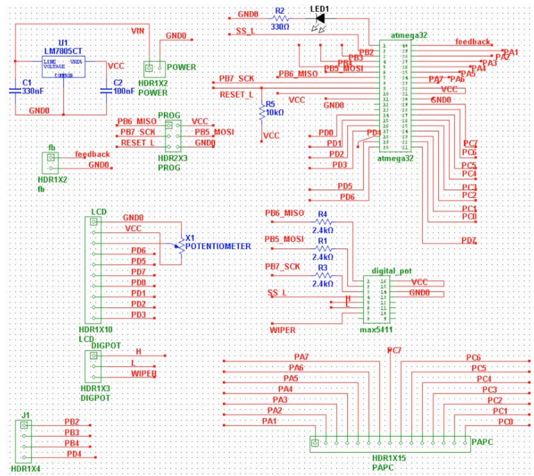

Figure 9: Digital PCB

5.1 The Digital PCB

It was decided that one printed circuit board (PCB) would be built to handle the digital components and another board would handle the analog elements. The first digital board was built to house the atmega32 processor, digital potentiometer (maxim, Max5411), 12-volt to 5-volt regulator (LM7805), and headers that plug into an LCD display. Figure 10 shows how a cosine signal was attenuated using the digital potentiometer.

The MAX5411 was chosen because it had a logarithmic taper digital, with 32-tap points each. It can replace mechanical potentiometers in audio applications

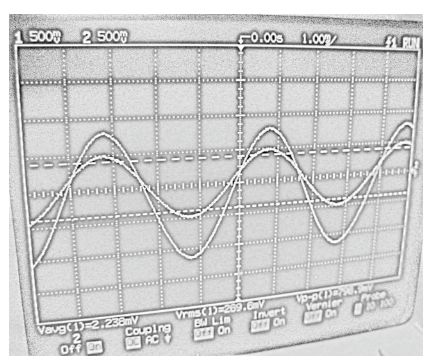

Figure 10: Attenuation of a cosine signal

requiring digitally controlled resistors. The chip also has a SPI-compatible serial interface that controls the wiper positions. The MAX5411 has a factory-set resistance of 10kΩ. A zero-crossing detection feature minimizes the audible noise generated by wiper transitions. Switching amplitudes through a digital potentiometer when there is any sort of voltage cause small audible cracks. The zero-crossing feature has the potentiometer wait until the low input and high input are equal to change its resistance. For the purpose of mounting the MAX5411 on our PCB the 16-pin quarter-size small-outline package (QSOP) package was chosen. 10 mil traces were routed to the 16-pin QSOP.

The digital board would receive the audio signal needing attenuation through its high input on the digital potentiometer header. Since the digital potentiometer's power supply is ground the 5-volts the music needs a vcc/2 bias voltage. The music into the high input terminal of the pot is then biased by 2.5 volts. The input to the low terminal is simply 2.5 volts. The wiper is then connected to a voltage follower op-amp is then plugs into the large personal amplification system.

5.2 The Analog PCB

The digital PCB also received a feedback input into its analog to digital conversion port (ADC). This feedback input was stepped down using the analog board. The schematic for the analog board is shown on the right. The board consists of two power regulators to regulate +12 volts and -12 volts to +5 volts and -5 volts. A number of resistors were using to bias to high and low terminals, which are sent to the MAX5411. Two large 33uF capacitors were used to couple and DC signal connected to the music inputted. Two capacitors were used because 33uF capacitor are polarized electrolytic, thus the terminal charges must be symmetric. So, the two positive sides of each capacitor were connected leaving the negative terminals on the outside. The power was regulated to +5 volts and -5 volts because the op-amps used the analog PCB circuitry required +/-5 volts. Fast rail-to-rail op-amp built by TI (TLC2272) was used as a voltage follower for the small signal output.

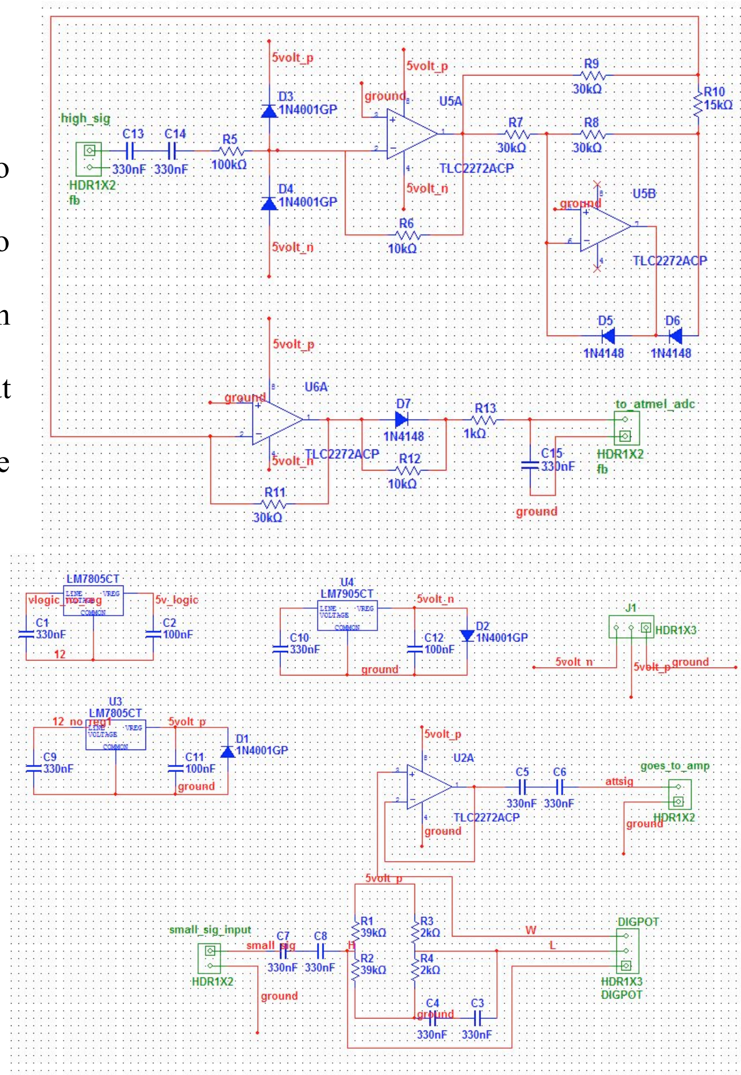

Figure 11: Analog schematic

The TLC2272 was also used as a step-down op-amp. Since the large personal amplifier outputs a high voltage signal of up to 100 volts peak-to-peak. The first stage (U5A) of the step-down circuit op-amp reduces the gain of the signal to 10 volts peak-to-peak. The next two op-amps (U5B and U6A) full wave rectify the signal. To smooth out the signal a low-pass filter was built with a capacitor and resistor. The full-wave rectification circuitry was based on figure X from the book "Introduction to Operational Amplifier Theory and Applications" (Huelsman,1975). The top of figure 12 shows how a simple cosine signal swinging from +50 voltage to -50 volts is rectified and then smoothed out to be about 4 volts. The bottom of figure 12 shows how the rectified signal changes with music and jump up and down depending on the music's amplitude. The figure below takes one through the step of how the cosine signal voltages look at each stage of the analog board.

