Class D Amplifier - EE 391

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SUMMARY: This paper discusses the class D audio amplifier. There are many classes of amplifiers, but the class D is known to be the most efficient. Here, we will talk about the theory, operation, and efficiency of the amplifier. Note: this is an adapted version of the final project of EE 101B, with a few modifications

1. Introduction

Audio amplifiers are ubiquitous, their applications ranging from broadcasting and media production, automotive systems, scientific equipment, or personal audio such as speakers. Due to their versatility and necessity, people have come up with many creative solutions. Specifically, there are several classes of audio amplifiers. Some of the most popular include Class A, Class AB, and Class D audio amplifiers.

The Class A amplifier is composed of a transistor operating in linear mode. The audio signal is sent to the gate, and the load is placed at the collector of the transistor. The audio quality of the output is relatively good, with a low total harmonic distortion (THD). The efficiency is quite low, however, around 30 - 40 percent [2].

The class AB amplifier is a compromise between audio quality and efficiency. It essentially operates the same way as a Class A, but with a 50-degree conduction cycle. This amplifier has slightly less linearity than a class A, but it is slightly more efficient [3].

The class D amplifier is known to be the most efficient. Invented in the 1950s, it modulates the input audio signal onto a PWM wave, and sends the PWM signal through a lowpass filter. In theory, this amplifier should have 100 percent efficiency, but it is slightly less due to the on resistance of the mosfets [4].

2. The Class D Amplifier

Figure 1 shows the general workflow of a class D amplifier. The audio signal is fed into one input of the comparator, and the other input is a very high-frequency triangle wave. The comparator outputs a PWM signal, which is essentially a pulse train modulated with the audio signal. Then, the signal is sent through a half-bridge, amplifying the PWM signal so that it spans between 0V and the power supply. Afterward, the signal goes through a lowpass filter, cutting off the higher-order harmonics that compose the PWM signal. The MOSFETs here are used as switches instead of operating in linear mode, so the efficiency is ideally 100 percent [4].

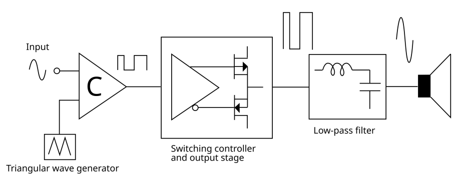

Fig. 1: This is the class D amplifier [1]

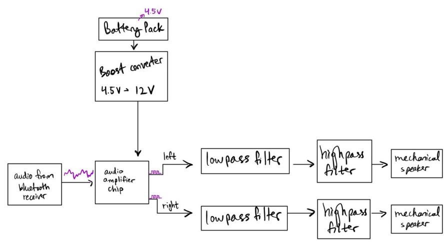

Fig. 2: This is the system block diagram.

3. Block Diagram, Schematic and Layout

Figure 2 shows the block diagram of the system. The power module involves a battery pack with 3 double A batteries (4.5V) and a boost converter (PTN04050c) that steps the voltage up to 12V. The audio amplifier chip used here is the TPA3122D2 and includes the comparator and output mosfet stage as

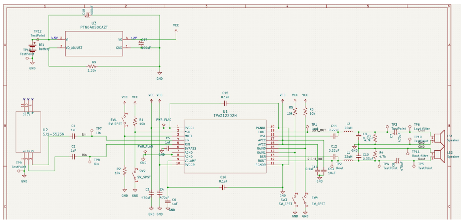

Fig. 3: Schematic

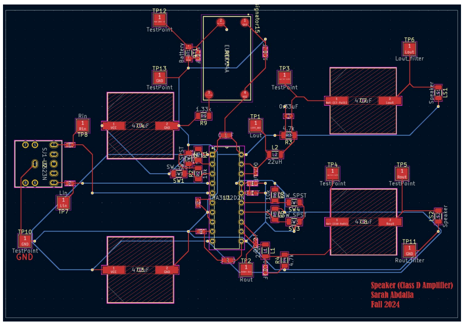

Fig. 4: Layout

built-in features. The audio signal is captured using a commercial, off-the-shelf Bluetooth receiver. The TPA3122D2 datasheet recommends using a 330nF capacitor and a 20uH inductor for the lowpass filters. This provides a cut-off frequency of approximately 60kHz, based on the following formula:

$$f = \frac{1}{2\pi\sqrt{LC}} \tag{1}$$

The PWM wave comprises the baseline audio signal's frequency and extremely high-frequency signal from switching elements, far greater than 60kHz. So the filter successfully eliminates the unnecessary harmonics. Also, since the filter is composed of an inductor and capacitor, it has two poles. So, the drop-off rate of the filter is about -40dB/decade.

DC signals should not be fed into the mechanical speakers, so a DC-blocking capacitor is placed right after the lowpass filter. It is approximately 470uF.

Figure 3 and 4 show the schematic and layout of the board. The schematic design is mostly adapted from the TPA3122D2 data sheet, with a few modifications to account for a new power system. The speakers were modeled as 8 ohm resistors. KiCAD was used to sketch the schematic and layout. The board has two layers, and was fabricated by JLCPCB. The total cost of the project, excluding the fabrication cost, was approximately 60 dollars, with the most expensive components being the mechanical speakers themselves (each costs 10 dollars).

