DUAL CHANNEL MATRIX SWITCH AUDIO RECEIVER

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ABSTRACT

The Dual Channel Matrix Switch Audio Receiver controls 2 separate audio output channels. Each channel plays any of the system's 3 inputs. This controller enables a user to play two separate audio signals through two separate speaker channels. The system design allows audio input from 2 RCA sources or 1 RCA source and a phono source. The system outputs an audio signal for each output simultaneously at up to 36W on each channel for an 8Ω load. The device allows a user to control the audio input and the volume of each output channel. An Arduino Uno R3 microcontroller interfaced with front rotary potentiometers and a display creates this capability. The device operates using standard 120V, 60Hz AC power and outputs audio signals ranging from 20Hz to 20kHz.

Acknowledgments

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Chapter I: Introduction

The home audio industry contains a huge variation of home stereo products available to consumers. However, buying a receiver with 2 separate output channels at a reasonable cost presents challenges. If a person desires to use the same receiver to control speakers in 2 separate locations then they are simply out of luck. To do this, 2 separate receivers are required. This costs a large sum of money and simply shouldn't be necessary to accomplish the goal of playing audio in 2 different locations when one device could accomplish this.

The Dual Output Matrix Switch Audio Receiver solves this problem. This device contains two separate, controllable audio output channels, each capable of driving 8Ω speakers at up to 36W. The device routes any of the 3 possible inputs to the two system outputs through a series of MOSFET input switching stages. These MOSFET stages are controlled externally by the user with 3 slide switches. This enables the user to choose any of the 3 sources as the controlling input for each output. The two output channels can play the same source or 2 separate sources simultaneously and each channel has its own volume control. This device also costs much less than most industry receivers. The device cost me $250 to build and this cost could significantly reduce if it was manufactured on a commercial basis.

A 4 channel class D amplifier forms the basis of the Dual Output Matrix Switch Audio Receiver. A class D amplifier gives significant advantages in power dissipation, cost and efficiency over class A, AB and B amplifiers. These advantages come from the circuit topology of a class D amplifier. All audio amplifiers use transistors to drive their outputs. For class A, these output stage transistors stay on at all times. This topology yields superior audio quality but at the cost of great power loss due to the large DC bias current needed to keep the transistors on. Class B transistors eliminate this large DC bias current by operating in a push-pull manner. Although this greatly reduces the power dissipated, class B transistors yield far inferior sound quality. The loss of quality is due to crossover distortion caused by nonlinear behavior which occurs when the output transistors change between their on and off states. Class AB gives a compromise between class A and class B amplifiers. This type of amplifier operates with some DC bias current with a class B-like push-pull output stage. This topology prevents the crossover distortion of class B amplifiers. However, even a well designed class AB amplifier experiences significant power dissipation due to the distance between the large voltages needed to drive the push-pull output stages and the incoming signal's voltage level. Class D output stage transistors switch between the positive and negative power supplies in a way which creates a train of voltage pulses. This kind of topology greatly reduces power dissipation because the output transistors conduct no current when they are not switching and have a very low V_{DS} voltage when switching, thus using less power. Figure 1 below graphically displays the superiority of class D amplifiers in terms of power efficiency for varying load power.

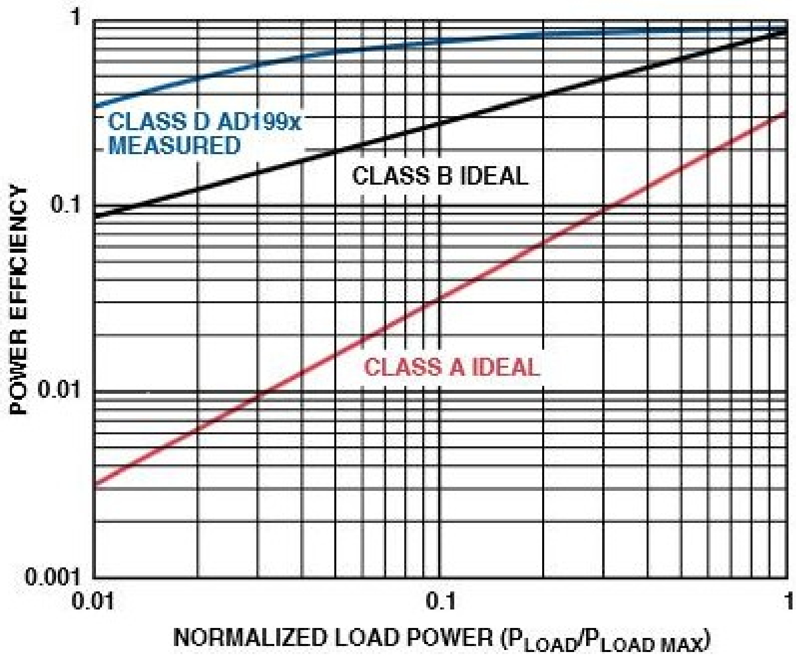

FIGURE 1: POWER EFFICIENCY OF CLASS A, B AND D OUTPUT STAGES [1]

Class D topologies use a modulator to convert the audio input into pulses where the frequency of the pulses contains the audio signal information. The switching output stage amplifies this pulse train through a low pass filter, which works to minimize electromagnetic interference and remove the high frequency components generated in the modulation process. Figure 2 below shows the basic class D topology.

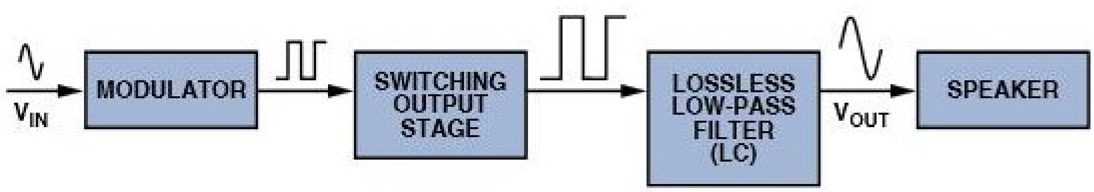

FIGURE 2: CLASS D OPEN-LOOP BLOCK DIAGRAM [1]

Chapter II: Needs, Requirements and Specifications

A. Customer Needs Assessment

The inspiration for this project arose from a problem with my parent's home stereo system. My father, John Fox, has a set of indoor speakers and a set of outdoor speakers playing through the same receiver. He would like to play one audio source on the indoor speakers while a separate audio source plays on the outdoor speakers. Unable to find a product which accomplishes this, my dad inspired me to design a new audio device. This device needs to simultaneously play 2 separate audio sources, at two separate volume levels through 2 different sets of speakers. The device must use 120V, 60Hz AC power to operate in its intended environment. The device must allow John to change the input between 2 RCA sources and a phono source and also to control the volume of each output channel.

B. Requirements and Specifications

The customer's needs directly drove the Dual Channel Matrix Switch Audio Receiver marketing requirements. They ensure proper device operation under normal conditions in a way which satisfies the customer's needs. Abstract, verifiable, unambiguous and traceable engineering specifications meet all marketing requirements efficiently and effectively while still allowing freedom and creativity in the system design and implementation. Table I below contains the marketing requirements and engineering specifications for the Dual Channel Matrix Switch Audio Receiver.

TABLE I

DUAL CHANNEL MATRIX SWITCH AUDIO RECEIVER REQUIREMENTS AND SPECIFICATIONS

Marketing Requirements	Engineering Specifications	Justification
5	1. The system should operate from standard 120V 60Hz AC power	A standard United States home contains this type of power
2, 3	2. The system should output 2 separate, stereo and mono capable audio channels	The customer desires 2 separate audio channels
3,4	3. The system should take in 2 RCA inputs and 1 phono input	This meets the audio input needs of the customer.
1, 2, 3, 4	4. Each output channel must output each system input with no audible signal corruption added	To comply with the customer need for 2 separate audio channels
7	5. The system should have no protruding wiring or components other than those necessary for input and output interfacing	Cannot use an unsafe device
6	6. A user must have control of each output channel's volume.	Volume control represents a fundamental capability of any device which plays audio signals
8	7. The system should visibly display volume and active input for each output channel.	The user must have access to the current state of the system to enable effective control.
9	8. Each amplified system output should drive an 8Ω load at a maximum average power between 25W and 30W for a line input source.	Impedance values derived from standard speaker impedance. Maximum deliverable power limited by 24VDC power supply.
9	9.Each amplified system output should drive an 8Ω load at a maximum average power between 8W and 10W for a headphone input source.	Impedance values derived from standard speaker impedance. A headphone input yields a lower voltage input signal and so high gain application results in less power.
9	10. Each amplified system output should have the capability to amplify a signal at a level of 63dB into an 8Ω load.	This decibel level represents a power gain which would generate 25W of output power from line input taken across a standard 48kΩ impedance.
10	11. The system enclosure should not exceed 16”x12”x8”	Dimensions derived from John's home stereo cabinet and typical receiver dimensions.

Marketing Requirements

1. High sound quality for each input/output pair
2. 2 audio output channels derived from the same set of inputs
3. 2 RCA audio inputs
4. 1 Phono input
5. Powered through a standard wall outlet
6. Volume controllable on each audio output channel
7. System operates safely and does not damage itself or the user under normal operating conditions
8. System displays volume and active input for each output channel
9. Drive at least 2 sets of standard home stereo speakers
10. The device fits in a home stereo cabinet

The requirements and specifications table format derives from [2], Chapter 3

Chapter III: Design

A: Functional Decomposition

I. Level 0

Level 0 consists of the system itself and all of its inputs and outputs. The device takes in 2 RCA inputs, a phono input, standard 120V AC power @ 60Hz and user inputs to produce 2 amplified speaker outputs. The device also displays the current volume and active input for both output channels. The device accepts user input in the form of rotary knobs for volume, switches for input/output control and a switch for power. Figure 3 below shows the level 0 block diagram and Table II shows the functional decomposition of level 0.

FIGURE 3: LEVEL 0 BLOCK DIAGRAM

TABLE II

LEVEL 0 FUNCTIONALITY TABLE

Module	Dual Output Audio Receiver
Inputs	• RCA1: Left and Right channel Audio input 5mV-1.1V_p • RCA2: Left and Right channel Audio input 5mV-1.1V_p • Phono: 5-50mV_p • User Input: Volume Control, Power Control, Channel Switching Control • 120V 60Hz AC Power
Outputs	• Amplified Audio CH1: 24 V_p Max(for 8Ω Load) • Amplified Audio CH2: 24 V_p Max(for 8Ω Load) • Volume Display: dB values displayed as determined by PGA2311 input equation I added to the constant gain of the amplifier board (see Figure 5) • Active Channel Display: LEDs indicate active input for each channel and 1 LED indicates phono or RCA2
Functionality	Takes in either 2 RCA inputs or 1 RCA and 1 Phono input as audio signals. Outputs 2 separate audio signals simultaneously at amplified level determined by volume control input. Powered through a standard 120V 60Hz wall outlet. Allows control of each output's controlling input signal through exterior switches controlled by the user. Displays current states of each output channel. Each powered output drives an 8 Ω load directly attached to the device.