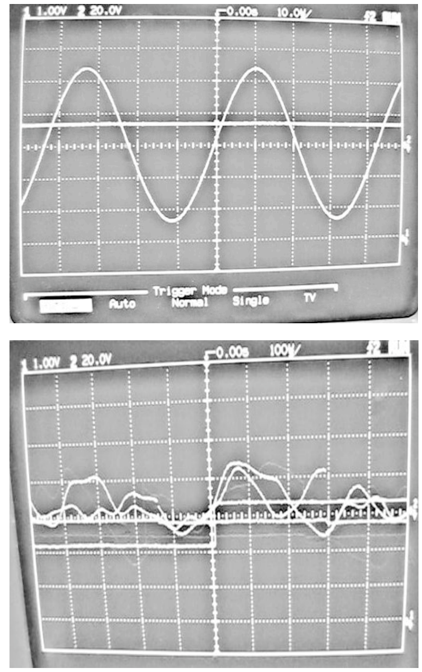

Figure 12: ADC step-down vs. high-voltage cosine (top) and high-voltage music (bottom)

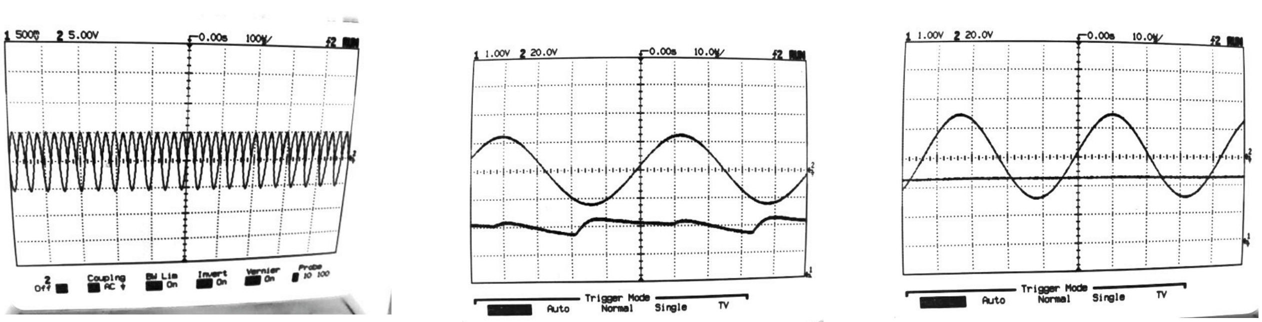

Figure 13: Full-wave rectification (left), diode in series (middle), low pass filter (right)

CHAPTER 6
SOFTWARE

6.1 SPI Communication

The use of Serial Peripheral Interface (SPI) communication was required for our microprocessor to be able to control the digital potentiometer model we had chosen: the MAX5411. As mentioned in chapter 3, the prime reason for using this SPI format of communication was that the Atmega 32 chip contains pins that are connected to internal hardware designed to carry out such communication. Also, to correspond to such hardware, there are registers featured on the microprocessor that enable the SPI hardware to be configured and utilized easily.

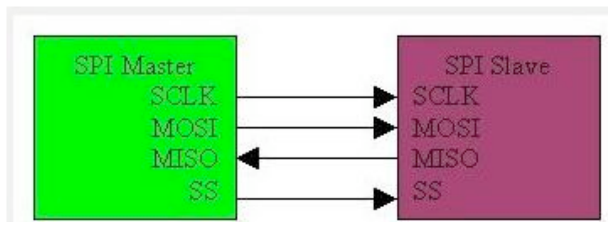

Figure 14: SPI communication wiring
Source: http://dev.emcelettronica.com/category/tags/spi-protocol

As shown in figure 14, SPI communication is carried out between two devices (the devices can be daisy-chained so that one master can communicate with multiple slaves, be we only require two-device communication in our project), one being specified as the "master" device and the other as a "slave" device. In our case we configure the internal SPI Control Register (SPCR)'s bit corresponding to master mode (MSTR) to have our microprocessor act as the master who is then responsible for sending data to the digital potentiometer slave (which acts as a slave by default, since it lacks the capabilities to perform the duties of a master device, such as generating its own clock signal). SPI communication requires 4 different wires running between the two devices as shown in figure 14, labeled SCLK (serial clock), MOSI (master out, slave in), MISO (master in, slave out), and SS (slave select). The SCLK line is the clock that synchronizes the timing between the two devices. This clock signal is generated by our microprocessor master and its frequency is determined by setting registers corresponding to certain division factors from the internally generated clock signal always present in the microprocessor. The MOSI line is the most important connection in our SPI communication system. This is where the transmitted message is sent as a series of ones and zeros from the microcontroller to the digital potentiometer. The timing of the transmission of this message is determined by the SCLK signal and the relationship between the phase and polarity relationship between the clock signal and the message signal are set up using registers in the microcontroller; this phase and polarity must match the phase and polarity required by the SPI slave you are communicating to or else the signal will not be interpreted correctly. The MISO line is meant to enable the slave to "talk" back to the master device, but is not used in our project. The potentiometer would be reading back to the microprocessor the current position of its wiper via the MISO line, but in our case we are merely concerned about telling the digital potentiometer where to position its wiper and based on that command from the master, as long as the slave device interprets the commands correctly, we should already have knowledge of the current wiper position at any point in time. The SS line is an active low line where the master prepares the slave device to receive a transmission. This line is most important when more than one slave is present in a system, where the master can define many SS output pins to be able to select each slave device to talk to one at a time. The master's SS line is usually connected to the slave device's CS (chip select) active low so that only the chip that is selected by the master will be able to actively interpret the master's commands. This same connection is made between our microprocessor and our digital potentiometer's chip select pin. The details about the functionality of this line in regards to our single-device project will be explained below in section 6.2.

6.2 Trial and Error, Lessons Learned

Our first step in creating our device was to ensure the functionality of the digital potentiometer, which was not only the pivotal function of our device, responsible for ensuring speaker safety, but also likely one of the most complex, due to the SPI communication protocol intricacies. At first, we made the mistake of ignoring the sample code provided in the Atmega 32 processor, thinking it would not provide fully SPI communication functionality, so we attempted to get some bits of sample code from online databases. Once we constructed an initial bread board-based design in order to facilitate a piece of sample code (commanding the digital potentiometer to move through each of its wiper positions) and hooked up a multimeter to the wiper to try and verify if it was in fact following the commands sent by the microprocessor, we were seeing voltage changes on the wiper and interpreted this to mean at least a partial success of the potentiometer's functionality. We learned on multiple occasions, however, that seeing an apparent movement of the wiper, unless it exactly matches the value your code is setting it to and is consistent upon running the program may times, does not at all mean that the SPI communication code is working correctly. The two major mistakes we initially found were that first we did not have the digital potentiometer wired up to power and ground correctly and secondly that our sample code was much too complex and outdated.

From this realization, we discovered that the sample SPI code for master mode operation on the microprocessor's data sheet was all the code you needed in order to carry out SPI communication. With our new set of code, we were seeing some very perplexing results. For example, on a couple occasions, we would observe the voltage on the wiper to be showing appropriate values most of the time, but would skip many values or oftentimes jump between the power or ground voltage levels in the middle of a voltage progression that was supposed to be in between those two values. We attempted to find our problem by inserting some debugging code that blinked an LED (light-emitting diode) at crucial points in our code to ensure it was running properly, but sometimes even this method was no use when the LED seemed to show that the system was running through each line of our code properly, but the wiper position was not changing at all.

Finally, we made our first step towards realization of our problem when we went through a simulation of our code line-by-line on the AVR Studio 4 program (the same program we were using to write, compile, and program our code into our device with), checking that the register values that we were setting in our SPI initialization command were setting properly and that our code was in general operating correctly. Through this process, we immediately discovered that the bit in the SPI Control Register (SPCR) that was responsible to setting the microprocessor to operate as the SPI master (the MSTR bit) was being set properly, but was being unset in the very next, completely unrelated command. Upon receiving some guidance from a TA (teaching assistant), we discovered that the active low chip select (CS) pin on the digital potentiometer worked more like a reset pin than it did like a traditional chip select pin whereby the device is enabled to work for the duration of time that the active low select signal is held low. Therefore, instead of outputting a continuously low signal from the SS pin of the microprocessor to the CS pin of the potentiometer, we would need to start with the SS pin in the high voltage position, transition this to a low right before communication with our potentiometer began, and pull it back high again once one byte of information was successfully transmitted. Prior to this, under the impression that we would simply need the CS pin on the digital potentiometer to always be low, signifying that we always wanted to talk to that chip, we had simplified the connections between the two devices by simply tying the CS pin on the microcontroller to ground and ignoring the SS pin on the microcontroller. Right away we saw this new “CS pin with reset functionality” concept, we knew that we needed to run a wire between the microprocessor and the potentiometer in order to enable communication to the device each time before any byte of information could be transmitted.

