Analog Sound System
14:332:468 Capstone Design Electronics

by

Andrew Palin: dpalin5@gmail.com
Scott Durant: dscottdurant@gmail.com
Ed Sung: Kingdabeen@gmail.com
Pavan Desai: pavand717@gmail.com

Advisor: Prof. Dr. Sigrid McAfee
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Abstract:

Ever since the world went digital there has been a continuous debate on whether audio signals should be amplified and recorded using digital or analog techniques. Some prefer digital because of the absence of noise and low power consumption. While others prefer analog because of warmth and integrity of the sound. The digital recording and amplification process involves digital to analog converters as well as analog to digital converters. This process employs sampling which leads to a loss of signal and information.

The loss of signal caused by sampling, in part, is a problem we set out to solve. The difference may not be enough to get the population to throw out their iPod’s and other MP3 player devices, though it may give others a greater respect for technology of the past and a greater appreciation for music.

Our system implements an analog sound system with a series of equalizers which will individually control the amplification over separate given frequency ranges through the use of filters. The overall frequency range of our system spans the same range to that of human hearing, 20 Hz to 20 kHz. Many sound systems on the market advertise similar ranges, though the sound quality is not always equal. Impedance matching between the amplifier circuits and the speakers can make the difference between a low-end and a high-end model. By impedance matching our system has the ability to reduce any noise at the output, only displaying a crisp, clear signal to be heard.
What started from a love for music began a design for preserving the highest quality of sound.

**Introduction/Overview:**

There are certain specifications all sound systems required. These specifications include amplifiers to increase the amplitude of the signal, filter design, and impedance matching to increase the quality of the sound. The amplifiers used in the system for the power amplifiers was the class AB amplifier depicted in figure 1.

![Figure 1 depicts a Class AB amplifier design.](image)

Class AB power amplifiers were used in the output stage of each of the amplifiers for the system. We decided to go with a class AB, because there are many advantages over a class A, B,
and D amplifiers (digital). Class A has the potential for higher sound quality since it is always running at max current and has the most linear amplification. This will cause there to be less distortion at the output. However, since it is always running even when no signal is applied class A amplifiers tend to dissipate more heat than that of Class AB amplifiers. Also class A amplifiers are much less power efficient and consumes power even when the signal is not applied. A class AB amplifier can idle (lower current) causing less strain on the system than that of a class A. Class B are even more power efficient than Class AB however are very nonlinear due to crossover distortion. Class D amplifiers are the most power efficient yet they employ sampling which seriously degrades the sound quality.

The volume control circuits uses operational amplifiers and were extremely easy to design and implement. The circuit is in the figure below

![Adjustable Gain](image)

For use of filter design the individual frequency ranges that were used were created using formulas for a second-order Butterworth band pass filter. A second-order Butterworth band pass filter for our design consist of a second-order low pass, cascaded with a second-order high pass filter.
Figure 2: Left we have a 2\textsuperscript{nd} order low pass Butterworth filter and right is a 2\textsuperscript{nd} order high pass filter.

Above shows the basic design for our Butterworth filter design. The output of the low pass filter is simply cascaded to the input of the high pass filter. The property ideal properties of op-amps are convenient since the load is isolated from the circuit.

The last key concept implemented in the sound system design was impedance matching. That is the impedance from our circuit and that of the speakers had to be matched. This allows for maximum power transfer and an overall higher quality of sound. We designed the output impedance of the amplifier to match the impedance of the speaker.

By using the designs, the ‘Analog Sound System’ created was able to be designed to meet all necessary specifications, as well as a good output sound quality intended for this system.

**Goal:**

The digitalization that took over the industry in modern times motivated us upon our concept and ultimately to launch our project. In hope to dethrone the digital process that fragments the signal, we began the creation of an analog sound system, which simply amplifies the input signal by stage circuit designs and outputs continuous, analog sound by using a non-digital process. Our sound system preserves the integrity of the analog sound, displaying clear distinction between pure analog signals from that of digital.
Scope of the Project:

Our analog sound system design mainly consists of four major parts: master amplifier, volume control circuits, frequency filter, and equalizer amplifiers. The master amplifier was designed by implementing a common emitter with resistive degeneration into a common collector. The filter circuit is cascaded to the master volume control circuit (inverting op-amp configuration). It produces a voltage gain of 4.2, providing the general amplification across the frequency range of human hearing, 20Hz-20kHz. The output of the master volume control is then split into six different channels which implements the equalization volume control circuit which is of the same form as the master volume control. The output of each volume control circuit is then fed into the filters. The filters employ a second order band pass Butterworth. There are six filters, each corresponding to an individual frequency range. Ranges are based upon music standards for bass, treble and mid-range. Lastly, the equalizer amp design employs a common emitter into two common collector followed by the class AB output stage, which is driven by the Wilson current source. The reason for adding the extra common emitter stage will be explained later. This amplifier has a very low output impedance near 8 ohms, which is the representation of the impedance of the speaker.