II. Level 1

Level 1 details the internal modules of the device and also demonstrates the signal flows throughout the device. Figure 4 below shows the block diagram for level 1. Input 1 contains only an RCA stereo input and this goes directly into both channel switching stages. Input 2 contains a switching stage before these channel switching stages to allow the use of a phono input. This preliminary switching stage determines if input 2 comes from the phono input or the RCA input. Each switching stage passes 1 of its 2 inputs to the next stage in the signal flow. The signal passed is determined by the state of the external switches on the device. If the phono input is selected, the phono signal passes through a stereo preamp which amplifies the signal to line level. Once the audio passes through the channel switching stage it enters a digital potentiometer stage. An Arduino controls these potentiometers using an SPI interface. The potentiometer attenuates the signal based on the external volume knobs controlled by the user. These volume knobs are denoted as "User Volume Control Input" on Figure 4. The class D amplifier then voltage amplifies the signal at a constant level determined by on-board switches (see Figure 5 below). The board uses this amplified signal to drive speakers at up to 36W. AC power enters 2 modules in the device which each generate a separate set of DC voltages. The Class D board power generator transforms 120VAC power into 24V at a max of 14A to drive the class D power amplifier and this power is toggled externally with a switch. The split supply voltage generator transforms the AC power into 4 different DC voltages: +5V, -5V, +8V and -8V. The other input stage modules in the circuit use these voltages as shown in Figure 4 below. The LCD display directly connects to and is controlled by the Arduino. The LED control circuit is turned on and off by the Arduino depending on the state of the external power toggle switch. When power is on, this circuit activates the LED corresponding to the active input/output settings selected by the user. Table III shows the functional decomposition of level 1 and contains a description of each module shown in the block diagram.

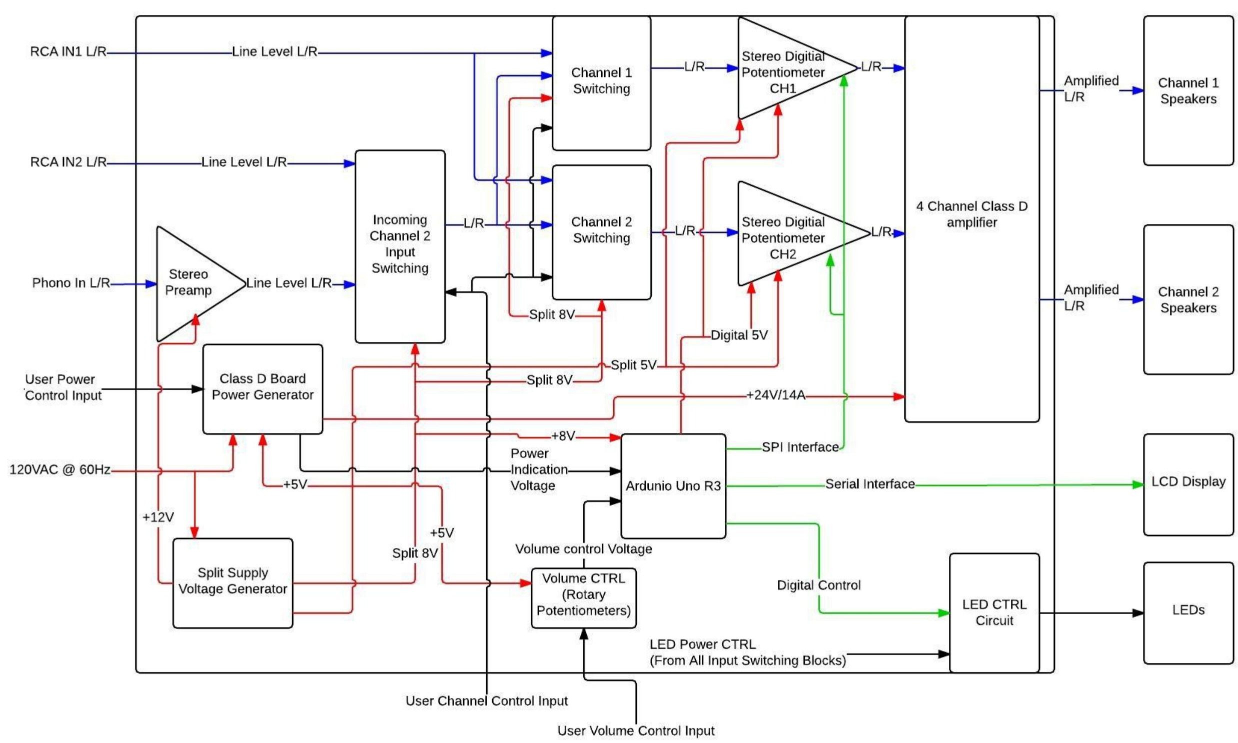

FIGURE 4: LEVEL 1 BLOCK DIAGRAM

TABLE III

LEVEL 1 FUNCTIONALITY TABLES

a)

Module	4 Channel Class D Amplifier
Inputs	• Attenuated Audio: 17uV_p - 2.2V_p • Power: + 24V DC, 14A max
Outputs	• Voltage Amplified Audio Signal: 24 V_p Max (for 8Ω Load)
Functionality	Amplify an incoming voltage signal by up to 37.6dB (typical for 8Ω) to drive up to 4 speakers. The board has an input impedance around 60kΩ and an output impedance of around 0.2Ω.

b)

Module	Audio Switching Stages
Inputs	• Audio Voltage signal 1: 5mV-1.1V_p • Audio Voltage signal 2: 5mV-1.1V_p • User input: position of external switch controlled by user switches internal FET gates between +8 and -8V. • DC Voltage: +8V and -8V for internal ICs and controlling switches.
Outputs	• Audio Voltage signal: 5mV-1.1V_p
Functionality	Switch between the two incoming audio inputs based on user input (external switch). The output signal then passes to the next stage in the audio signal chain. The buffer should have an input resistance > 1kΩ.

c)

Module	Split Supply Power Generator
Inputs	• 120V 60Hz AC Power
Outputs	• DC Voltages: + 8V, -8V, + 5V, -5V
Functionality	Converts AC power to DC voltage values for use with the Audio Switching Stages, Stereo Digital Potentiometers, Switches, Arduino, volume control and Class D board Power Generator.

d)

Module	Class D Board Power Controller
Inputs	• 120V 60Hz AC Power • 5V DC Voltage: from the split supply voltage generator • User Power Control Input: External Toggle Switch
Outputs	• DC Voltage: + 24V (14A max) • Power Indication Voltage: Tells Arduino the state of the external toggle switch
Functionality	Converts AC power to 24V DC to power the class D amplifier board. Toggles this DC voltage on and off with an external switch. Sends a voltage signal to the Arduino indicating the state of the toggle switch.

e)

Module	LCD Display
Inputs	• Serial interface: Programmed by the Arduino to display text. Arduino also supplies power.
Output	• Display of current channel volume: dB values displayed as determined by PGA2311 input equation I added to the constant gain of the amplifier board (see Figure 5)
Functionality	Displays the volume level of each channel, the voltage level of each channel as determined by the external volume control knobs

f)

Module	Pyle PP444 Mini Phono Preamp
Inputs	• Stereo Phono level audio signal: 5-50mV_p • DC Voltage: +12V
Outputs	• Audio Voltage signal: 5mV-1.1V_p
Functionality	Converts a phono level audio signal to a line level audio signal.

g)

Module	Volume Control
Inputs	• User Input: knobs • DC Voltage: +5V
Outputs	• Volume control voltage: analog voltage signal indicating the user desired volume level.
Functionality	Converts user input into an analog voltage signal to be used by the Arduino to control volume.

h)

Module	LED Control Circuit/LEDs
Inputs	• Digital Control: +5V or 0V depending on state of user power toggle switch • LED Power Control: Voltage from audio switching stages. LED active corresponds to the active FET gate in the audio switching stages. Thus they are also controlled by the external user channel control input.
Outputs	• Light from LEDs indicating audio switching stage states.
Functionality	Displays the active input and output states to the user through LEDs.

i)

Module	Stereo Digital Potentiometers (IC: PGA2311P)
Inputs	• Audio Voltage signal: 5mV-1.1V_p • SPI interface: MOSI data line, clock signal and a chip select all coming from the Arduino
Outputs	• Attenuated Audio: 17uV_p - 2.2V_p
Functionality	Takes an audio input and attenuates it based on the Arduino input. This Arduino input comes through an SPI interface. The 2 potentiometers share the same data line and clock signal but are activated using different chip select signals. This SPI interface programs the IC with 16 bits corresponding to a level of attenuation for the left and right channels. The bits are read in on the rising edge and there is 1 bit read per clock cycle after the chip select goes low.

j)

Module	Arduino Uno R3
Inputs	• DC Voltage: +8V. The Arduino internally regulates this voltage to 5V for power. • Volume control voltage: analog voltage signal indicating the user desired volume level. • Power Indication Voltage: Tells Arduino the state of the external toggle switch
Outputs	• SPI interface: MOSI data line, clock signal and 2 separate chip selects for controlling the digital potentiometers • LED (Digital) Control: +5V or 0V depending on state of user power toggle switch • Serial interface: Programs the LCD to display text. It also supplies power to the LCD.
Functionality	Takes in the volume control input from external volume control knobs and converts this into the corresponding byte values to transfer to the digital potentiometers. The SPI interface programs each IC with 16 bits corresponding to a level of attenuation for the left and right channels. The bits are read in on the rising edge and there is 1 bit read per clock cycle after the chip select goes low. Since 2 are programmed each cycle of PGA2311 programming takes 34 clock cycles (16 for each programming and 1 for each chip select change). It also uses the volume control data to display the current level of volume on the LCD screen. Takes in the power indication voltage and uses it to control the LED control circuit with a digital signal.