After some further research on our MSTR bit-setting issue, we found this sentence in a Wikipedia article entitled “AVR SPI C Snippets”: “When running master mode, a low level at the slave select pin SS will force the device into slave mode.” At first this seemed completely contradictory to our basic understanding of SPI communication, for clearly figure 14 shows the SS pin on the SPI master connected to the SS pin (in our case, called CS: chip select) on the slave, both being active low, with the master being in charge of selecting which device it will be talking to. However, we were not cautious in noticing a slight technicality in the operation of the SS pin on the microprocessor. In our case, we did not need the pin specified as SS on the microcontroller, for its primary function was in fact to operate as its own version of our potentiometer’s chip select pin in the case that someone were to need the capability of having the microprocessor serve the function of a SPI slave device, where this pin would be configured as an input and then driven low by an outside SPI master device when communication to this microprocessor was desired. However, the simple fact that this functionality was not needed by our system did not mean that we were safe in ignoring this pin. If this pin is configured as an output pin while the device is in master mode (for it cannot be configured as an output in slave mode or it would lose its ability to be selected as such a slave), it becomes a general-purpose output pin and the SPI communication system is not affected by this pin's value. However, we were not observant of the fact that in the code from the Atmega 32 datasheet we were using they left this SS pin set as an input, expecting the user to desire the microcontroller's option to act as a SPI slave (note that inputs are configured to a port value of 0, so that through the lack of mentioning the input/output functionality explicitly in the code, the SS pin's function as an input was implied). The simple fact of the SS pin being configured as an input would not have caused any problems, however, had this pin not been an active low pin. Unfortunately, an input pin with no signal to drive it high is sitting at a low value. Therefore, under the configuration we were using, our microcontroller was being informed that it was required to act as a slave and was not actively trying to send information but to receive it. Our problem was fixed once we combined these two bits of insight into the functionality of the chip and slave select pins of our two devices: we first set the SS pin on the microcontroller to be an output pin and then connected a separate general purpose output pin, having no ties to the SPI system itself, to act as an output signal to the potentiometer's CS pin that would carryout the reset-like function required for communication with our slave device.

6.3 Attenuation Bracket Considerations

The goal of our attenuation is to gradually limit our input signal as it approaches and exceeds levels that would be considered unsafe for the speaker in consideration. Since all of our considerations regarding what signal levels to attenuate are made based on this threshold of safe limits for our speaker and the safe limit for our speaker's operation is based on the impedance and power rating parameters as defined by the user inputs into the system, we first must establish this safety threshold which is referred to as the variable "overloadVoltage" in our code. As briefly mentioned in section 4.2, we use the relationship (power = voltage^2 / resistance) along with our user inputs of power and impedance to obtain a maximum voltage we want to have entering our speakers from the output of the amplifier. After this formula is used to find the correct voltage value, we must remember that the voltage information we are comparing this threshold value to is from the result of analog to digital conversions of the stepped-down feedback signal and being as such will be an integer value between 0 and 1023 due to the 10 bits of resolution available to the ADC hardware. Therefore, to get this within an equivalent range we must multiply this high-level voltage value by a factor of (1023*.08/5=16.4). This factor comes from the step-down circuitry. Since we know that this threshold value will be different for every given set of speaker parameters, we need a way of defining bracket levels, where each "bracket" corresponds to a range of incoming data from the ADC that translates into a certain command for the potentiometer's wiper position and hence a certain amount of attenuation of the input audio signal. Since this attenuation depends on the overloadVoltage value, we define each bracket based upon the division of the statistic (whether it is the peak or the average value) describing the set of incoming ADC values by the overloadVoltage value. For example, 1.0 would mean that either the peak or average value from the set of values that were sampled using the microprocessor's ADC function is exactly equal to the corresponding voltage threshold value that would likely blow the speakers and 25.0 would mean that the peak or average ADC value is twenty-five times the value of the safety threshold for the speaker (the point at which our device essentially begins to mute the incoming audio signal).

Next, actual levels must be defined to both keep our signal at safe levels and to avoid any abrupt changes in attenuation, so that the sound coming out of the speaker retains as much of its original volume continuity as possible. The resistance positions of the potentiometer's wiper change by two decibels per discrete position of the wiper, varying from 0 dB with the wiper connected to the high terminal of the potentiometer all the way down to -62 dB where the wiper has very little resistance between it and the low terminal of the potentiometer. The resistor also features a mute function that guarantees a resulting gain of less than -90 dB. For our purposes, however, we must take into account that the stated attenuation of each wiper position is for use of the digital potentiometer alone in developing that amount of attenuation. In our case, the digital potentiometer is merely one component, albeit the most important one, in our analog attenuation circuitry (see chapter 5 for details). This translates into the fact that not only are the actual changes in attenuation we get for each wiper step less than -2 dB, but also that even at the potentiometer's 0 dB level, we will still have a series of resistors present that will function as attenuators to the incoming signal and thus the actual attenuation we experience will be -1.22 dB (see table 2 for discrepancies between wiper positions and actual circuit gain).

Once we had defined what gain levels were achievable by our circuit, we decided that a Microsoft Excel spreadsheet would be most useful in developing a list of the range of input values and then determining the best level of attenuation amongst those values. Since we are basing all of our calculations around the overloadVoltage safety threshold value, our list of inputs is given in terms of multiples of this value, from 0 meaning an input voltage that is zero times the overloadVoltage value and 25 meaning an input voltage that is twenty-five times the overloadVoltage value.

We must make a side note here that since the speaker thresholds in general should be located a reasonable amount higher than the levels of normal operation, a signal level that is twenty-five times higher than that threshold will likely not ever be encountered in actual operation. However, since the range of the potentiometer and associated gain function is capable of attenuating even a signal of this abnormally large magnitude back down to within safe levels, we decided we would program the device to be able to handle such levels. This way, we can be sure to account for the case of a speaker with a very low power rating and resistance that could easily be overblown and could potentially, though not likely, experience levels that are in fact twenty-five times that of its safety threshold. It should also be noted that in fact the attenuation our device offers could potentially limit input levels even above this (25 * overloadVoltage) point. For example, -62 dB is the lowest position offered by the wiper within its normal range of operation and even below that the device features a mute function that guarantees attenuation of -90 dB or below. Converted to a gain in units of Volts/Volts, -62 dB translates into 0.006345 and -90 dB translates into 0.0002542, which would mean that using one of these attenuation levels would theoretically allow for an input that is either 141.8 or 3540.5 times the level of our overloadVoltage safety threshold level while still attenuating this input to attain an output value that is still only 0.90 times that threshold value or less. However, first of all, the -90 dB wiper position is intended for use as a mute function and if we were to specify that our signal could function up until such extremely high input levels under the -90 dB setting we would no longer have a way of muting the input signal if the level was ever exceeded or for other emergency purposes. More importantly, as previously stated, a level of (25*overloadVoltage) should be high enough that, regardless of the speaker's parameters, the input would never reach levels high enough to even come close to this limit; thus, we would expect the mute function of our device never to have to be employed.

Our final attenuation brackets allow for inputs to be smoothly attenuated until the input reaches a level twenty times the threshold value. Between twenty and twenty-five times this threshold value we start attenuating the signal toward a zero output in order to alert the user that they are operating at extremely high levels and the amplifier's gain either needs to be turned down significantly or if the gain continues to increase our device will soon merely mute the input signal (see figure 15 for this trend). Finally, we set the attenuation ranges or input signals in the normal operating range of 0 to (20 * overloadVoltage). Our first decision we make is to keep the output at least 10% below the overloadVoltage safety threshold level. This will leave a small margin of error in our attenuation to first account for the slight time delay between the time the audio signal is input into the device until our device has calculated the necessary attenuation and has

