Class-D Power Amplifier
Senior Project

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I. INTRODUCTION

The goal of audio amplification in an audio system is to accurately reproduce and amplify the given input signals. The biggest obstacle to overcome is to have high output power with as least amount of power loss as possible. In today’s technologies portable music devices are becoming more popular with a growing demand of external sounds in portable music devices. If the audio amplifier is not efficient it will deplete the device’s battery much sooner than desired. Audio amplification is sometimes done with tube amplifier technology but these are bulky in size and not desirable for portable electronics. For most audio amplification needs, engineers choose to use transistors in linear mode to create a scaled output based on a small input. This is not the best design for audio amplifiers because transistors in linear operation will continuously conduct, generate heat, and consume power. This heat loss is the main reason why linear mode is not optimal for battery operated portable audio applications.

Human audible frequencies range from about 20 Hz to 20 kHz, but many people cannot hear near the extremes. This range requires that any audio amplifier have a good frequency response over the human audible range. To accurately reproducing the audio input there should be no added harmonics, clipping or aliasing of the original input signal in the amplified output. Some harmonics can be reduced or eliminated by applying a low-pass filter before connecting the load. The way that we are going to amplify a small audio input signal is with a Class D power amplifier.
II. BACKGROUND

There are many classes of audio amplifiers; A, B, AB, C, D, E and F. The different classes are broken down into two different operating modes linear or switching. Class A, B, AB and C are all linear mode amplifier that have an output that is proportional to their input. Linear mod amplifiers do not saturate, fully turn on or fully turn off. Since the transistors are always conducting, heat is generated and continuously consuming power. This is the reason why linear amplifiers have lower efficiency when compared to switching amplifiers.

Switching amplifiers are Class D, E and F. Switching amplifiers have higher efficiency which theoretically should be 100%. This is because there is little to no energy being loss to heat dissipation. Class D is a switching amplifier and when it is in the “on” state it will conduct current but have almost no voltage across the switches, therefore no heat is dissipated due to power consumption. When it is in the “off” mode the supply voltage will be going across the MOSFETs but due to no current flow the switch is not consuming any power. The amplifier will only consume power during the on/off transitions if leakage currents are not taken into account.

The Class A amplifiers topology is shown in figure 1. This topology uses one transistor as a dc current source configuration. This source is responsible for supplying as much audio current as is required by the speaker. The quality of amplification and sound reproduction is good but the power that is dissipated is excessive because of the large dc bias current that is flowing in the output-stage of the transistors and not to the speakers. Theoretically the maximum efficiency for a Class A amplifier is 50%.
Figure 1: Class A Amplifier Topology

Figure 2 depicts the Class B amplifier topology. Compared to Class A amplifiers, the Class B amplifiers do not have the dc bias current and therefore consume considerably less power. The output transistors in Class B topology are controlled individually. That is when the high MOSFET is on it supplies positive current to the speaker and when the low MOSFET is on it tries to sink negative currents. This is why Class B amplifiers dissipate less power but in trade off with power we get mediocre sound quality. Theoretically, the maximum efficiency for a Class B amplifier is about 78.5%.

Figure 2: Class B Amplifier Topology
Class AB amplifiers topology is shown in figure 3. It's a hybrid of both Classes A and B. It uses some DC bias current, less than that of the Class A design. The power dissipation is typically closer to Class B but still not good enough for audio. This is because the midrange output voltages are too far from either the positive or negative supply rails, so the transistors are always conducting. There also needs to be some sort of control to allow the Class AB circuit to supply or sink such large output currents.

![Class AB Amplifier Topology](image)

Class C amplifiers, topology shown in figure 4, have zero idle bias current at the output, which makes them similar to Class B. What makes Class C different from Class B is that Class C has an area where the idle current is zero, which is more than 50% of the total supply voltage. Therefore there is even more distortion caused by crossover of source and sink currents then in Class B. This distortion causes poor sound quality and makes Class C amplifiers inadequate for audio applications. Theoretically the maximum efficiency for a Class C amplifier is 100% at zero output but 78.5% at full output.
Figure 4: Class C Amplifier Topology

Class D amplifiers operate by switching between two states, “on” and “off”. The “On” state is when there is current flowing through the device but, theoretically, no voltage across the MOSFET. The “Off” state occurs when there is a voltage across the MOSFET but no current flowing. There are two main topologies in Class D amplifiers, half-bridge and full-bridge. Both are depicted in figures 5 and 6. Class D amplifiers are the best choice when it comes to power dissipation consideration because there is far less loss in power consumption when the MOSFETs are not always in the “on” state. Theoretically the maximum efficiency for a Class D amplifier is 100%.