III. Level 2

The different modules listed above in the level 1 block diagram are further elaborated on in the following section. I designed the schematics for the audio switching stages, the split supply power generator, the volume control and the LED circuitry. All other components were integrated into the system via their peripherals.

a. 4 Channel Class D Amplifier

The high power audio reproduction capabilities of the system derive themselves from the capabilities of the 4x100W Channel Class D Amplifier. This board has 4 separate channels each capable of driving a speaker at up to 36W. This allows separate 2 stereo channels to exist in the system. The board uses 2 TDA7498 dual channel class D amplifier ICs to amplify incoming signals. This board also contains an input and output stage which allow the board to receive an audio input and drive a set of speakers. Appendix C below contains the schematic for this board. The board has a set of 2 switches for each output channel which control the amount of gain the board produces. The functionality of these switches is detailed below in Figure 5.

Parameter	Condition	Min.	Typ.	Max.
Gain(SW1/SW3 Setting)	K1 ON, K2 ON	24.6	25.6	26.6
	K1 OFF, K2 ON	30.6	31.6	32.6
	K1 ON, K2 OFF	34.1	35.1	36.1
	K1 OFF, K2 OFF	36.6	37.6	38.6

FIGURE 5: GAIN SETTING SWITCH OPTIONS FOR CLASS D BOARD [3]

b. Audio Switching Stages

The Audio Switching Stages give the system the ability to switch the controlling input of each output. As shown below in Figure 6, a pair of NMOSFET transistors tied at their drains creates the switching capability. Each transistor receives an audio input at its source terminal, which is also connected through a 1 kΩ resistor to ground. The gates of the transistors are tied to either side of an inverter which drives the gate at either 8V or -8V depending on the state of the external user controlled switch. If the user selects 8V then the transistor driven by the non-inverted signal becomes active while the inverted signal disables the other transistor. Because the bulk contact of the transistor is connected to -8V, the transistors are able to turn off when -8V is applied to the gate terminal. This ensures only 1 signal enters the tied drains at any given time. The capacitor on the non-inverted side of the inverter ensures that when the switch is in its floating state, which is not represented in the schematic, the voltage driven into the gates stays at its previous value. Without this capacitor, multiple signals could be driven simultaneously causing audio corruption. The tied drains pass the audio signal to the next point in the audio signal path. 6 of these audio switching stages are present in the circuit and each pair of transistors exists on a CD4007UBE IC. 3 Inverters are used in the circuit and they all exist on the same CD4069UBE IC.

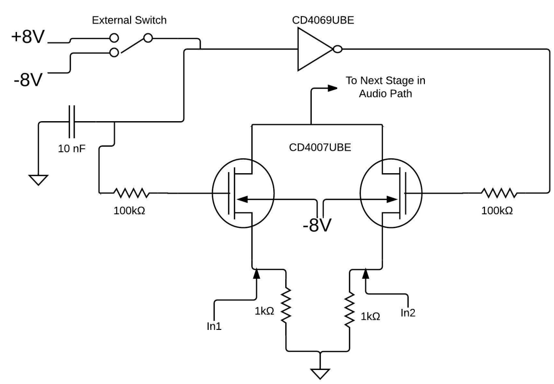

FIGURE 6: AUDIO SWITCHING STAGE SCHEMATIC

c. Split Supply Power Generator

The Split Supply Power Generator, shown below in Figure 7, creates a series of DC voltage used throughout the system. A set of 12V wall transformers tied together create an unregulated split 12V supply. The positive 12V is bypassed by a Mylar 367nF capacitor before it enters the LM7805 and LM7808 ICs. These LDO positive voltage regulators take the +12V as an input and regulate it down to +5V and +8V respectively. The output of each of these ICs is bypassed by a ~100nF polystyrene capacitor before being used by the system's circuitry. The negative 12V is bypassed by a 22uF electrolytic capacitor before entering the LM7905 and LM7908 ICs. These LDO negative voltage regulators take the -12V as an input and regulate it up to -5V and -8V respectively. The output of these 2 ICs are each bypassed by a 10uF electrolytic capacitor before being used by the system's circuitry. Electrolytic capacitors ensure stability of the negatively regulated voltages and help eliminate noise. The ground created in this power generator is the ground of the entire circuit.

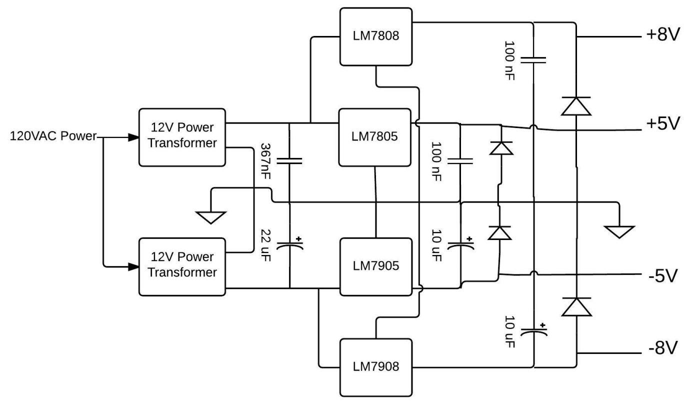

FIGURE 7: SPLIT SUPPLY POWER GENERATOR SCHEMATIC

d. Class D Board Power Controller

The Class D Board Power Generator, shown below in Figure 8, creates a toggle-able 24V/14.6A signal used to power the class D amplifier board. This voltage is generated using the regulated switching power supply shown in the schematic which takes in 120V AC power and converts it to the desired 24 VDC signal. A relay circuit driven by an NPN transistor creates the toggle functionality of the power generator. The relay passes the AC voltage to the switching supply when 5V is applied across its coil terminals. An external switch tied between 5V and a 1 kΩ base driving resistor turns the NPN transistor on and off. When the transistor is off the diode turns on, dropping 0.7V across it which is insufficient to turn on the relay. Thus, AC power is not passed to the switching power supply and the class D board stays off. When transistor turns on, the diode turns off, becoming an open circuit. This creates a 5V difference across the coil of the relay and so it activates and passes the AC power to the switching power supply. The collector terminal of the NPN is also monitored by the Arduino. If it sits above 4V, indicating an off state, the Arduino turns the LED circuitry off. If the collector sits at 0V, indicating an on state, the Arduino turns the LED circuity on.

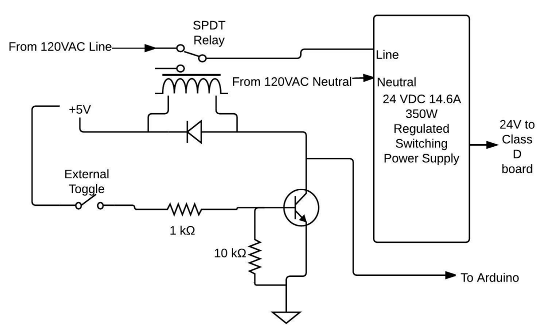

FIGURE 8: CLASS D POWER GENERATOR SCHEMATIC

e. SainSmart Arduino LCD Display Shield

The SainSmart Arduino LCD display shield allows for easy interfacing between the LCD screen and the Arduino. A potentiometer exists on the shield which controls the contrast of the LCD screen. See Appendix C for Schematic

f. Pyle PP444 Mini Phono Preamp

This preamp takes in a phono level signal and amplifies it to line level. Schematic not available.

g. Volume Control

An external linear rotary potentiometer creates the volume control capabilities of the device as shown below in Figure 9. One side of the potentiometer is tied to the +5V generated by the split supply and the other end is tied to ground. The middle terminal of the potentiometer is given to the Arduino as a voltage input. This signal varies between 5V and ground and the Arduino uses it to program the stereo potentiometers. A 5V input corresponds to the maximum volume while a 0V input corresponds to the lowest volume. The voltages don't reach a full 5V or ground due to the nature of the potentiometer but they come close.

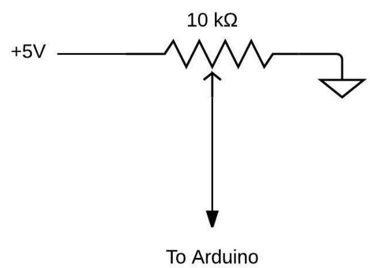

FIGURE 9: VOLUME CONTROL SCHEMATIC

h. LED Control Circuit/LEDs

The LED control Circuitry, shown below in Figure 10, displays the current input/output configurations of the device. 6 MPS750 PNP transistors drive 6 LEDs. Each LED is connected between the collector of the PNP and ground. These LEDs have an on voltage of approximately 1.7V each. The bases of each transistor are tied together and are controlled with a digital +5V or 0V signal coming from the Arduino. The Arduino supplies 0V when the external power switch is in an on state and a +5V signal when the switch is off. The emitter terminals of each PNP are tied through a resistor to 1 of the 6 points in the circuit which change with the external user switches. Assuming the Arduino is supplying a low signal to the gates, the PNP transistor turns on when its emitter is tied to a +8V signal and turns off when its emitter is tied to a -8V signal. The blue LED becomes active when the phono input is selected. The first 2 red LEDs can be considered a pair as neither are on simultaneously since they are tied to either side of an inverter. They indicate the active input of channel 1. The next 2 red LEDs create the same functionality for channel 2. The green LED is tied to the +8V of the split supply generator and thus becomes active any time the Arduino supplies a low voltage signal at its base. This LED thus gives an indication of the system's power state.

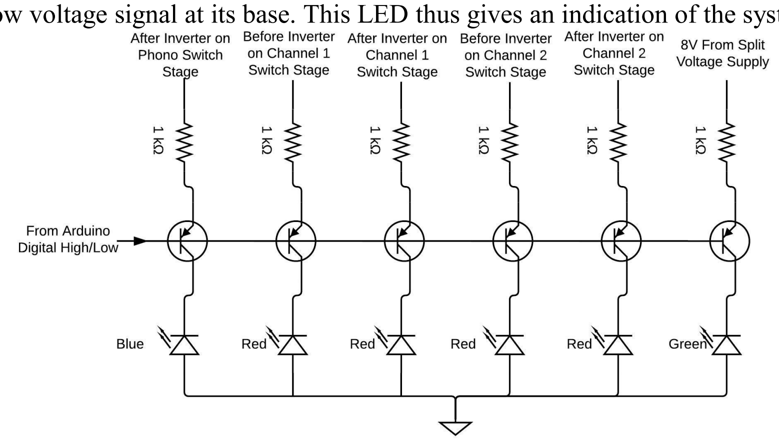

FIGURE 10: LED CONTROL CIRCUIT SCHEMATIC

i. Stereo Digital Potentiometers (IC: PGA2311P)

I chose to use the PGA2311P stereo digital potentiometer from TI for my digital audio control due to its low noise characteristics and its SPI digital interface. The schematic for the device is shown below in Figure 11. This device attenuates an audio input based on the settings programmed by the controlling digital interface. The 2 potentiometers share the same data line and clock signal but are activated using different chip select signals. This SPI interface programs the IC with 16 bits corresponding to a level of attenuation for the left and right channels. The bits are read in on the rising edge and there is 1 bit read per clock cycle after the chip select goes low. The bits programmed correspond to an integer N which determines the gain of the potentiometer through equation I below.