dB	dB in V/V	Potentiometer resistance (in Ohms)	Overall Attenuation Circuit gain (V/V)	Overall Attenuation Circuit gain (dB)	SPI Command sent to Potentiometer
-90	3.16228E-05	0.313065488	2.54174E-04	-71.89736676	10011111
-62	7.94328E-04	7.863849524	0.006344658	-43.95183576	00011111
-60	0.001	9.9	0.007974009	-41.96646563	00011110
-58	0.001258925	12.46336158	0.010017459	-39.98484857	00011101
-56	0.001584893	15.69044261	0.012577757	-38.00793614	00011100
-54	0.001995262	19.75309692	0.015781718	-36.03691468	00011011
-52	0.002511886	24.86767567	0.019785044	-34.07325957	00011010
-50	0.003162278	31.30654884	0.024777644	-32.11879988	00011001
-48	0.003981072	39.41260988	0.030989194	-30.1757943	00011000
-46	0.005011872	49.61753613	0.038694486	-28.24701832	00010111
-44	0.006309573	62.4647771	0.048217748	-26.3358615	00010110
-42	0.007943282	78.63849524	0.059934712	-24.4464316	00010101
-40	0.01	99	0.074270619	-22.58365911	00010100
-38	0.012589254	124.6336158	0.091691784	-20.75339154	00010011
-36	0.015848932	156.9044261	0.112687809	-18.96246128	00010010
-34	0.019952623	197.5309692	0.137741483	-17.21870491	00010001
-32	0.025118864	248.6767567	0.167284115	-15.53090594	00010000
-30	0.031622777	313.0654884	0.201636184	-13.9086306	00001111
-28	0.039810717	394.1260988	0.240937014	-12.36192953	00001110
-26	0.050118723	496.1753613	0.285072579	-10.9008911	00001101
-24	0.063095734	624.647771	0.333616218	-9.535056912	00001100
-22	0.079432823	786.3849524	0.385800493	-8.272744426	00001011
-20	0.1	990	0.440536686	-7.12035839	00001010
-18	0.125892541	1246.336158	0.496489492	-6.081798781	00001001
-16	0.158489319	1569.044261	0.552199867	-5.158074045	00001000
-14	0.199526231	1975.309692	0.606233744	-4.347197873	00000111
-12	0.251188643	2486.767567	0.657325434	-3.64439126	00000110
-10	0.316227766	3130.654884	0.70448641	-3.042547604	00000101
-8	0.398107171	3941.260988	0.747061825	-2.532869108	00000100
-6	0.501187234	4961.753613	0.784732911	-2.105562662	00000011
-4	0.630957344	6246.47771	0.817476546	-1.750493971	00000010
-2	0.794328235	7863.849524	0.845499799	-1.45772983	00000001
0	1	9900	0.869166975	-1.217935669	00000000

Table 2: Potentiometer settings and attenuation statistics

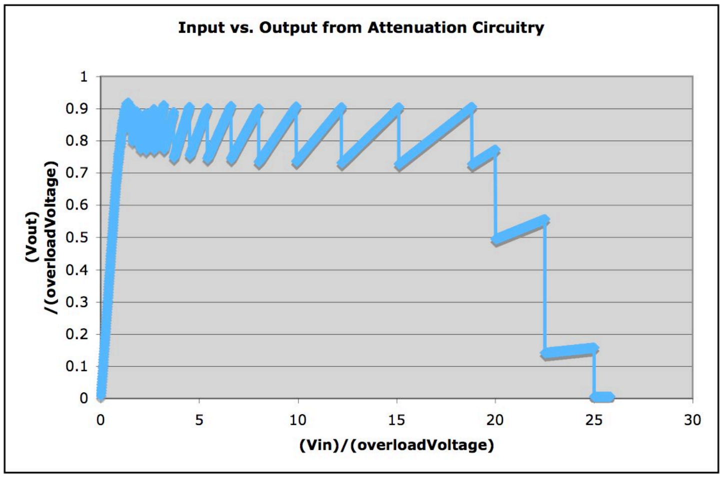

Figure 15: General input vs. output trend of attenuation circuit throughout the range of operation

implemented that attenuation. For instance, if we detect a significant jump in the signal level, if that signal was in fact a crescendoing signal (increasing steadily in volume), by the time our device begins to attenuate that signal the signal will be even higher than we anticipated. Under this 10% safety net system, that signal would have room to still be higher than we expected and be attenuated to a safe level. The other possible discrepancy this allows us to compensate for is the existence of peaks in the system that are either undetected by our ADC sampling or are overlooked by our calculation of the ADC statistic we decide to base our attenuation on. The possibility of our system not detecting a peak in the system is very unlikely, since based on a 8 MHz internal clock rate of the Atmega 32 microprocessor, our use of a division of 64 in the initialization of our ADC conversion (see avr32_adc.h file in our code in the Appendix), and the 8 clock cycles that is typically required to carry out each analog to digital conversion, that gives us a sample rate of about 15.625 kHz. If we consider that the human range of hearing can only detect tones up to 20 kHz and that 4.186 kHz is the highest frequency of a standard piano, we can state with a reasonable amount of confidence that our system should be able to detect nearly all the spikes within the human range of hearing. We should also near in mind that when we talk about "spikes" we are implying that these high amplitude signals are very short-lived and will likely not cause any damage to the speakers, for if they are of a longer and more threatening duration, our system should be fully capable of detecting them and setting the attenuation accordingly. As for a discrepancy between the statistic values we choose from our ADC sampling sets, as briefly mentioned in section 4.2 (and as shown in our code in the ADCAveragingandLCDFunctionsBethany.h header file) under most circumstances we base all of our attenuation off of the incoming voltage's peak value from the set of ADC samples we have taken. However, we must take into account that sometimes we may measure a very short-lived and therefore essentially harmless spike to our system. In that case we have decided that it would be more appropriate to use the average value of the incoming ADC sample data in order to get a better characterization of what the incoming signal is like without consideration of the spike. Therefore, our selection criteria becomes: 1) calculate both the peak and the average of the incoming voltage signal 2) use the peak as the determining ADC statistic to characterize that data unless that peak is greater than 3.5 times the average value 3) if the peak is in fact greater than 3.5 times the average value, use the average value as the statistic instead.

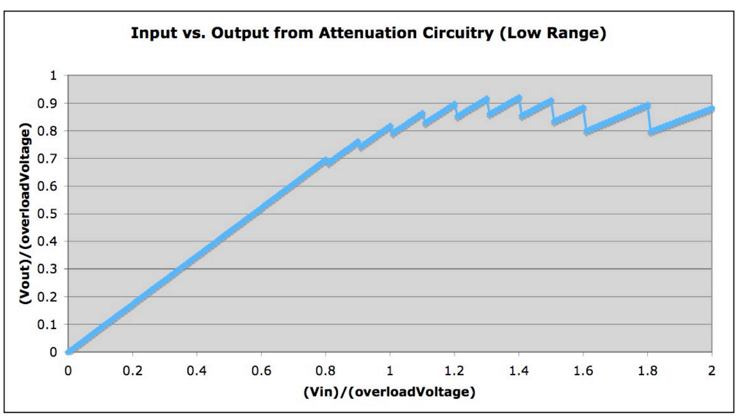

Figure 16: Input vs. output trend of attenuation circuit, low range only

Now that we have justified our use of imposing a (0.9*overloadVoltage) limit on our output signals, we then are left with simply calculating the definitions between where one attenuation level (potentiometer wiper position) stops and another starts based on when the output signal approaches that limit while using the first attenuation bracket. Such transitions can be seen clearly in Figure 16.

As the signal approaches the (0.9*overloadVoltage) output line, the signal shows a sharp drop, signifying a change in the potentiometer's wiper position. Note, however, that even though these drops appear to be abrupt and of a significant size, from figure 17 it can be seen that these drops between attenuation levels are well less than 1 dB in magnitude for reasonable levels of signal input magnitude. This graph does alert us to the fact that as the input levels, and thus attenuation levels, increase this drop becomes larger and thus more in danger of being heard. As mentioned earlier, the just noticeable difference (JND) of a change in volume that a human ear can detect is 1 dB (this was how the system of decibels was created: to define a unit that corresponded to the capacities of human hearing), so any change in signal attenuation levels would not be perceived until the input voltage reached the value corresponding to (1.8*overloadVoltage). Under normal operation, though there may be a few peaks in this region and slightly beyond, a majority of the time the signal will likely stay below this level and thus the wiper level changes should not lead to an abrupt drop in signal level. The other consideration was determining at what level we should begin our attenuation in order to produce a smooth transition up towards the safety limit. The trade-off here was that if we started the attenuation too early, then we would unnecessarily shrink the dynamic range of the audio signal with excessive attenuation even before it approached dangerous levels, but if we waited too long before attenuating, we would have an abrupt drop in signal level once it reached its overloadVoltage limit that would likely be distracting to the listener. We sought a compromise between these two extremes and decided that attenuation beginning at (0.8*overloadVoltage) would be adequate for our purposes and the smoothness of our attenuation and the signal's dynamic integrity can be seen best from figure 16.