Technical approaches, Results, and Discussion:
Figure 3 shows the schematic of the bass range band pass Butterworth filter. All the filters were constructed employing the same filter design with different resistors and capacitor values, which are omitted without loss of generality. We built the filters using the breadboards donated by Rutgers University. The left operational amplifier is a low pass filter, cutting off any frequencies above 140 Hz. The right is a high pass filter that cuts off frequencies below 20 Hz. Figure 4 provides graphs of the resulting output of each filter with two delta functions that mark the 3 dB cut-off range.
In figure 4 the gain can be seen for the filter design from figure 3. The 3 dB cut-off on the left is just below 20 hertz and on the right corresponds right above 140 hertz. The average attenuation of these filters is approximately 1-4 decibels. Since the same logic is used for the other five filters, figure 5 provides a table of the remaining five filters used. Each pertaining to a common range for treble and mid frequency bands as they pertain to that of your average sound systems. The overlap of the frequencies is accommodate for any loss experienced.

<table>
<thead>
<tr>
<th>Frequency Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>120-400</td>
</tr>
<tr>
<td>380-2600</td>
</tr>
<tr>
<td>2500-5200</td>
</tr>
<tr>
<td>5100-12000</td>
</tr>
<tr>
<td>11800-20000</td>
</tr>
</tbody>
</table>

Figure 5: Ranges of all Butterworth filters in Hertz
This amplifier, from figure 6, is a rather simple design mainly used for voltage gain, the 10k resistor represents the input impedance of the master volume control circuit. The master amplifier employs a common emitter into a common collector. The AC voltage gain of this circuit is set relatively low at 5.2 V/V since we are limited by the 12 volts supplies and the power consumption of 84mW.
Figure 7 is the schematic for the power amp also knows as equalizer amp, which drives the speakers at a certain frequency range. The equalizer amp design employs a common emitter into two common collector followed by the class AB output stage. The amplifier has a very low impedance and the potentiometer controls the volume of the selected frequency range. Figure 8 below shows a maximum gain of approximately 252 V/V across the band of interest. It can be seen that the power amplifier and pre-amplifier have some common features. The first stage is used for voltage gain while the second stage is used as a high resistance buffer to make the gain of the first stage sufficiently high. The third stage is another buffer whose 26 ohm is used for maximum signal swing and low output impedance. The output stage power amplifier also has the
addition of the Wilson current source which is used to drive the output current of the Class AB amplifier. The max power output of our each speaker was around 8 Watts.

There are some features that needed to be taken into account with the power amplifier design. Since it has a much larger gain and a current source that raises the current of the output, this circuit could not be done on a bread board. The current at higher volumes can and did surpass 300 mA, which is above the boards limitations. Unlike the filters and master amplifier this portion of the project had to be soldered to make it functional.

**Individual contribution:**

We met frequently throughout this semester and worked on much of the project together. We all worked on the filter designs, this part was relatively simple. Also we were all included in the building of the power amplifiers. Since these were soldered and six amplifier circuits were required there was a lot of time and work in their assembly.

Pavan specifically worked on building and designing the filters. He was also the group
leader and the go to for our poster design and PowerPoint’s. Ed worked on the master amplifier design (our volume control) and was in charge of debugging circuits. Drew worked on the center tapped bridge rectifier design and voltage regulator. Since we did not implement the transformer, neither of these designs were implemented. He also worked on the power amp design and was the main writer for reports (everyone else in the group proof read). Scott designed the power amp, master amp, and implemented volume control.

Over the span of this semester we have all devoted a lot of time and effort in the creation of the ‘Analog Sound System’.

**Impact of Work:**

Practice makes perfect when it comes to becoming good at anything in life, and engineering is no different. Through creativity and design work on a system such as ours allows for a huge learning experience. Our design has its flaws, but through these flaws we learned ways to improve upon our system. There is a lot of specification and knowledge needed to work on such project. One must balance power with efficiency, especially in the case of an analog system as there is more power consumption versus its digital counterpart.

By finally researching and creating a complex system of our own, we have gained a great insight of how many small concepts involving electrical engineering can be used to create a great and extremely useful product. This will help us lead the way in our careers to understanding even more complex electronic systems.

**Conclusion:**
We set out to create an analog sound system to preserve the clear, crisp sound analog provides versus that of digital. Since digital is sampled some of the signal is lost and therefore a small amount of information is lost in the process. In our goal to create a quality analog sound system we first had to find a way to properly approach the design. This was done by first figuring out the amplifier type to use, class AB in our case. Next was designing a circuit to properly filter the signal into separate bandwidths. This was implemented by combining a second-order Butterworth low-pass with a second-order Butterworth high-pass filter. The last important step involved creating a circuit purely for matching the circuits’ output impedance with the input impedance of the speakers. This being the basic concept of the design.