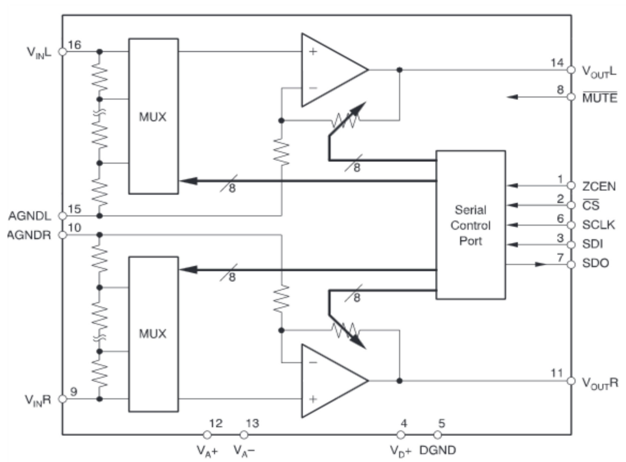

FIGURE 11: STEREO DIGITAL POTENTIOMETER SCHEMATIC [4]

For N = 1 to 255:

Gain (dB) = 31.5 − [0.5 w (255 − N)]

This results in a gain range of +31.5dB (with N = 255) to −95.5dB (with N = 1).

EQUATION I: PGA2311P GAIN EQUATION [4]

j. Arduino Uno R3

The Arduino microcontroller takes in the volume control input from external volume control knobs and converts this into the corresponding bit values to transfer to the digital potentiometers. The SPI interface programs each IC with 16 bits corresponding to a level of attenuation for the left and right channels. The Arduino also uses the volume control data to display the current level of volume on the LCD screen. The Arduino also takes in the power indication voltage and uses it to control the LED control circuit with a digital signal. The schematic for the Arduino is available in Appendix C.

B: Housing

A 1 inch thick wooden board acts as the bottom foundation of the system. All subsystems attach to this foundation with mounting tape. To further protect the system and to mount the input and output interfaces, peg board covers the 3 sides of the foundational wooden board. The top remains open to help with heat dissipation and in case of needed repairs. The full device measures 14” long, 10.75” wide and 4.75” tall.

C: Software

Standard Arduino sketch software was used to generate the code necessary for all tasks performed by the Arduino Uno. The code used in the device is given in Appendix D and was generated using the arduino website as a reference. [5]

<u>Chapter IV:</u> Issues Encountered

A: Input Switching Stage

The type of transistor was the first design decision made in the design process. To choose between BJTs and MOSFETs, LTSpice simulations were conducted on the 2 circuits. These simulations yielded no visible differences between the two except for increased supply voltage needed for MOSFETs. However, MOSFETs have a much higher input resistance than BJTs [6], which is needed to receive the audio signal. MOSFETs also operate much better in switching applications where they operate in either an on or off state. This is because BJTs have significant leakage when off due to the nature of the electron channel present.[7] This does not occur in MOSFETs due to the separation of the gate contact from the rest of the device. Leakage in this circuit would detrimentally affect audio quality and thus MOSFETs were chosen.

The first design of the MOSFET input switching portion of the project resulted in a clipped audio running through the FETs. During this time, the IC containing the FETs operated from a split 5V supply. Due to the typical 3-4V V_{DS} of the FETs, this 5V level was not enough to accommodate a $1V_p$ input signal. Changing the supply voltage of the IC to a split 8V fixed this issue right away. With the split 8V the FETs have plenty of room to operate above their necessary V_{DS} and allow the entire audio signal to pass through unaffected.

B: Input Stage Split Power Supply

The decision to use linear regulators came as a result of numerous failed experiments with split voltage creation from a single supply voltage. This was attempted by trying to create a virtual ground with an op-amp using the circuit shown below in Figure 12.

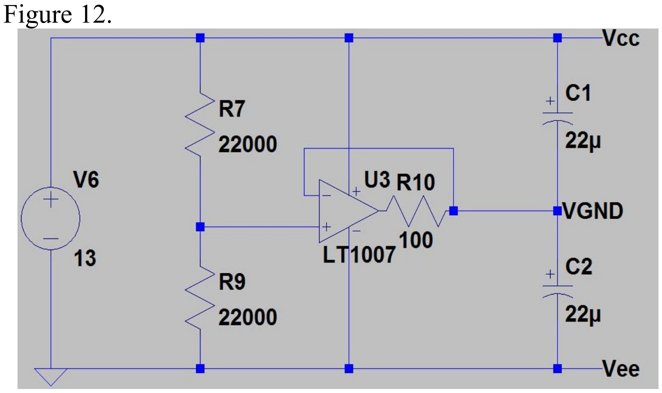

FIGURE 12: SPLIT SUPPLY VOLTAGE CREATION WITH AN OP-AMP

Since the virtual ground was actually 6.5V above the rest of the system ground, this point could not act as a ground reference for the whole system. Thus, a true split supply voltage was needed. To accomplish this, I tied 2 readily available 12V Netgear wall transformers together to create a split 12V power supply. This split voltage was then regulated to a split 5V and a split 8V via linear regulators.

C: Stereo Volume Control

Initially, my goal with this project was to keep the entire audio path in the analog domain. However, after observing the price of analog volume control options which often exceeded $40 for each channel, I decided to use the PGA2311P digital potentiometer for volume control. This device uses an SPI interface to program the desired attenuation level and operates from a split 5V supply. Controlling 1 potentiometer posed no issues but I needed to control 2 separate channels simultaneously. This presented a challenge since the Arduino Uno R3 only has 1 SPI data line. To fix this issue, I connected the SPI lines of both potentiometers to the same Arduino output but I gave the devices 2 separate chip select signals. This allowed for continuous programming of 2 digital potentiometers without much latency or signal corruption.

D: Speaker Popping

Once the input stage and output stage were connected together, issues arose with speaker popping during power down. When power was removed from the circuit, the input stage power turned off immediately while class D switching power supply turns off in a gradual fashion. This results in a small window where the output stage is on with the input stage off. This results in the amplification of any leftover signals present on the amplifier's input terminal and creates a loud speaker popping if the volume is at a substantial level. To fix this, a power sequence was added which turns the output stage off while leaving the input power on. This was accomplished with the class D power controller circuit described above in part d of the level 2 description found in chapter III of this report.

E: Preamp

Initially, I designed my own preamp shown below in Figure 13 for the system based on the TI LM833 OP-Amp [8].

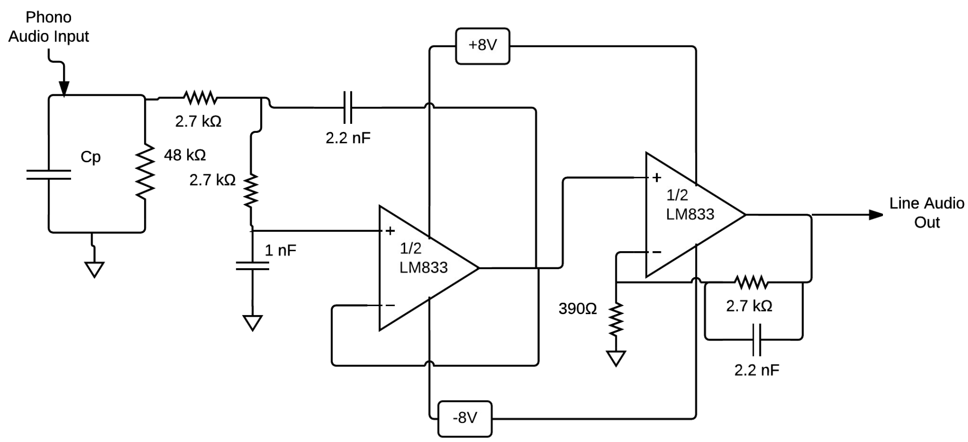

FIGURE 13: INITIAL PREAMP DESIGN [8]

This preamp accomplished the gain needs of a preamp but caused magnetic interference with my turntable. This magnetic interference is due to the large inductor present in the music capturing circuit of the turntable. To prevent audio distortion during the acquisition of this signal a capacitance needs to correctly match the inductance so that a pole is not introduced in the audible frequency range while also resonating out the inductance over the audible frequency range. After many hours testing various capacitance and design tweaks, I was unable to build a preamp which did not introduce noise. My amp introduced significant noise at certain volume levels. This was likely due to the introduction of parasitic poles from my op-amp based design. During this stage of my project, I was very close to completion and I ran out of time to design a more useful preamp. So, I purchased a Pyle PP444 Mini Phono Preamp and integrated it into my project by soldering my +12V supply line directly to its power terminal. This preamp works as desired, introducing no noise over the audible frequency range and achieving a near constant gain response.

<u>Chapter V:</u> Results and Testing

Final System:

The final system fulfills all main needs of the customer. There are 2 separate output channels, each with a left and right speaker terminal, on the back panel as shown in Figure 14.