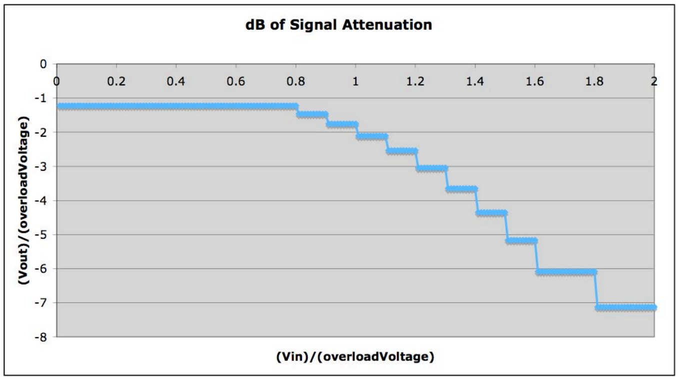

Figure 17: Circuit Attenuation in dB for a Low Range of Input Signals

Once we had our signal attenuation levels defined, there was one more issue to consider in order to smooth out the general purpose operation of our device. Once we had assigned a specific potentiometer wiper position command to each appropriate range of inputs, all we had was code that essentially said: "Given this level of input, move potentiometer wiper to this position." What we failed to take into account, however, was the possibility that the wiper would sometimes need to jump over a significant number of attenuation positions to adjust for a changing input value. For example, if a signal had been at a level corresponding to (0.7*overloadVoltage), which would put it in attenuation bracket number 0 (see digpotControlFinalBethany.h header file in code in the Appendix for attenuation bracket numerical definitions) and lead to a positioning of the wiper at its 0 dB position, but then abruptly jumped up to a level of (1.35*overloadVoltage), corresponding to a wiper in the -12 dB position, under the current "direct command" method we would command the wiper to make the leap between these two positions instantaneously. This abrupt and dramatic change in wiper levels would cause an abrupt and unnatural drop in sound presented to the listener that would likely be very distracting and would take away from the aesthetic sound of the music. In order to make up for this, we altered our code to step through each wiper position of the potentiometer between the previous wiper position and the position the wiper is currently being commanded to go to, pausing for a very short amount of time at each position to allow the gradual smoothing of the attenuation have audible results. Originally, this stepping code only allowed the wiper to step through 1-3 additional positions, depending on how far away the previous wiper position was in relation to the position being currently commanded. However, this was determined to be both inefficient in both the bulkiness of the code and in its ability to produce a genuinely smooth transition between positions, since stepping through one extra position is no where near as smooth as a transition as is stepping through all possible wiper positions between the origin and destination positions. Our final code to implement this feature was an implementation of two if loops, each containing a for loop to cycle through all the potentiometer positions (see the calculateAttenuationBracket function within the digpotControlFinalBethany.h header file in code in the Appendix for full implementation details). The introduction of this feature into our code was crucial to maintaining the aesthetic integrity of the sound we are processing.

> NOTE: In our final code, some unexpected practical considerations had to be accounted for based on the results of some of the final phases of the testing our device, causing some major changes to the overall nature of our system's functionality. See section 4.4 for updates to the attenuation algorithms and the function of the aforementioned attenuation bracket calculations.

CHAPTER 7: PROJECT LOGISTICS

7.1 User Manual

Operation of our device is simple and nearly self-explanatory. The left most jack receives the power amplifier's high voltage signal. This signal is captured using the red cable that came with our device. The cable splits the output of the power amplifier. One end goes to the speaker, and the other goes to this left most high signal jack. The middle jack receives the small signal output of the your instrument. The right most jack is the output into your power amplifier. Once the device has been properly connected, the user first powers on their amplifier and speaker with the amplifier gain knob at a modestly low setting (since the user has not entered the correct parameters for their speaker at this point and proper attenuation to ensure speaker safety is therefore not yet operational) and power on our device's power supply by flipping the black power switch at the rear of the device. Next, the user dials in the speaker impedance and power rating as given by the speaker manufacturer's specifications. The impedance is to be dialed in on the knob on the left and has settings of 2, 4, or 8 Ohms; the values increase with a clockwise turn of the knob and the current value set can be seen on the LCD screen display as it is changed. The power rating is to be dialed in using the knob on the right and has settings of 50 Watts to 800 Watts with the current power rating also being displayed on the LCD screen. Once these two values have been dialed into the system, the amplifier's gain can be changed by the user at will and our system will protect the speaker from being damaged. Also displayed on the LCD screen is the "overloadVoltage" value of the safety threshold corresponding to the user specified speaker ratings and the amount of attenuation currently affect on the device. Once the use of the amplifier and speaker is no longer needed, cut off the power to the speaker and amplifier first before cutting off power to our device (by flipping the black power switch again at the rear of the device) to ensure that speaker safety is maintained for the entire duration of speaker operation.

7.2 Separation of Work

Bethany	**Paul**
Processor breakout	Processor breakout
Atmel coding design	Design of PCB
uP code testing	Hardware testing
UI coding and testing	Power sensing testing
Final board population	Final board population
Test/debug	Test/debug

Table 3: Division of Labor

In general, Bethany was in charge of the software of our device and Paul was in charge of the hardware of our device. In addition to designing the attenuation scheme we intended to use in our limiting algorithms, writing the software code, debugging the code, and testing the code's functionality with various hardware components at different stages during the design process, Bethany was also the main contributor towards composing and doing research for the final report and she designed and constructed the poster for the final demonstration in the NEB rotunda. In addition to deciding what exact functionality was needed in hardware in order to implement our design, designing both the attenuation and step-down circuits for our system, designing and populating the PCBs, and debugging the hardware during both the breadboard and PCB stages, Paul also was responsible for the "finishing touch" work on our hardware in order to make it look professional for our last presentation in the NEB rotunda (including buying and soldering custom plugs for our device's external wirings, adding LED lights to show audio output, changing out the LCD screen, cleaning up any wire-wrappings, etc.).

7.3 Bill of Materials

Product	**Quantity**	**Price Per Unit**	**Total**
Aluminum Face	1	$12.45	$12.45
1/4' Mono Input Jacks	3	$4.34	$13.02
Red LEDs	21	$0.95	$19.95
LCD screen 4x20 Winstar	1	$18.97	$18.97
Power Supply +12/-12/+5	1	$0.00	$0.00
19' Rackmount Casing	1	$0.00	$0.00
Electrolytic Capacitors	20	$0.20	$4.00
Operational Amplifiers TLC2272	5	$0.13	$0.65
THAT electronics	12	$1.56	$18.72
PCB standoffs	3	$4.56	$13.68
Atmel Atmega32	1	$0.00	$0.00
Toggles	2	$0.00	$0.00
Digital Potentiometer MAX5411	3	$0.00	$0.00
		Total:	$101.44