4. Performance Evaluation

The board takes 4.5V as input and draws roughly 100mA of current. The TPA3122D2 provides four different gain options which can be configured with switches. This board was configured with half the maximum possible gain, providing very effective amplification. To evaluate the amplifier's performance, total harmonic distortion and efficiency were measured, and a frequency sweep was performed to generate a bode plot, evaluating the filter's effectiveness.

4.1. Total Harmonic Distortion (THD)

THD is a metric that indicates how linear a signal is. The more distortion there is from higher-order harmonics, the less linearity the signal has, and the higher the THD is. The lower the THD is, the better the audio quality. THD is calculated with the following formula:

$$\text{THD} = \frac{\sqrt{V_2^2 + V_3^2 + \dots}}{V_1} \tag{2}$$

where Vn is the RMS voltage of the n-th harmonic.

To measure THD, the 8-ohm speakers were replaced with 10 ohm resistors. A 500mV (peak to peak) sine wave was fed into the input channel of the TPA3122D2 at 1kHz. The output signal (measured with an oscilloscope, probed at the 10-ohm resistor node) was a 10.9V (peak to peak) sine wave. The FFT of the output signal was captured, and the RMS voltage of each harmonic was found (up to the 10th harmonic).

The calculated THD was 76 percent. This is a large distortion, but most of it is coming from the second harmonic. All other harmonics are attenuated, on the order of -40 to -60 dB. The most likely cause for such a significant second harmonic is ripple in the 12V power rail. A boost converter is used to create the 12V rail, so the internal switching elements are likely contributing to PWM signal generation (since the signal spans 0 to Vcc), and thus showing up in the output audio signal.

4.2. Bode Plot of Filter

A bode plot was created to evaluate the effectiveness of the lowpass filter and DC blocking capacitor. An input signal of 100mV (peak to peak) was fed into the TPA3122D2 input channel and swept through a range of frequencies, spanning 10Hz to 100kHz. The output signal was measured with an oscilloscope probe across the same 10-ohm resistor as the previous test setup. Figure 5 shows the results of the measurements.

The frequency ranges between 10Hz and 100kHz. There is significant attenuation at 10Hz, which is expected due to the DC blocking capacitor. For frequencies in the audible range, around 20Hz to 20kHz [4], there is ample gain, nearly 12dB. Then, after 50kHz, there's a significant drop of, which is expected due to the 60kHz cut-off frequency of the filter. The drop-off is very steep, due to the two poles creating a -40dB/decade drop-off rate.

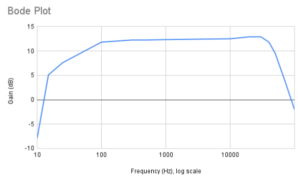

Fig. 5: Bode Plot of LC Lowpass Filter (with a DC blocking capacitor)

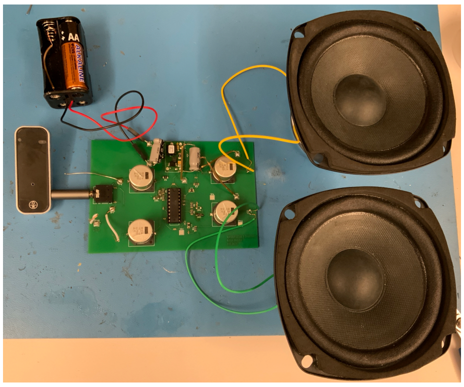

Fig. 6: Image of Board with speakers

4.3. Efficiency

Lastly, the efficiency of the audio amplifier was calculated. To find efficiency, calculate the power delivered to the load and divide it by the power provided by the supply. The board takes in 4.5V from the battery pack, and it draws 292mA of current for an audio signal of 200mV (peak to peak) at 1kHz. So power from the supply is roughly 1.314W. The power seen by the load is:

$$\frac{V_{\text{peak}}^2}{2 \cdot R_{\text{load}}} \tag{3}$$

The load is 10 ohms, Vpeak of the output signal (measured across resistor) is 4.5V, or roughly 9V peak to peak. So the power seen by the load is roughly 1.0125W. Therefore, efficiency is roughly 77 percent.

5. Conclusion

Overall, the audio amplifier is very effective. It sufficiently amplifies audio and can handle audio with both very high and low frequencies. Some potential improvements to the board include addressing the ripple in the 12V rail. A solution for this could be a large capacitor to stabilize the ripple more, or simply using a larger battery pack to reach 12V, eliminating the need for a boost converter. Another potential improvement is delivering the audio signal to a woofer, midrange speaker, and tweeter instead of sending it to a single speaker. Woofers and tweeters are optimized to handle lower and higher frequencies, respectively. Woofers have large areas, so low frequencies can resonate more, and tweeters are smaller to accommodate high frequencies. This greatly enhances the audio quality.

6. Acknowledgments

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7. Citations

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