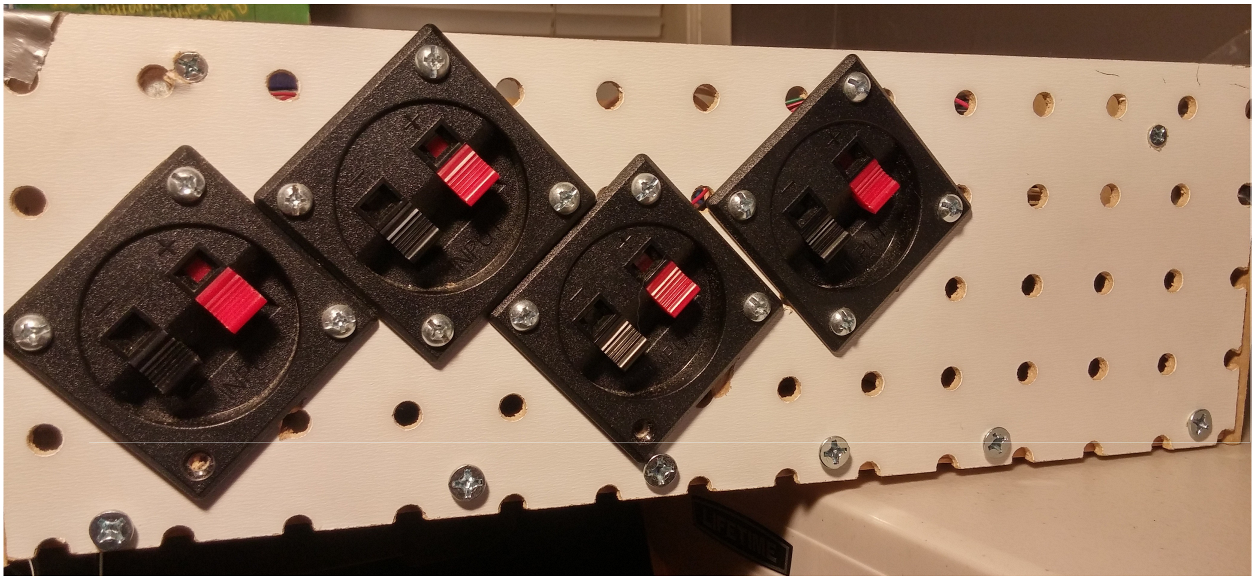

FIGURE 14: BACK PANEL OF DEVICE SHOWING SPEAKER TERMINALS

Each of these output channels is controlled by the switches on the front panel, shown in Figure 15. The first output has one switch which changes between source 1, on the left, and source 2 on the right. If source 1 is chosen the bottom LED lights up and if source 2 is chosen the top LED lights up. Output channel 2 is controlled by one switch in the same way with similar indicating LEDs. The third switch on the far right determines what enters the device through audio source 2. If the blue LED activates, this indicates the phono source on the far right is active and if the blue light is off the line source in the middle right is active. The 2 volume knobs on the right of the LCD control each channels volume with the top one controlling channel 1 and the bottom controlling channel 2. This volume is also displayed on an LCD screen to the user. The toggle switch on the far left controls the power to the class D amplifier board. When on, the LCD and LEDs display their states and output can be taken from the device. When this switch is off, no LEDs are active, the LCD screen stays blank and all device outputs are disabled.

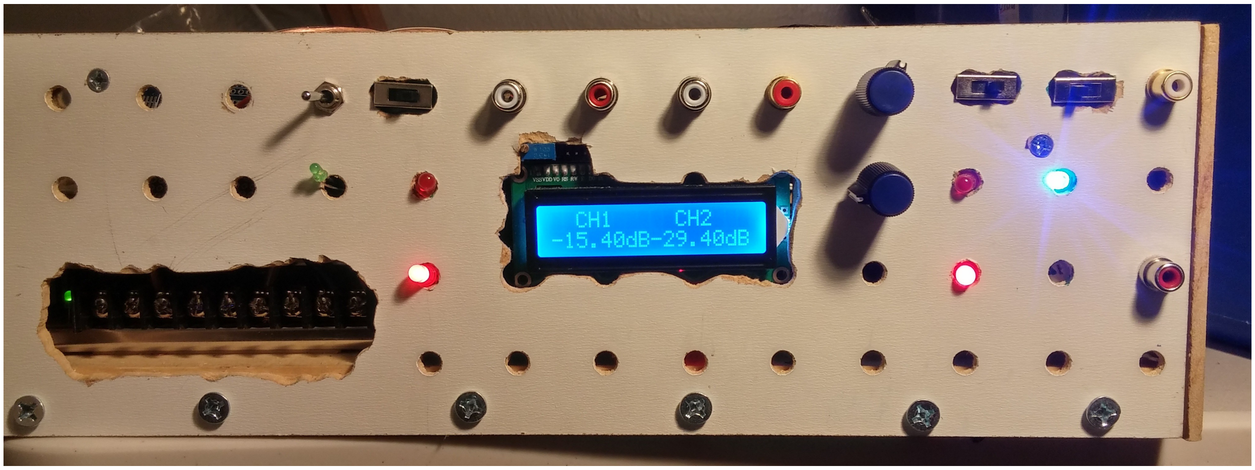

FIGURE 15: FRONT PANEL OF DEVICE

Figure 16 below shows all of the internal circuitry of the device. All power circuitry is on the left, the amplifier board is in the top middle and everything else is part of the input stage.

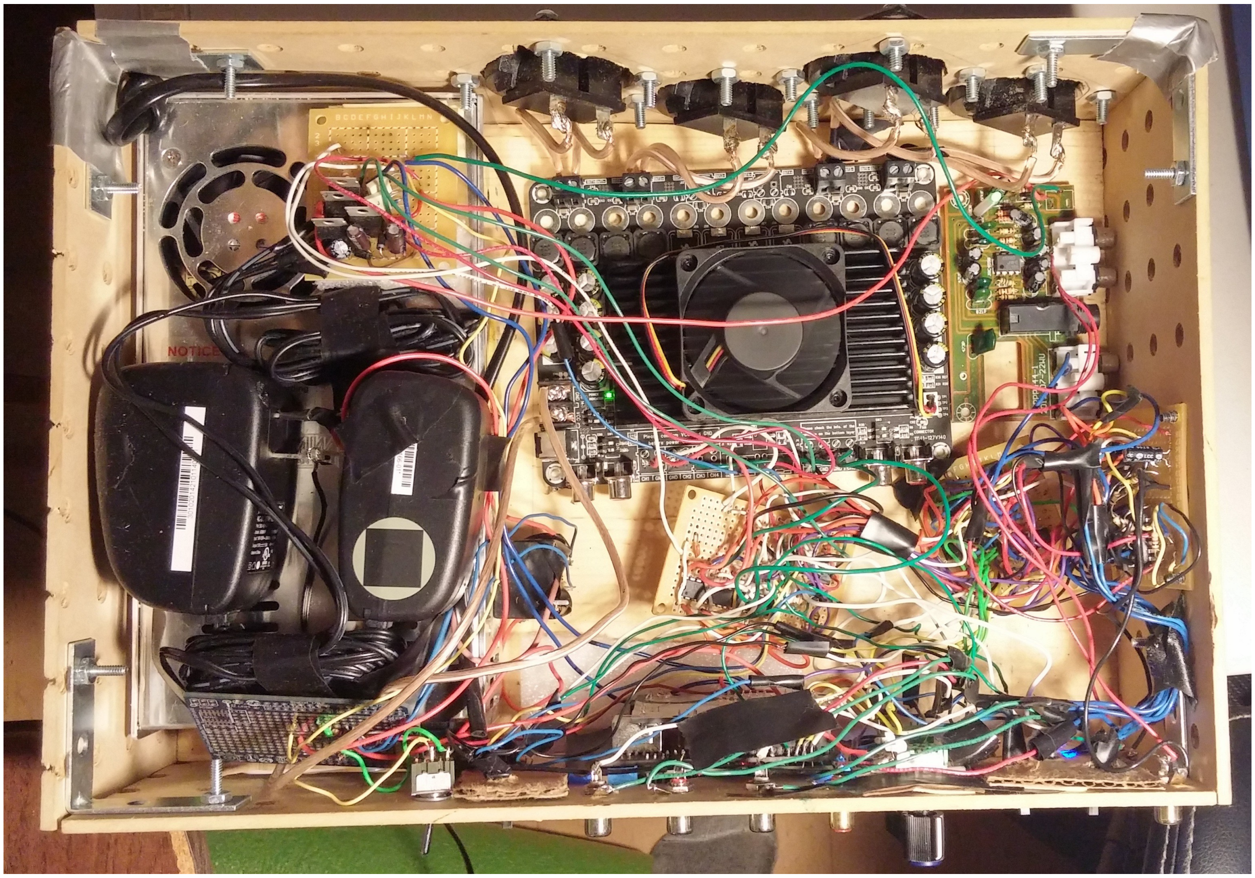

FIGURE 16: TOP VIEW OF DEVICE SHOWING ALL INTERNAL CIRCUITRY

Due to a lack of access to high quality audio testing equipment, device testing for characterization proved to be difficult. I could not monitor the waveforms with any oscilloscope due to their low impedance. Thus, I was unable to accurately measure THD, system gain, output power or input power. However, I was able to monitor the RMS voltages of the input and output waveforms and also theoretically determine the input impedance of the device. Using these values, I obtained data on the system gain which was then extrapolated to determine output power, input power and gain for the various device configurations.

Input Impedance:

This amplifier is a form of a common gate amplifier. The full small signal model for this amplifier is shown below in Figure 17. This model ignores the gate-drain capacitance and all connections with the bulk terminal.

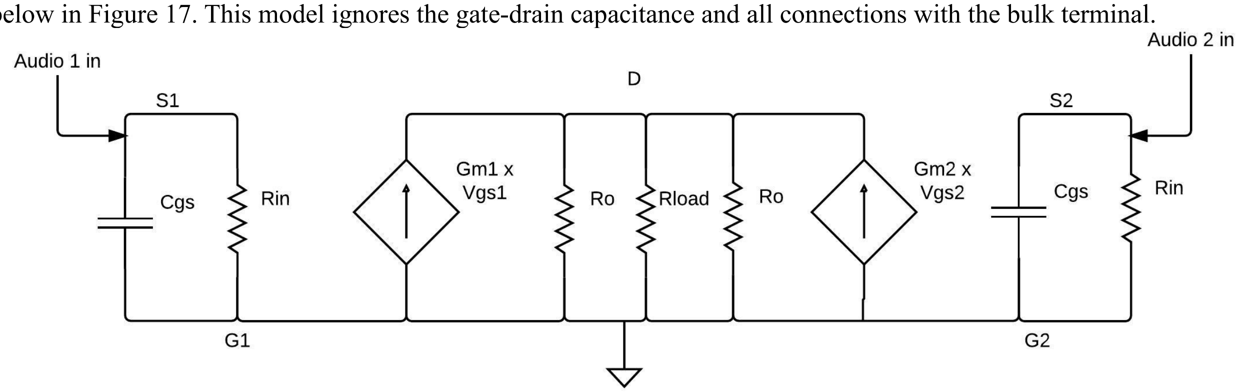

FIGURE 17: FULL SMALL SIGNAL MODEL FOR AUDIO SWITCHING STAGES

Only one of the 2 MOSFETs are on at any given time. Thus, the small signal model can be simplified to the schematic shown below in Figure 18.