Table 4: Bill of Materials

7.4 Gantt Chart

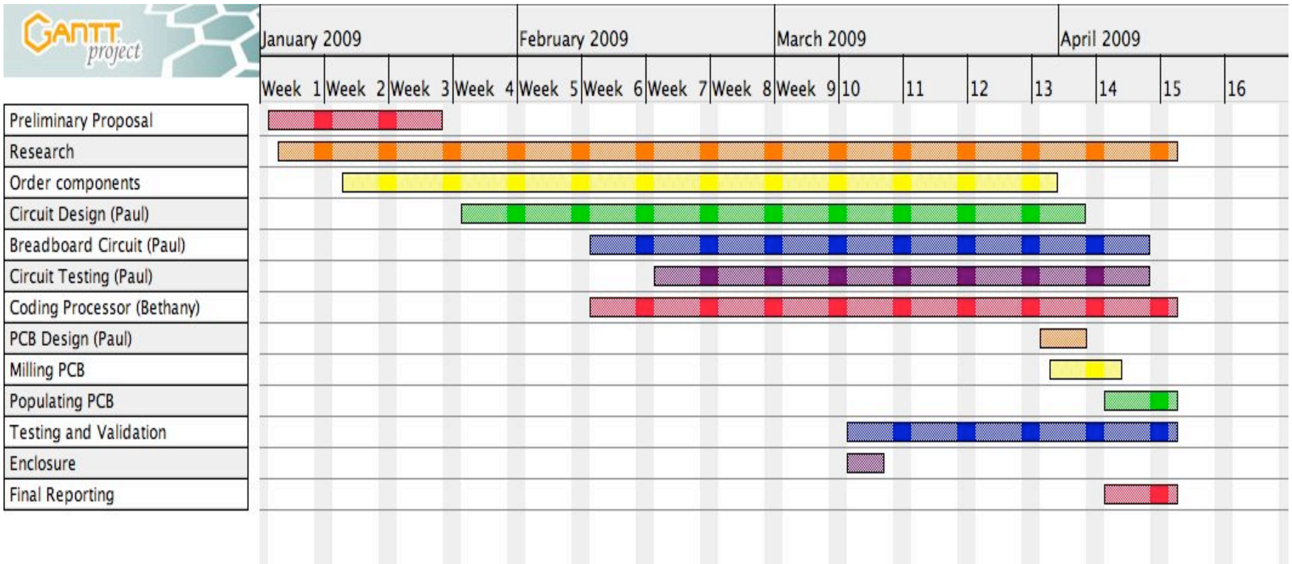

Table 5: Gantt Chart

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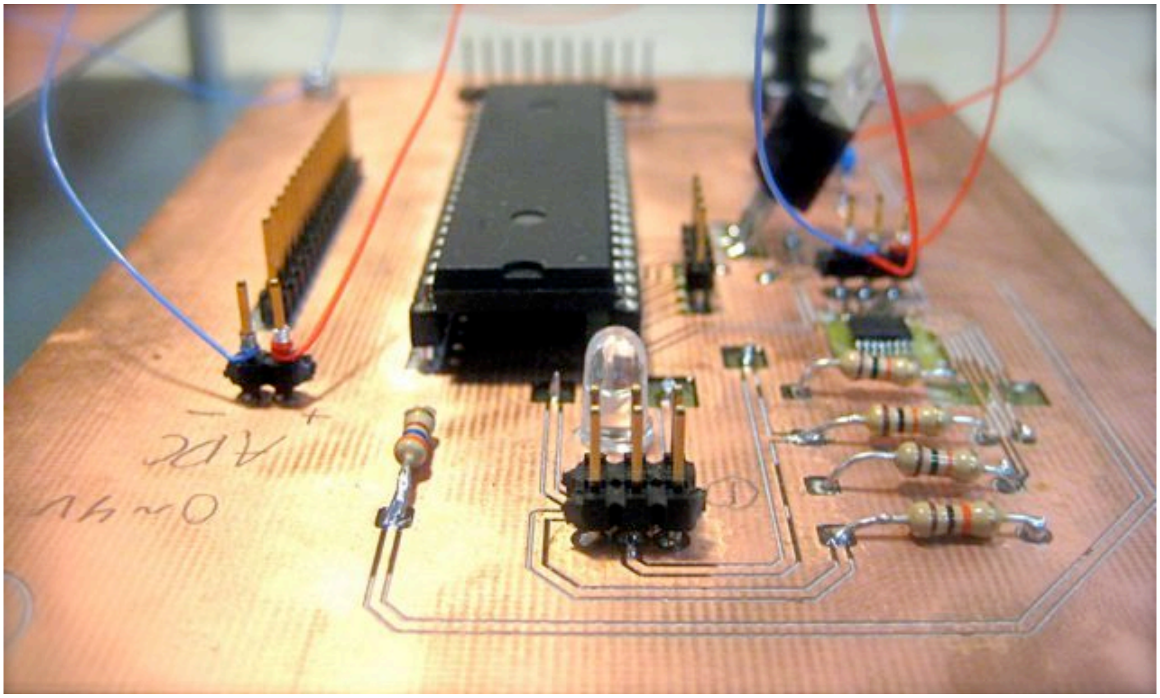
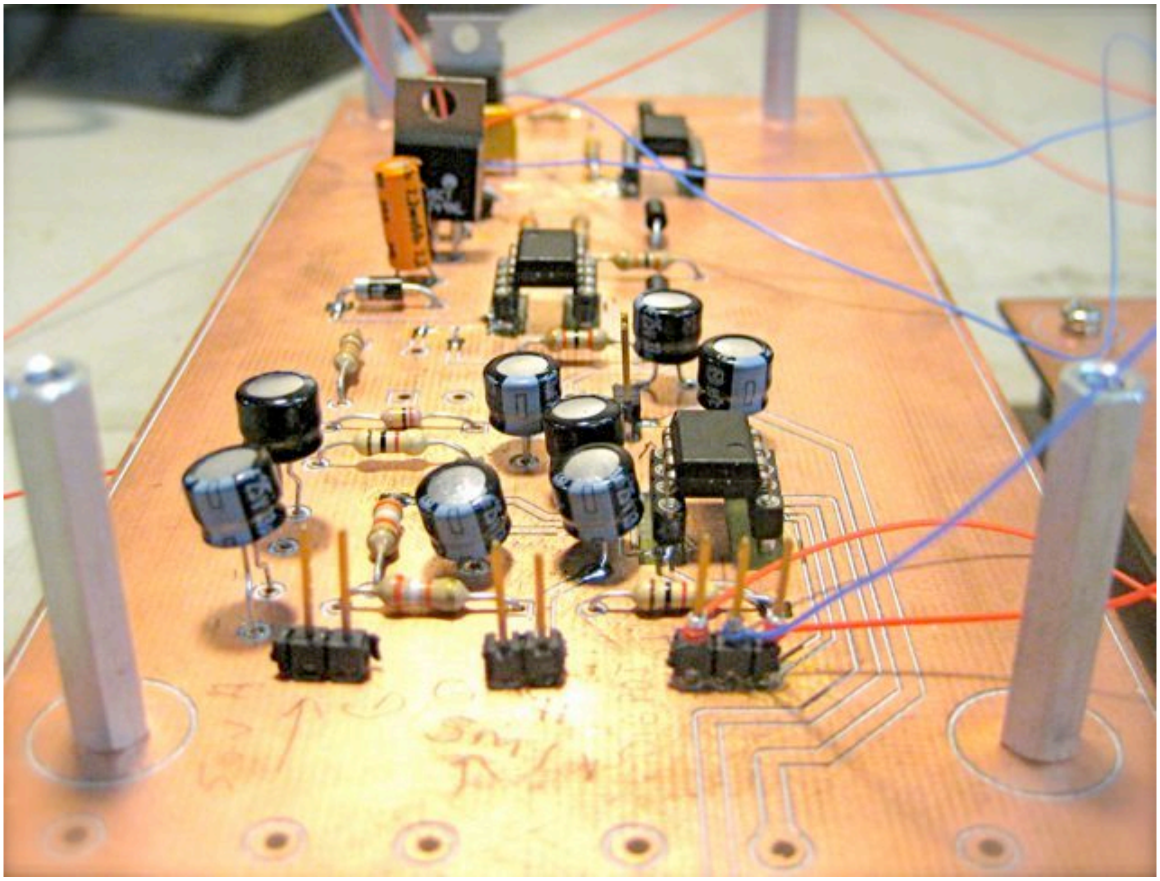
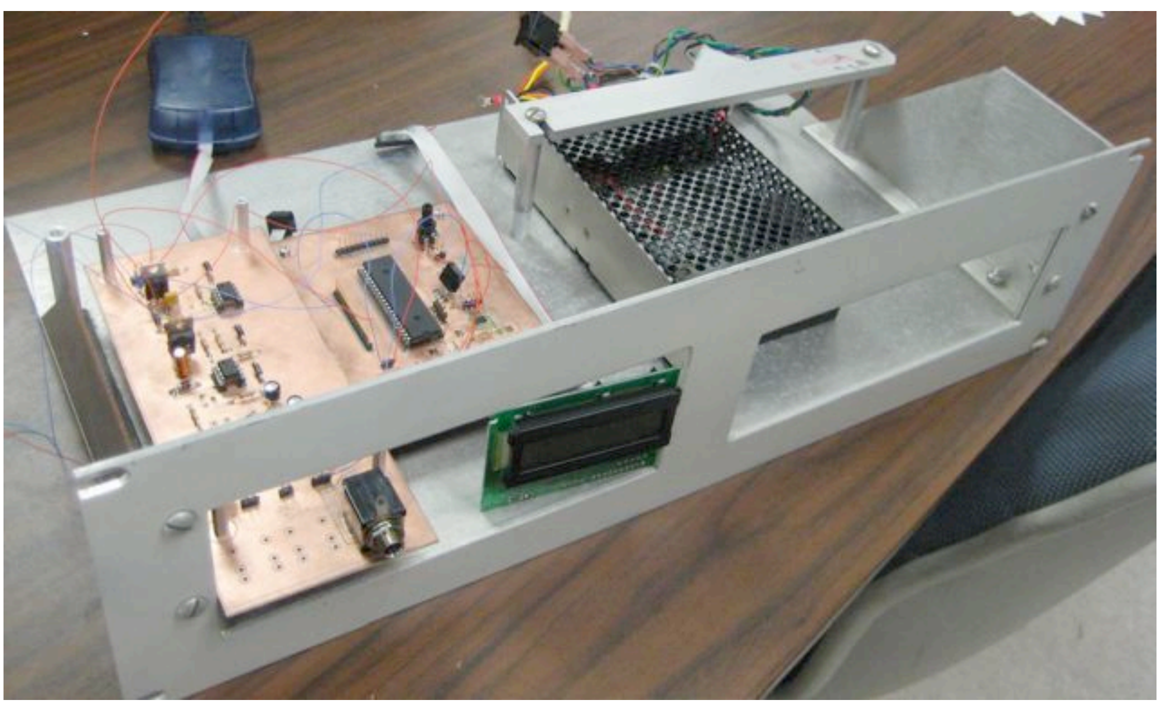
APPENDIX
ADDITIONAL PROJECT MATERIAL







```
//******************************************************************
// File Name   : 'Bethany4-10.c'
//******************************************************************
#include <avr/io.h>
//#include <avr/interrupt.h>
#include "util/delay.h"
//#include "spi.h"
#include <math.h>
#include "adcAveragingAndLCDfunctionsBethany.h"
#include "digpotControlFinalBethany.h"
#include <stdio.h>

#define SPI_PORT PORTB
#define SPI_DDR DDRB
#define DD_MISO 6
#define DD_MOSI 5
#define DD_SCK 7
#define DD_SS  4
#define POT_CS 1
#define LED 0



int main(void)
{
DDRC = 0x00;
SPI_PORT |= 0x01;
delay_ms(10);

int speakerImpedence;
int speakerPowerRating;
int overloadVoltage;
int previousBracketNumber = 0;                      //initially no attenuation
unsigned int adcStatistic;

LCD_INIT();                                         //initialize the LCD screen
SPI_MasterInit();                                   //initialize SPI communication
SPI_MasterTransmit(0xFF);                           //analog power up
SPI_MasterTransmit(0b10111111);                     //enable zero-crossing detection

while(1)
{
//take in user impedence
if( (PORTC | (0b11110000)) == (0b11110001)) speakerImpedence = 2;
if( (PORTC | (0b11110000)) == (0b11110010)) speakerImpedence = 4;
if( (PORTC | (0b11110000)) == (0b11110100)) speakerImpedence = 8;

//take in user’s speaker power rating
if( (PORTC | (0b00001111)) == (0b00001111)) speakerPowerRating = 100;
if( (PORTC | (0b00001111)) == (0b00011111)) speakerPowerRating = 200;
if( (PORTC | (0b00001111)) == (0b00101111)) speakerPowerRating = 300;
if( (PORTC | (0b00001111)) == (0b00111111)) speakerPowerRating = 400;
if( (PORTC | (0b00001111)) == (0b01001111)) speakerPowerRating = 500;
if( (PORTC | (0b00001111)) == (0b01011111)) speakerPowerRating = 600;
if( (PORTC | (0b00001111)) == (0b01101111)) speakerPowerRating = 700;
if( (PORTC | (0b00001111)) == (0b01111111)) speakerPowerRating = 800;
if( (PORTC | (0b00001111)) == (0b10001111)) speakerPowerRating = 900;
if( (PORTC | (0b00001111)) == (0b10011111)) speakerPowerRating = 1000;

overloadVoltage = (sqrt(speakerImpedence*speakerPowerRating)*0.08*1023)/5;
                                     //calculate the overloadVoltage
                                     //safety threshold
adcStatistic = adcPeakOrAve();
previousBracketNumber = calculateAttenuationBracket(adcStatistic, overloadVoltage,
       previousBracketNumber);
       //The above code goes through one cycle of adc converting a set of
       //100 values, returning either its peak or its average, then using that
```

```
        //info along with the overloadVoltage and the old wiper position the command
        //1) determines the destination attenuation bracket the wiper needs
        //to reach while defining each level as a ratio comparison of the adc reading
        //to the overloadVoltage 2) determines how far away from that destination it
        //is currently 3) goes through gradual steps to set the wiper to its new position
        //4) returns the new bracket number of the wiper's position so that it can be set
        //as the new previousBracketNumber in the main function

displayADCPeakandAttenuation(adcStatistic, overloadVoltage, previousBracketNumber);
        //update the display of the LCD screen to show the peak from the current set of
        //ADC data (a value between 0 and 1023), the overloadVoltage value calculated
        //from the user's input data (on the 0 to 1023 range), and the attenuation level
        //of the digital potentiometer (from 0 dB and -90 dB)
}
return 0;
}
```

```
//*****************************************************************
// File Name   : 'adcAveragingAndLCDfunctionsBethany.h'
//*****************************************************************
#include <avr/io.h>
#include <stdio.h>
#include <stdlib.h>
#include <string.h>
#include "LCD.h"
#include "avr32_adc.h"


unsigned int adcPeakOrAve()
{
ADC_INIT();                                                 // Initialize ADC

unsigned int adc_val[100];
unsigned int adcSum=0;
unsigned int adcPeak =0;
        for(int i = 0 ; i < 100 ; i++)
        {
        adc_val[i] = ADC_START(0);                  //set current adc value at channel 4
                                                    //to value in int
        adcSum = adcSum + adc_val[i];               //keep tally of adcSum for this set
                                                    //of 100 adc conversions
                if(adc_val[i] > adcPeak)
                {
                adcPeak = adc_val[i];               //if new high value, set that as peak
                }
                _delay_ms(10);
                                                    //this is the wait time between