Although we make all of this sound relatively simple, there were some mistakes made and redesigns to fix the mistakes. However not all of our mistakes were made from design error. Although our system had decent sound quality, it has not met the standard we set out to reach. Breadboards were used for the filters and master amplifier, this has caused us a few issues. Breadboards, at least the ones we used, have unreliable connections and results. We soldered the equalizer amplifiers and when the master amp and filters are bypassed the quality of the output is greatly increased. Another factor that degrades the overall quality of the sound signal is in part from the wires we used. In pursuit to keep the cost down, we used wires provided by the University of Rutgers electrical engineering labs. These wires were not up to our standard of quality for the system and with many kinks and bends in the wire there was an increasing possibility for bad connections.

Despite some difficulties we were able to work together on a large project that allowed for the birth of our ‘Analog Sound System’ with the budget we had. So far we have a system
with decent sound quality, however there is still work to do on the system to increase the overall quality of the output.

**Future Work:**

Small mentions were made in the presentations about future fixes and tweaks to the sound system, but there are few more details to discuss. As mentioned better quality of wire will be crucial to truly perfect the system. This does not just include the type of wire used, we also have plans to make the circuit a bit cleaner. In doing so we can assure there are no faulty connections and it will also make debugging quite a bit easier. Also to improve connections we intend for every circuit to be soldered, this is something we should have considered from stage 1.

The sound system design was also missing something important in making this product easy to use. That is a center tapped transformer that would power all of our circuits. By implementing a centered tapped transformer, we would be able to step down the voltage from an outlet to create the positive and negative voltages needed to power the systems transistors and operational amplifiers. This is also much more portable then lugging around two giant DC power supply machines as was done. We were unable to purchase the power transformer that fitted our required specs because they are out of stock. The circuit also needs heat sinks, we are unable to order those do to our limited budget. It would be hard to sell a system that could only play music for about 15 minutes before having to turn it off for fear of smoking or even a fire.

Besides the internal changes we must also look at outwards. Some additions must be made, for instance having a sleek and attractive casing for it. Also having actual knobs for volume control and having each circuit locked in place will cut down the risk of connections becoming faulty.
Even with all the work put into the system there are so many ways for us to continue to improve upon it. We did the best we can with what we had.

**Cost/Sustainability:**

During the creation of the project there were some costs sustained from purchasing the materials. Fortunately, much of what we used was given to us by Steve Orbine, the tables below show the prices of the parts given to us and of what was actually paid for, as well as what will be needed in the future for improvement.

**Parts Donated:**

- Tip 41/42………………………………….$0.67
- Female Auxiliary Cable…………………..$4.99
- Various Resistors…………………..$0.05-$0.60
- Various Power Resistors………………..$0.85
- Various Capacitors…………………..$0.10-$0.85
- Variable Potentiometer……………….$1.55
- Various Lab Testing Equipment……………N/A

**Parts Purchased:**

- Subwoofer…………………………….$21.78
- 10x’s Full Range Speaker……………….$159.00
- Speaker Housing Supplies…………...$64.95

Total……………………………………..$245.74

**Future Parts:**

- Heat Sinks…………………………….$9.35
- Power Transformer……………………..$31.99
- Voltage Regulator……………………..$24.99
Despite much of the materials being provided for us, to reproduce and market our product even the materials provided to us have to be taken into account. To make this ready for market, a voltage regulator and a transformer are necessary. As the zener diode used in the regulator circuit has to work at voltages of around 20 volts, it proved to be rather expensive, approximately $15. Also the voltage regulator is another $25 minimum, this already increases the price $40. Making a proper speaker housing would also incur a cost that must be considered. All things considered it makes sense how high end systems can end up being in the thousands of dollars. If we were to market ours, taking into consideration all of the costs and time spent, not to mention work to improve upon it, I would estimate the system would cost between $500 and $600 dollars to complete. These estimates took into consideration the prices of various parts we researched throughout the project.

It is also important to look into the environmental impact of the project. There is nothing unusual as for as the materials go that would be especially bad for the environment. Some plastic and metal pieces would be required for the system, though no more than other sound systems. Since it is an analog sound system it will consume more power than that of a digital system. This is not significant however since it does not contain tube amplifiers. TIP41/42 were the transistors we used to make the amplifier and they consume relatively low power.

At this moment in time the system itself is not quite ready for market. Realistically a product like this can be sold to a niche market group. After some tweaks and fixes when the sound is as clear, crisp and perfect as we would like it to be, this could easily be sold as a mid to high range system. Since most would prefer to use digital the price would be higher on a product such as this make a profit in this niche market of ours.
References