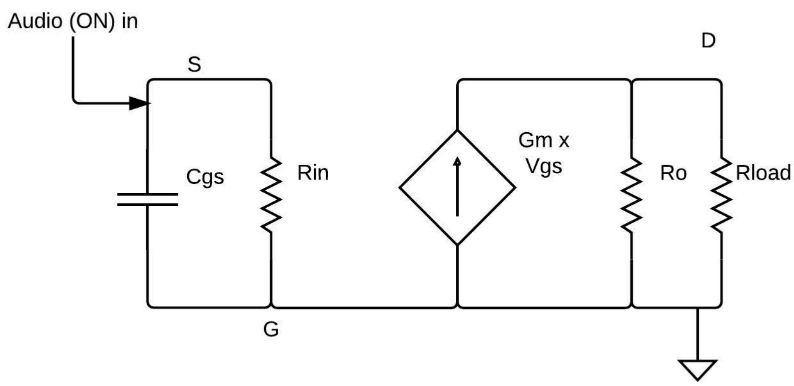

FIGURE 18: SIMPLIFIED SMALL SIGNAL MODEL FOR ON TRANSISTOR

The input impedance (R_{in}) of the audio switching stage is this parallel combination of the 1kΩ source resistor and the resistance seen looking into the source terminal of the MOSFET, which is $1/g_m$. In the on state, g_m is given by equation I below.

$$g_{mNMOS} = 2\frac{I_D}{(V_{GS} - V_{tn})}$$

EQUATION II: EQUATION FOR FINDING THE TRANSCONDUCTANCE (G_M) OF AN NMOS DEVICE

Using a threshold voltage of 1.2V for the 4007UBE NMOS transistor [9], a gate-to source voltage of 7.6V, which was measured, and assuming the max quiescent current as given by the CD4007 data sheet, a g_m of 0.15625 µS results. This results in an impedance of 6.4MΩ looking into the source terminal. Since this value is 3 significant figures above that of the 1kΩ source resistor, it can be ignored and the input impedance of the switching stage can be approximated to 1kΩ.

Test Setup

I first performed various functionality tests on the input stages. These tests consisted of setting up each circuit as they are displayed in the level II part of Chapter III above and testing their functionality. I was able to observe successful functionality of each input stage subsystem. Once each stage was tested, I built it on a protoboard and connected it with the other subsystems of the input stage until the input stage was complete. At that point I connected the input stage to the output stage and began testing full system functionality.

Once the device was completed, I performed a test to measure the input and output voltages, in RMS, as shown below in Figure 19.

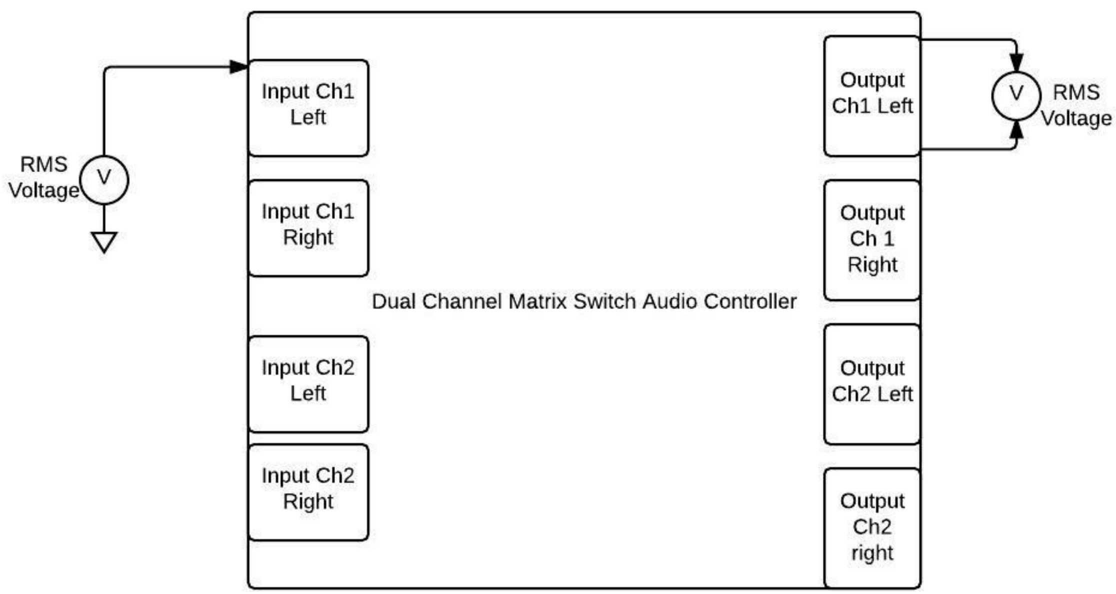

FIGURE 19: MEASURING RMS INPUT AND OUTPUT VOLTAGE

Voltage was taken using 2 Agilent multimeters. This test was done using a headphone source and the maximum possible gain setting both on the class D board, which is approximately +37.6dB [3], and the PGA2311P, which is approximately +31.5dB as determined by equation I.

The data obtained from this test is displayed below in Table IV and the results are further detailed in the extrapolated results section.

Test Results

System Gain:

TABLE IV
RMS VOLTAGE MEASUREMENTS AND VOLTAGE GAINS FOR MAX VOLUME

	CH1 IN1 L	CH1 IN1 L	CH1 IN1 R	CH1 IN1 R	CH2 IN1 L	CH2 IN1 L	CH2 IN1 R	CH2 IN1 R
PGA Integer	250.00	250.00	250.00	250.00	250.00	250.00	250.00	250.00
Input Voltage (VRMS)	0.006101	0.006053	0.006113	0.00623	0.00621	0.006266	0.005017	0.005016
Output Voltage (VRMS)	8.02	8.55	8.6948	8.8323	8.9054	8.7552	9.1957	9.0928
Gain (VRMS)	1314.54	1412.52	1422.35	1417.70	1434.04	1397.26	1832.91	1812.76
Gain (dbV)	62.38	63.00	63.06	63.03	63.13	62.91	65.26	65.17

Equations derived from [10]

Extrapolated Results:

The TDA7498 class D amp used in this device uses a bridge-tied load configuration. This configuration consists of one amplifier driving one side of the load while a second amplifier drives the other side of the load with an inverted version of the first signal.[11] As a result, the full 24VDC supply can be used to drive the speakers instead of just half of the supply as is the case with a single ended load. Using Equation III below with an assumed 8Ω load, the maximum power deliverable to the load was calculated to be 36W.

$$P=\frac{\left(\frac{V_{peak}}{\sqrt{2}}\right)^{(2)}}{R} \qquad Gain=10\log\left(\frac{P_{out}}{P_i}\right)$$

EQUATION III: AVERAGE POWER CALCULATION FROM PEAK VOLTAGE AND POWER GAIN IN DB [10]

The same equation was used with the data in Table IV to calculate the input and output power shown below in Table V. 1KΩ was used as the input resistance, as determined previously, and 8Ω was used as the output resistance.

TABLE V
INPUT AND OUTPUT POWER AND RESULTING POWER GAIN FOR HEADPHONE SOURCE

	CH1 IN1 L	CH1 IN1 L	CH1 IN1 R	CH1 IN1 R	CH2 IN1 L	CH2 IN1 L	CH2 IN1 R	CH2 IN1 R
Pin	3.722E-008	3.66E-008	3.737E-008	3.881E-008	3.856E-008	3.926E-008	2.517E-008	2.516E-008
Pout	8.04005	9.1378125	9.44994338	9.75119041	9.91326865	9.58169088	10.5701123	10.3348765
Pgain (dB)	83.34	83.97	84.03	84.00	84.10	83.87	86.23	86.14

To calculate the correct gain setting of the device for a line input, I first calculated the power gain needed to amplify a line level to the maximum voltage level of the output. I started by first assuming that a line source inputs a voltage with $1.1V_p$, or $0.778V_{RMS}$, and the maximum voltage output of the device is $24V_p$, or $16.9706V_{RMS}$. This leads to an input power of 0.605mW and a resulting output power of 36W, calculated the same way as for the headphone source. These power's result in a gain of 47.75dB. This gain represents the maximum gain that the device can handle if a line source is used as the input. Since the device is capable of achieving a gain of 83-86dB with the PGA and the output board both set to maximum, the two must be limited to prevent damaging of speakers. The values for each part of the device to correctly limit the gain are detailed below in Table VI.

TABLE VI

GAIN SETTINGS TO USE FOR EACH TYPE OF SYSTEM INPUT

Input	**Usable Input Source**	**PGA Gain Setting**	**Amp Board Gain Setting**	**Total Max Gain**	**Input Sensitivity**	**Max Output Power**
Line	Either Line Input	Limit to 0dB. Maximum integer of 192 as determined by equation I.	25.6 dB (K1 On, K2 On)	39.5–42.5 dB	1.1 V_p	36W
Headphones/PC	Either Line Input	No Limit needed Allow for full 255 integer range as determined by equation I.	37.6 dB (K1 Off, K2 Off)	83-86 dB	5mV_p -20mV_p	36W
Phono	Phono Input Only	No Limit needed Allow for full 255 integer range as determined by equation I.	37.6 dB (K1 Off, K2 Off)	83-86 dB (extra gain from Preamp)	5mV_p -50mV_p (Typical for phono input)	36W
Recommended*	All Inputs	Limit to 18.6dB. Maximum integer of 220 as determined by equation I.	31.6 dB (K1 Off, K2 On)	64.1-67.1 dB	5mV_p -100mV_p	36W

*The recommended settings ensure a good level of volume for headphone/PC inputs, which is the device's intended use, while still placing some limiting on the gain. This ensures a greater range of volume control over the listenable regions of the gain curve while maintaining a consistent maximum gain of 67.1 dB. If a line source is used on this setting, great care must be taken by the user to ensure their speakers can handle the output power.

Verification of system specifications is detailed on the following page in Table VII.

TABLE VII

TESTING AND VERIFICATION OF ENGINEERING SPECIFICATIONS

Engineering Specifications	Testing	Verification
1. The system should operate from standard 120V 60Hz AC power	Visual/Functionality Test	System has a single wall plug for operation.
2. The system should output 2 separate, stereo and mono capable audio channels	Visual/Audible inspection	The device contains all necessary outputs.
3. The system should take in 2 RCA inputs and 1 phono input	Visual/Audible inspection	The device contains all necessary inputs.
4. Each output channel must output each system input with no audible signal corruption added	Audible Inspection	Each I/O combination tested and confirmed.
5. The system should have no protruding wiring or components other than those necessary for input and output interfacing	Visual inspection	Only protruding components are those necessary for user interfacing and display of device states to the user.
6. A user must have control of each output channel's volume.	Audible Inspection	Volume control knobs change the output volume level.
7. The system should visibly display volume and active input for each output channel.	Visual inspection	Device displays volume in dB for each channel.
8. Each amplified system output should drive an 8Ω load at a maximum average power between 25W and 30W for a line input source.	Frequency sweep test (multiple sinusoids of different frequencies while monitoring the output)	All audio input to device plays on output. However, more testing is needed to verify full frequency range.
9.Each amplified system output should drive an 8Ω load at a maximum average power between 8W and 10W for a headphone input source.	Extrapolate from measured gain in dB and note that it is done this way	System meets output power specs as shown above in the results section.
10. Each amplified system output should have the capability to amplify a signal at a level of 63dB into an 8Ω load.	Extrapolate from measured gain in dB and note that it is done this way	System meets gain specs as shown above in the results section.
11. The system enclosure should not exceed 16”x12”x8”	Measurements taken on device.	Device measures 14” x 10.75” x 4.75”

<u>Chapter VI:</u> Future Work

To improve the characterization of this device, I need access to better audio testing equipment. This would allow me to test THD, output power, gain and efficiency and give a much better representation of the capabilities of the system. One such device which allows for this kind of testing is the Keysight Technologies U8903A Audio Analyzer. [12]

Creation of mass produced, PCBs and housing units would drastically improve the efficiency, cost and performance of this project. This would eliminate much of the labor intensive soldering and interface creation which took up a majority of the project's labor hours. It would also improve the safety of the device because it would eliminate the large amount of interconnecting wires used in project construction. If produced commercially, this strategy would definitely be implemented, as well as the creation of a cheaper amplifier integrated into the system.