                                                    //samples (plus the 8
                                                    //clock cycles or so it takes between
                                                    //each ad conversion as occurs
                                                    //in the ADC_START function

        }
return adcPeak;
}




void displayADCPeakandAttenuation(int adcPeak, int overloadVoltage, int
        previousBracketNumber)
{
        char adcPeak_string[20];
        char previousBracketNumber_string[20];
        char overloadVoltage_string[20];

        sprintf(adcPeak_string,"%d",adcPeak); // Convert ADC int value to ASCII string
```

```
        sprintf(previousBracketNumber_string,"%d", (previousBracketNumber*2));
                        // Convert ADC binary value to ASCII string
        sprintf(overloadVoltage_string,"%d", (overloadVoltage));
                        // Convert overloadVoltage binary value to ASCII string
        LCD_COMMAND(LCD_CLEAR_HOME);
                        // Clear the LCD and send cursor to the beginning
                LCD_STRING("ADC: ");                    // Print text to LCD
                LCD_STRING(adcPeak_string);             // Print ASCII adc value
                LCD_STRING(" OV: ");                    // Print text to LCD
                LCD_STRING(overloadVoltage_string);     // Print ASCII overloadVoltage value
                LCD_ADDR(0x40);                         // Move cursor to second line
                LCD_STRING("Attenuation: -");
                LCD_STRING(previousBracketNumber_string);
                LCD_STRING("dB");
                _delay_ms(30);                  // Pause for 100ms before looping again
}
```

```
//*********************************************************************
// File Name   : 'avr32_adc.h'
//*********************************************************************

void ADC_INIT(void)
{
        ADMUX = 0b00000000;
        ADCSRA = 0b10000110;            // ADC clock prescalar of 64
}

int ADC_START(unsigned char channel)
{
        ADMUX = channel;
        ADCSRA |= (1 << ADSC);          // Start conversion
        while(ADCSRA & (1 << ADSC));    // wait for conversion to complete
        return ADC;
}
```

```
//*********************************************************************
// File Name   : 'digpotcontrolfinalbethany.h'
//*********************************************************************

#include <avr/io.h>
#include "util/delay.h"
#include "math.h"
#define SPI_PORT PORTB
#define SPI_DDR DDRB
#define DD_MISO 6
#define DD_MOSI 5
#define DD_SCK 7
#define DD_SS 4
#define POT_CS 1
#define LED 0
#define Bitset(port,pin) port|=(1<<pin)
#define Bitclr(port,pin) port&=~(1<<pin)

void SPI_MasterInit(void)
```

```
	{
	/* Set MOSI and SCK output, all others input */
	SPI_DDR = (1<<DD_MOSI)|(1<<DD_SCK)|(1<<DD_SS)|(1<<POT_CS)|(1<<LED);
	/* Enable SPI, Master, set clock rate fck/16 */
	SPCR=(1<<SPE)|(1<<MSTR)|(1<<SPR0);
	Bitset(SPI_PORT, 4);
	}

//____________________________________________________________________________________


void SPI_MasterTransmit(unsigned char cData)
	{
	/* Set SS low */
	Bitclr(SPI_PORT,1);
	/* Start transmission */
	SPDR = cData;
	/* Wait for transmission complete */
	_delay_ms(100);
	/* Set SS high */
	Bitset(SPI_PORT,1);
	_delay_ms(10);
	}



//_____________________________________________________________________________________



int calculateAttenuationBracket(int currentVoltage, int overloadVoltage, int
previousBracketNumber)
{
int currentBracketNumber;
//setting up the attenuation bracket definitions
	if(currentVoltage < (0.9*overloadVoltage))
		{
		currentBracketNumber = 0;
		}
	else if(currentVoltage < (1.0*overloadVoltage))
		{
		currentBracketNumber = 1;
		}
	else if(currentVoltage < (1.1*overloadVoltage))
		{
		currentBracketNumber = 2;
		}
	else if(currentVoltage < (1.2*overloadVoltage))
		{
		currentBracketNumber = 3;
		}
	else if(currentVoltage < (1.3*overloadVoltage))
		{
		currentBracketNumber = 4;
		}
	else if(currentVoltage < (1.4*overloadVoltage))
		{
		currentBracketNumber = 5;
		}
	else if(currentVoltage < (1.5*overloadVoltage))
		{
		currentBracketNumber = 6;
		}
	else if(currentVoltage < (1.6*overloadVoltage))
		{
		currentBracketNumber = 7;
		}
	else if(currentVoltage < (1.7*overloadVoltage))
		{
		currentBracketNumber = 8;
```

```
        }
else if(currentVoltage < (1.8*overloadVoltage))
        {
        currentBracketNumber = 9;
        }
else if(currentVoltage < (2*overloadVoltage))
        {
        currentBracketNumber = 10;
        }
else if(currentVoltage < (2.3*overloadVoltage))
        {
        currentBracketNumber = 11;
        }
else if(currentVoltage < (2.7*overloadVoltage))
        {
        currentBracketNumber = 12;
        }
else if(currentVoltage < (3.2*overloadVoltage))
        {
        currentBracketNumber = 13;
        }
else if(currentVoltage < (3.7*overloadVoltage))
        {
        currentBracketNumber = 14;
        }
else if(currentVoltage < (4.5*overloadVoltage))
        {
        currentBracketNumber = 15;
        }