Figure 5: Class D Half-Bridge Topology [4]
Figure 6: Class D Full-Bridge Topology [4]

The topology for a Class E is shown in figure 7. This class of amplifiers has a pulsed input and an output that is tuned to a certain frequency. They are commonly used in radio transmitters where the output is a single or narrow band of frequencies. This is undesirable for audio applications because there is such a wide range in audible frequencies. Therefore we would not be able to get the range of audible frequencies using a Class E amplifier. Theoretically the maximum efficiency for a Class E amplifier is 100%.
Class F amplifiers are used mainly in high frequency applications. A common Class F topology is shown in figure 8. It is similar to the class C except that the Class F has two or more circuits that are tuned. A Class F amplifier transmits below a certain cutoff frequency and reflects above the cutoff frequency. There is only one transistor in Class F topology and two load networks. This topology causes harmonics to be replicated and is not suitable for accurate audio reproduction. Theoretically the maximum efficiency for a Class F amplifier is 100%.
III. REQUIREMENTS

a) Triangle Wave Generator

In a Class-D amplifier design an accurate triangle wave signal is required to achieve the pulse width signals. The generation of the triangle wave can be done with the use of simple circuits which consist of a square wave generator and an integrator which integrates the square wave to a triangle wave. Most of these circuits consist of comparators and op-amps that can handle the required frequency and not exceed their respective slew rates. More in-depth explanation on the chosen circuit for the triangle wave generator is in the design part of the report. The frequency of the triangle wave must be at least three times the switching frequency. Having the triangle wave frequency at least two or three times greater will minimize distortion from harmonics [3]. The output filters cutoff frequency plays an important role in choosing the triangle wave frequency due to harmonic distortion that is generated from the switching of the amplifier.

b) Pulse Width Modulation

Class-D amplifiers use pulse width modulation (PWM) to sample the audio signal. With the use of a half-bridge or full-bridge design it amplifies the sampled pulse train wave. The choice of sampling frequency is very critical to have an accurate representation of the sampled audio signal. The audible audio signal frequency is from 20 Hz to 20 kHz. Knowing this we have to choose a sample frequency that is at least twice that of the audio signal due to the Nyquist Theorem. For low distortion a factor of 5 to 50 times greater is used but due to chosen components the sampling frequency was chosen to be 200 kHz. One thing to keep in mind is that as the frequency is increased so are the losses in the MOSFET’s, due to their gate
capacitance. The trade-off is that the signal to noise ratio is reduced along with reducing the output filtering requirements. Figure 9 shows the block diagram of how the PWM signal is generated. The comparator is generating the PWM waveform; this is accomplished by comparing the amplitudes of the triangle wave and the audio signal. As shown in figure 10 a positive pulse is generated when the audio signal’s magnitude is greater than the triangle wave; a negative pulse is generated when the triangle wave’s amplitude is greater than the audio signal’s amplitude. The chosen amplitude is an important factor that determines whether the compared signal will be linear, over or under modulated.

The amplitude modulation index is calculated from the ratio of the audio signal amplitude to the triangle wave amplitude. A modulation index between the values of \([0,1]\) will ensure linear modulation while an index greater than 1 will yield over modulation and saturation and an index greater than 3.24 yields a square wave modulation [6]. Figure 11 shows the three different regions of modulation according to amplitude modulation index. The PWM signal and its compliment is what will drive the gates of the MOSFET’s in the bridge configuration to control the signal the speaker receives.

**Figure 9: PWM Circuit [2]**
c) **MOSFET’s**

It’s very crucial to the design that the driver circuits produce signal that do not overlap or else you run into the problem of shorting your supply straight to ground.
or if using split supply shorting the supplies. This is better known as shoot through but it can be reduced or prevented by introducing non-overlapping gate signals to the MOSFETs. The non-overlapping time is better known as Dead time. In designing these signals we must keep the dead time as short as possible to maintain an accurate low-distortion output signal but must be long enough to maintain both MOSFETs from conducting at the same time. The time that the MOSFETs are in linear mode must also be reduced which will help ensure that the MOSFETs are working synchronously rather than both conducting at the same time. For this application Power MOSFETs must be used due to the voltage and currents in the design. The Class-D amplifiers are used for their high efficiency but MOSFETs have a built in body diode that is parasitic and will allow the current to continue to freewheel during dead time. A Schottky diode can be added in parallel to the drain and source of the MOSFET to reduce the losses through the MOSFET. This reduces its losses because the Schottky diode is faster than the body diode of the MOSFET ensuring that the body diode does not conduct during dead time. To reduce the losses due to high frequency a Schottky diode in parallel with the MOSFET is practical and necessary. This Schottky ensure that the voltage across the MOFETs before turning off. The overall operation of the MOSFETs and output stage is analogous to the operation of a synchronous Buck converter.

d) Output Filter

The final stage of a Class-D is the output filter which attenuates and removes the harmonics of the switching frequency. This can be done with a common low pass filter topology but the most common is an inductor and capacitor combination. A 2\textsuperscript{nd} order filter is desired so that we have a -40dB/Decade roll-off. The range of cutoff frequencies is between 20 kHz to about 50 kHz due to the fact that humans cannot hear anything above 20 kHz. Figure 12 shows the second order Butterworth filter, which is what we chose to design and implement. The main reason we choose a
Butterworth filter is because it requires the least amount of components and has a flat response with a sharp cut off frequency.