Another place where this project needs work are the power supplies. Instead of using 1 supply for the output stage and an entirely separate one for the input stage, the two supplies could be integrated together. This would free up a lot of space within the device enclosure as well as eliminating a great amount of power waste. Also, increasing the capacity of the switching power supply driving the class D board will greatly increase the output power capabilities of the device. Using a 36V supply, which is the max allowed by the board, would increase the output power capabilities of the device from a max of 36W to a maximum of 81W for an 8Ω load. Increasing the input resistance of the audio switching stages would also greatly increase the efficiency of the device as well as the usable output power.

Adding a bluetooth input and a digital hard drive source would greatly expand the capabilities of the device and would not be hard to implement. Initially, a bluetooth source was included in the device and coupled in via another set of input switches. However, the module was damaged due to a power fluctuation and thus was not included. The digital hard drive would require a more powerful microprocessor due to the significant speed and data limitations of the Arduino Uno. With this improved microprocessor the power control circuitry of the device could also vastly improve. Since the new microprocessor would be much faster and have more I/O ports, more output channels could be added to the device, expanding its capabilities.

As shown in the result section of this report, the gain is changed by both the PGA2311 code and the constant gain switches on the class D amplifier board. To improve this project, an option could allow the user to select their desired input and the software would limit the gain appropriately to accommodate for that input. This way the user would never have to reprogram the Arduino to change the digital potentiometer settings.

TABLE VIII

DUAL OUTPUT AUDIO RECEIVER DELIVERABLES

Delivery Date	Deliverable Description
12/5/2014	Design Review
3/13/2015	EE 463 demo
3/13/2015	EE 463 report
5/29/2015	EE 464 demo
11/3/2014	ABET Sr. Project Analysis
5/29/2015	Sr. Project Expo Poster
6/10/2015	EE 464 Report
6/10/2015	Deliver finished product to John Fox

References

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[2] R. Ford and C. Coulston, *Design for Electrical and Computer Engineers*, McGraw-Hill, 2007, p. 37

[3] Sure Electronics, “4x100Watt Class D Audio Amplifier Board – TDA7498 User's Guide,” datasheet, 2011 Available: http://www.parts-express.com/pedocs/manuals/320-305-parts-express-manual.pdf

[4] Burr-Brown Products from Texas Instruments, “Stereo Volume Control,” PGA2311 Datasheet, Dec. 2001 [Revised June 2002] Available: http://www.ti.com/lit/ds/sbos218a/sbos218a.pdf

[5] Arduino, “Language Reference,” Arduino Sketch Language Reference, 2015 Available: http://www.arduino.cc/en/Reference/HomePage

[6] Peter Blair, “MOSFET or bipolar, which should you use?”, *EE Times*, UBM Tech, 9, Feb. 2005, Available: http://www.eetimes.com/document.asp?doc_id=1272218

[7] Dr. Alan Doolittle, “Bipolar Junction Transistors (BJT): Part 1 Qualitative Understanding – How do they work?”, Georgia Tech Available: http://users.ece.gatech.edu/~alan/ECE3040/Lectures/Lecture17-BJT%20QualitativeAndPerformParam.pdf

[8] Texas Instruments, “LM833-N Dual Audio Operational Amplifier”, datasheet, May 2004 [Revised May 2012] Available: http://www.ti.com/lit/ds/symlink/lm833-n.pdf

[9] Nikhil Keeppanasseril “Tutorial for I-V Characterization using Cascade Probe Station and Agilent Parameter Analyzer,” San Francisco State University, Fall 2013 Available: http://userwww.sfsu.edu/necrc/files/cadence%20and%20agilent%20tutorial/MOS_Characterization.pdf

[10] Joe Roberts, “Audio Power Amplifier Power Rating Mysteries Explained”, Rev F, *Joe's Tech Notes,* 5 April 2007, http://www.rocketroberts.com/techart/powerart_a.htm

[11] Texas Instruments, “Calculating Gain for Audio Amplifiers,” Application Report, Oct. 2003 [Revised Sep 2005] Available: http://www.ti.com/lit/an/sloa105a/sloa105a.pdf

[12]Keysight Technologies, “Keysight U8903A Audio Analyzer,” Datasheet, 2 Aug. 2014 Available: http://literature.cdn.keysight.com/litweb/pdf/5990-8499EN.pdf?id=1724283

[13] *IEEE Std 1233, 1998 Edition*, p. 4 (10/36), DOI: 10.1109/IEEESTD.1998.88826

Appendices

Appendix A: Bill of Materials

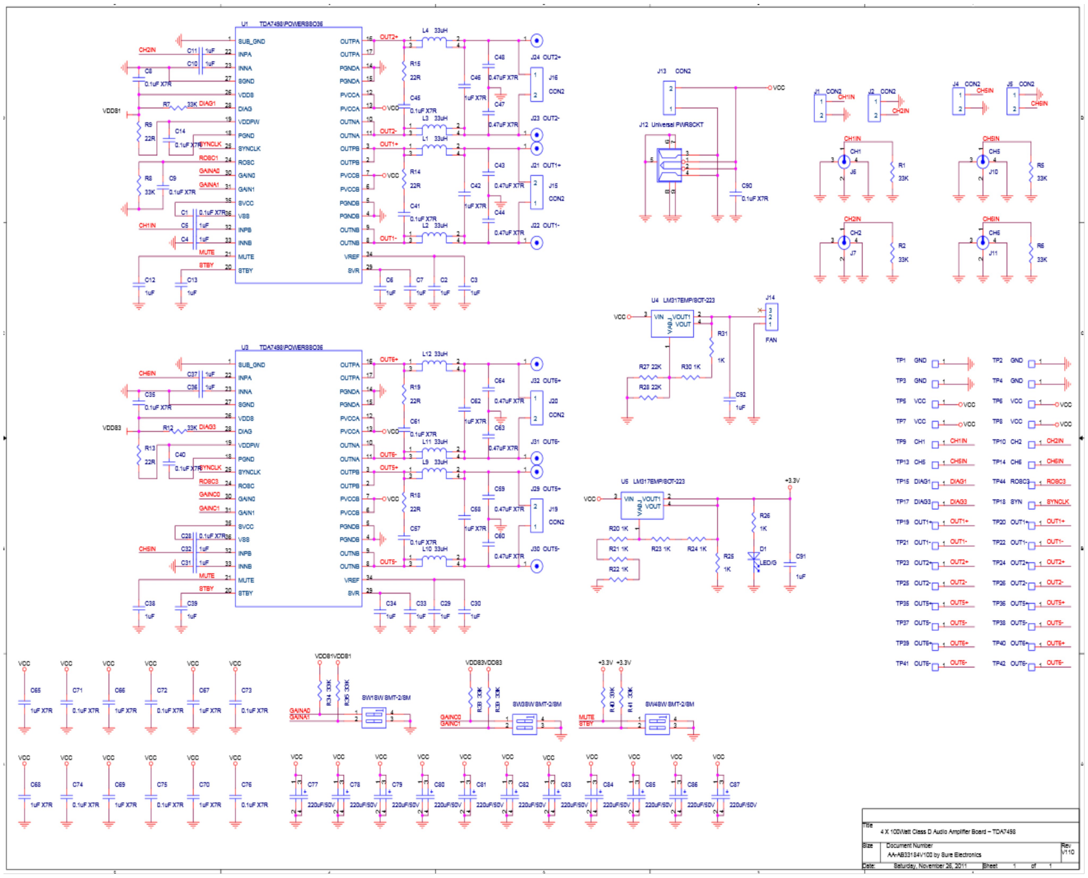
Table IX
Bill of Materials

Item	Quantity	Cost Per	Total Cost
4x100W DA7498 Class-D Amplifier Board	1	$49.90	$49.90
Arduino Uno R3	1	$30	$30
SainSmart LCD Keypad Shield for Arduino	1	$12.98	$12.98
24 VDC 14.6A 350W Regulated Switching Power Supply	1	$49.62	$49.62
12V Netgear Power transformers	2	Reused	-
LM7805 +5V LDO regulator	1	$0.67	$0.67
LM7905 -5V LDO regulator	1	$0.60	$0.60
LM7808 +8V LDO regulator	1	$0.70	$0.70
LM7908 -8V LDO regulator	1	$0.68	$0.68
Slide Switch SPDT (300mA, 125V)	3	$1.68	$5.04
RCA input jacks	3	$1.76	$5.28
1.7V LEDs (4 Red, 1 Green, 1 Blue)	6	$0.30	$1.80
1kΩ Resistors	20	$0.10	$2.00
100kΩ Resistors	6	$0.10	$0.60
10kΩ Resistor	1	$0.10	$0.10
10kΩ 14MM Dual Bush Linear Rotary Potentiometers	2	$0.98	$1.96
SPDT Toggle Switch (6A, 125V)	1	$3.31	$3.31
1N4001 Diodes	5	$0.10	$0.50
Capacitors (1 367nF Mylar, 1 Electrolytic 22uF, 2 ~100nF polystyrene, 2 10uF electrolytic, 3 10nF ceramic)	9	$0.20	$1.80
MPS750 PNP Transistors	6	$0.10	$0.60
2N2222 NPN Transistor	1	$0.10	$0.10
Protoboards	4	$2.49	$9.96
TI PGA2311P Digital Potentiometers	2	Free sample	Free sample
COM-10924 Relay SPDT Sealed – 20A	1	$2.95	$2.95
CD4069UBE CMOS Hex Inverter	1	$0.52	$0.52
CD4007UBE CMOS Dual Complementary Pair Plus	6	$0.52	$3.12