else if(currentVoltage < (5.4*overloadVoltage))
        {
        currentBracketNumber = 16;
        }
else if(currentVoltage < (6.6*overloadVoltage))
        {
        currentBracketNumber = 17;
        }
else if(currentVoltage < (8*overloadVoltage))
        {
        currentBracketNumber = 18;
        }
else if(currentVoltage < (9.9*overloadVoltage))
        {
        currentBracketNumber = 19;
        }
else if(currentVoltage < (12.2*overloadVoltage))
        {
        currentBracketNumber = 20;
        }
else if(currentVoltage < (15.1*overloadVoltage))
        {
        currentBracketNumber = 21;
        }
else if(currentVoltage < (18.8*overloadVoltage))
        {
        currentBracketNumber = 22;
        }
else if(currentVoltage < (20*overloadVoltage))
        {
        currentBracketNumber = 23;
        }
else if(currentVoltage < (22.5*overloadVoltage))
        {
        currentBracketNumber = 24;
        }
else if(currentVoltage < (25*overloadVoltage))
        {
        currentBracketNumber = 25;
        }
else if(currentVoltage >= (25*overloadVoltage))
        {
```

```
            currentBracketNumber = 26;
            }




        if(previousBracketNumber == currentBracketNumber)
        {
        SPI_MasterTransmit(potCommandValue(currentBracketNumber));
        }



        else if(previousBracketNumber > currentBracketNumber)//too soft
        {
                SPI_MasterTransmit(potCommandValue(previousBracketNumber-1));
                currentBracketNumber = previousBracketNumber-1;
        }



        else if(previousBracketNumber < currentBracketNumber)//too loud
        {

                SPI_MasterTransmit(potCommandValue(previousBracketNumber+1));
                currentBracketNumber = previousBracketNumber+1;
        }





displayADCPeakandAttenuation(currentVoltage, overloadVoltage, currentBracketNumber);

        return currentBracketNumber;

        }







//___________________________________________________________________________________________





int bracketNumber(int PotCommand)
{
        if(PotCommand == 0b00000000) return 0;
        else if(PotCommand == 0b00000001) return 1;
        else if(PotCommand == 0b00000010) return 2;
        else if(PotCommand == 0b00000011) return 3;
        else if(PotCommand == 0b00000100) return 4;
        else if(PotCommand == 0b00000101) return 5;
        else if(PotCommand == 0b00000110) return 6;
        else if(PotCommand == 0b00000111) return 7;
        else if(PotCommand == 0b00001000) return 8;
        else if(PotCommand == 0b00001001) return 9;
        else if(PotCommand == 0b00001010) return 10;
        else if(PotCommand == 0b00001011) return 11;
        else if(PotCommand == 0b00001100) return 12;
        else if(PotCommand == 0b00001101) return 13;
        else if(PotCommand == 0b00001110) return 14;
        else if(PotCommand == 0b00001111) return 15;
        else if(PotCommand == 0b00010000) return 16;
```

```
        else if(PotCommand == 0b00010001) return 17;
        else if(PotCommand == 0b00010010) return 18;
        else if(PotCommand == 0b00010011) return 19;
        else if(PotCommand == 0b00010100) return 20;
        else if(PotCommand == 0b00010101) return 21;
        else if(PotCommand == 0b00010110) return 22;
        else if(PotCommand == 0b00010111) return 23;
        else if(PotCommand == 0b00011001) return 24;
        else if(PotCommand == 0b00011111) return 25;
        else if(PotCommand == 0b10011111) return 26;
}




//_______________________________________________________________________________________




int potCommandValue(int bracketNumber)
{
        if(bracketNumber == 0) return 0b00000000;
        else if(bracketNumber == 1) return 0b00000001;
        else if(bracketNumber == 2) return 0b00000010;
        else if(bracketNumber == 3) return 0b00000011;
        else if(bracketNumber == 4) return 0b00000100;
        else if(bracketNumber == 5) return 0b00000101;
        else if(bracketNumber == 6) return 0b00000110;
        else if(bracketNumber == 7) return 0b00000111;
        else if(bracketNumber == 8) return 0b00001000;
        else if(bracketNumber == 9) return 0b00001001;
        else if(bracketNumber == 10) return 0b00001010;
        else if(bracketNumber == 11) return 0b00001011;
        else if(bracketNumber == 12) return 0b00001100;
        else if(bracketNumber == 13) return 0b00001101;
        else if(bracketNumber == 14) return 0b00001110;
        else if(bracketNumber == 15) return 0b00001111;
        else if(bracketNumber == 16) return 0b00010000;
        else if(bracketNumber == 17) return 0b00010001;
        else if(bracketNumber == 18) return 0b00010010;
        else if(bracketNumber == 19) return 0b00010011;
        else if(bracketNumber == 20) return 0b00010100;
        else if(bracketNumber == 21) return 0b00010101;
        else if(bracketNumber == 22) return 0b00010110;
        else if(bracketNumber == 23) return 0b00010111;
        else if(bracketNumber == 24) return 0b00011001;
        else if(bracketNumber == 25) return 0b00011111;
        else if(bracketNumber == 26) return 0b10011111;
}
```