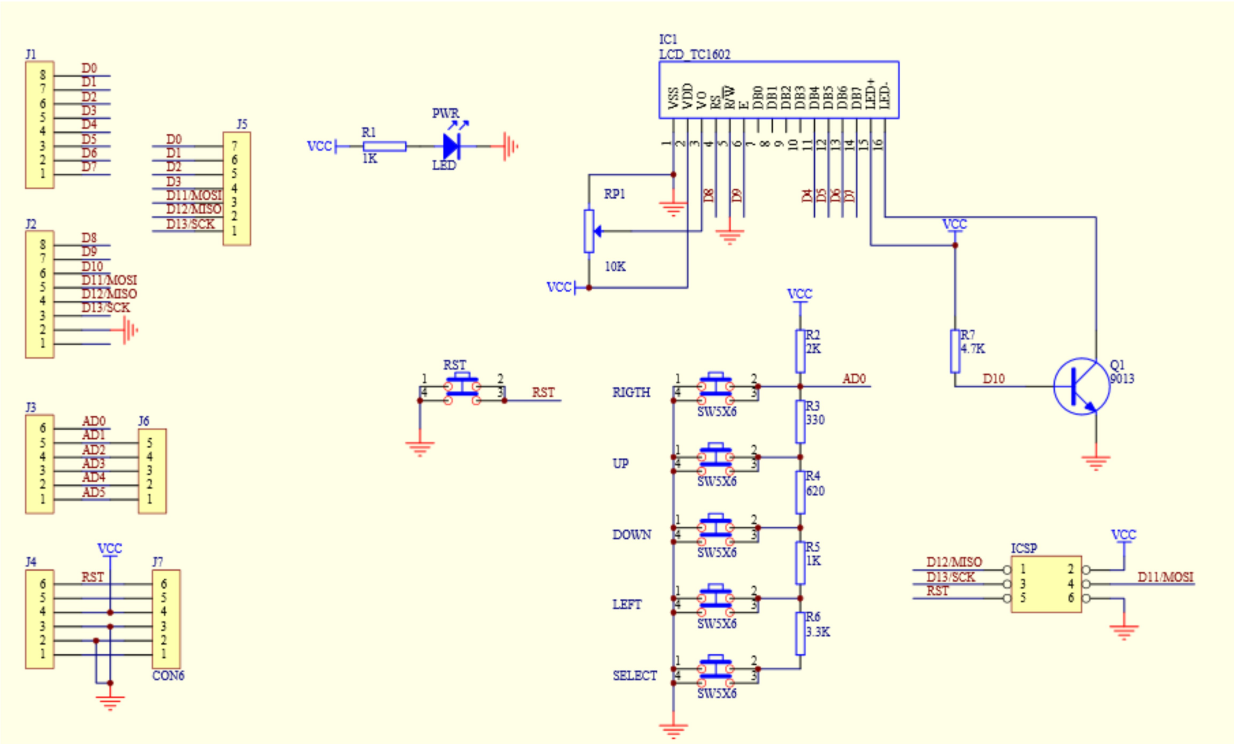
inverters			
Potentiometer cap knobs	2	$1.50	$3
Barrier Block Speaker Terminals	4	$2.24	$8.96
(Machine Screws and Nuts)	16	-	$2.36
(Wood Screws and Nuts)	8	-	$2.36
Wood Screws	16	-	$2.36
L Brackets	4	-	$2.87
Peg Board for case	-	$8	$8
1 ½ Inch thick board (Size)	1	Reused	-
Mounting Tape	1	$10	$10
Pyle PP444 Mini Phono Preamp	1	$25	$25
TOTAL COST	-	-	$249.70

Appendix B: Schematics

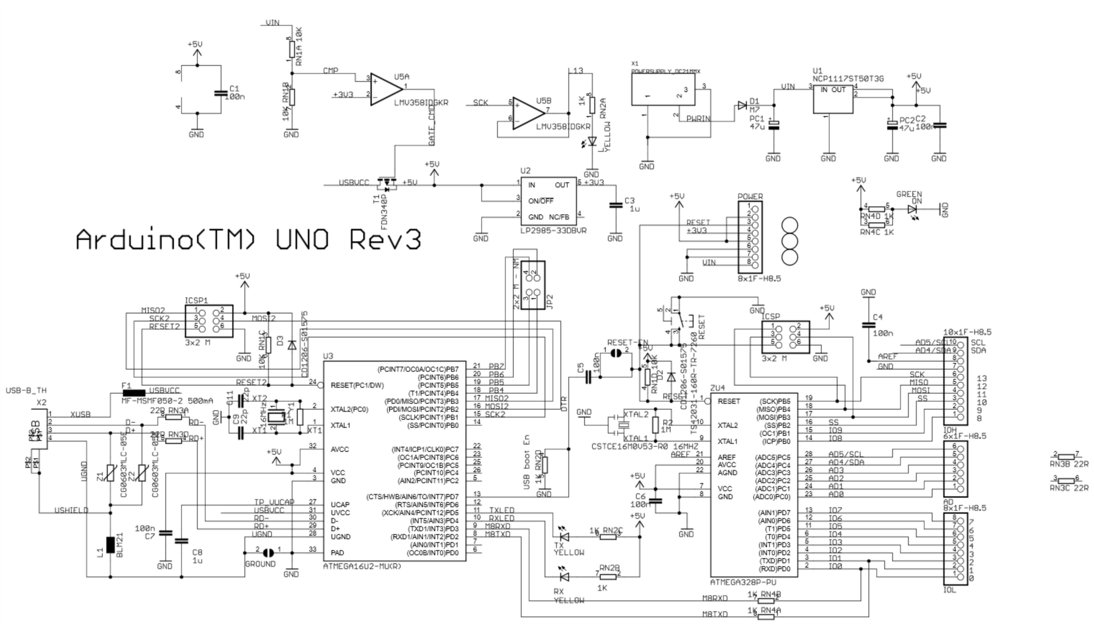
Class D Board Schematic



LCD Shield Schematic



Arduino Uno Schematic



Appendix C: Arduino Code

```
/*Austin Fox
3/4/15
Dual Output Audio Receiver
This program controls 2 PGA2311 stereo volume controllers and an LCD Display.
*/
#include <LiquidCrystal.h>
#include <SPI.h>
//#include <LCDKeypad.h>
//#include <LCD4Bit_mod.h>
//Initialize LCD Pins
LiquidCrystal lcd(8,9,4,5,6,7);

int cs1 =3;
int cs2 = 2;
int LEDctrl = 1;
//int mute = 8;

//initialize

//Initialize Input Pins
 int ch1in = 0;
 int ch2in = 0;
 int n1 = 0;
 int n2 = 0;


//The setup() function is called when a sketch starts. Use it to initialize variables, pin modes,
//start using libraries, etc. The setup function will only run once, after each powerup or reset of the
Arduino board.
void setup()
{
   //pinMode(mute, OUTPUT);
   //digitalWrite(mute, LOW);


   SPI.begin();
   lcd.begin(7,2);
   lcd.clear();//screen cleared, cursor at 0,0
   lcd.print("  CH1     CH2");
```

```
  //SPI.setClockDivider(divider);
  /*
divider:
SPI_CLOCK_DIV2
SPI_CLOCK_DIV4
SPI_CLOCK_DIV8
SPI_CLOCK_DIV16
SPI_CLOCK_DIV32
SPI_CLOCK_DIV64
SPI_CLOCK_DIV128
*/
 SPI.setDataMode(SPI_MODE0); //Clock Idles low and rising edge triggered
 SPI.setBitOrder(MSBFIRST);
 pinMode(cs1,OUTPUT);
 pinMode(cs2,OUTPUT);
 pinMode(LEDctrl,OUTPUT);
 digitalWrite(cs1, HIGH);
 digitalWrite(cs2, HIGH);
 digitalWrite(LEDctrl, LOW);
 //digitalWrite(mute, HIGH);
}

int aquirech1(void)
{
 //ch1in = analogRead(A2)/4.01176; //full range
 ch1in = 50 + analogRead(A2)/6.01765; //gives -31.4dB to 18.6dB with board gain added
 //reads input on pin A2
 //Note analogRead returns an integer value will be between 0 and 1023 with 1023 corresponding to 5V
 //The 5.01471 comes from the scale of the PGA values between 0 and 100 with an offset of 52dB
 //This gives dB values between -44dB and 6dB
 return ch1in;
}

int aquirech2(void)
{
 //ch2in = analogRead(A1)/4.01176; //full range
 ch2in = 50 + analogRead(A1)/6.01765; //gives -31.4dB to 18.6dB with board gain added
 //reads input on pin A1
 //Note analogRead returns an integer value will be between 0 and 1023 with 1023 corresponding to 5V
 //The 10.23 comes from the scale of the PGA values between 0 and 204 (-96dB -> +6dB)
```

```
  //integer values between 52 and 152
  return ch2in;
}


//This Function sends the two volume levels to the screen and displays them
void senddisplay(int ch1, int ch2)
{
  lcd.setCursor(0,1);
  if(ch1>215)
    lcd.print("MAX");
  else
  {

    lcd.setCursor(0,1);
    //lcd.print(ch1);
    lcd.print(double(38.6+31.5-(.5*(255-ch1))));
    lcd.print("dB");
  }

  lcd.setCursor(8,1);
   if(ch2>215)
     lcd.print("MAX");
   else
   {
     lcd.setCursor(8,1);
     //lcd.print(ch2);
     lcd.print(double(38.6+31.5-(.5*(255-ch2))));
     lcd.print("dB");
   }
}


//After creating a setup() function, which initializes and sets the initial values,
//the loop() function does precisely what its name suggests, and loops consecutively,
//allowing your program to change and respond. Use it to actively control the Arduino board.
void loop()
{
  //LED CTRL

 //value between 0 and 1023
```

```
  if(analogRead(A4) >800)
    {
     lcd.noDisplay();
     digitalWrite(LEDctrl, LOW);
    }
  else
  {
    lcd.display();
    digitalWrite(LEDctrl, HIGH);
  }
  delayMicroseconds(10);
  //Channel 1
  n1 = aquirech1();
  digitalWrite(cs1,LOW);
  SPI.transfer(byte(n1));
  SPI.transfer(byte(n1));
  digitalWrite(cs1,HIGH);
  //Channel 2
  n2 = aquirech2();
  digitalWrite(cs2,LOW);
  SPI.transfer(byte(n2));
  SPI.transfer(byte(n2));
  digitalWrite(cs2,HIGH);

  senddisplay(n1,n2);

  delayMicroseconds(10);
}
```