![Second Order Butterworth Low pass filter](image)

**Figure 12: Second Order Butterworth Low pass filter**

The transfer functions of the Butterworth along with component design equation are as follows:

\[
H(S) = \frac{1}{S^2 + \sqrt{2}S + 1}
\]

\[
H(S) = \frac{(\omega c)^2}{(S^2 + \omega c\sqrt{2} + \omega c^2)}
\]

The design equation for the required components are chosen from the below equations.

\[
\omega c^2 = \frac{1}{LC} \quad C = \frac{1}{2\sqrt{2}\pi f c R} \quad L = \frac{\sqrt{2}R}{2\pi f c}
\]

In the above equations, \(f_c\) is the cutoff frequency chosen by designer, \(L\) is the inductor value, \(C\) is the capacitor value, and \(R\) is the resistive value of the speaker usually 4 or 8 ohms. The inductor and capacitor tolerances must be chosen to withstand the high output voltage and current ratings. There are many different types of inductors commercially available but the most commonly used are gapped ferrite cores and powder alloy cores. They each have their respective drawback and...
advantages. Ferrite core inductors have an almost linear or flat saturation response while powder alloy cores do not. Figure 13 shows the saturation curves for both the powder alloy and ferrite core. Choosing the correct type of inductor is important so that it does not saturate its core.

An electrolytic capacitor at the output is usually desired due to the fact that electrolytic capacitors are cheap and physically small. The choice of dialectic class is more important to pay attention to. A class 1 dielectric COG capacitor would be ideal due to the fact that it will have a very stable temperature coefficient and make it a more suitable for a Class-D amplifier application.

![Figure 13: Ferrite Core vs Powder Alloy Core Saturation Curves [1]](image-url)
IV. DESIGN

a) Triangle Wave Generator

The first attempt at designing a triangle wave, shown in figure 14, was using a 555-timer to create a square wave with 50% duty cycle and a frequency of 200 kHz. This was very difficult but we were able to get very close to 50%. Then we built an integrator and put it on the output of the 555-timer. This was so that we could integrate the square wave into a triangle wave at 200 kHz. We were unable to get this to work properly. When we interfaced the two devices the 555-timers ground lifted and the zero was too high for the integrator and we were unable to easily correct this, so we moved on to another triangle wave design from Design with Operational Amplifiers and Analog Integrated Circuits, 3rd edition by Sergio Franco.

![Figure 14: 555-Timer and Integrator, Triangle Wave Generator](image-url)
Our Second attempt at a triangle wave is shown below in figure 15. After constructing this we were able to see a square wave but were unable to see a triangle wave output. This did not make sense because if we were able to see a square wave we should be able to see the triangle wave at the output of the first operational amplifiers. The triangle wave is an input to the square wave and sets the frequency of the square wave. We were unsure why this was not able work.

Figure 15: Sergio Franco’s Triangle Wave Oscillator

Figure 16 is the circuit diagram of the Voltage controlled oscillator chosen to generate the required triangle wave. Choosing the correct values for the timing capacitor (C) and resistor (R) are determined by equations and figures from the datasheets. A precise 200 kHz was needed for our project and by using the XR-2206 we got a precise triangle wave at 200 kHz that is variable with a potentiometer. This method was much better than trying to design our own triangle wave. The only problem we ran into with this chip is that the chip outputted a DC off-set triangle wave. The audio signal required a level shift so that the triangle wave and audio signal would be centered at the same DC point.
b) Pulse Width Modulation

Figure 17 depicts the circuit designed to generate the two non-overlapping signals. The first sets of comparators are used for pulse width modulation by comparing the triangle wave and the audio signal. The offset on one triangle wave is to introduce dead time between the pulses and therefore neither MOSFET is on at the same time. After the pulses we buffer and invert the signal so that it can be used to power the MOSFETs gate.
c) Output Stage

We figured out that the output voltage swing from the LM339 is sufficient to drive the MOSFETs. From that we were able to choose appropriate MOSFET’s, so that we did not have to build a circuit to drive the MOSFET’s. The output filter was the same second order Butterworth filter that was described previously in the requirements part of the report. The diagram for the MOSFET’s and the low-pass Butterworth filter are shown below in figure 18.

![Figure 17: PWM Comparators with Variable Dead Time](image1)

![Figure 18: Butterworth Filter with MOSFET’s](image2)
V. CONSTRUCTION

Figure 19 is the final constructed circuit but for complete pin out of final circuit refer to appendix a. This diagram includes the triangle wave generator, PWM circuit, MOSFET drivers and output filter. After working through all of our difficulties in the design stages we had a fairly simple time constructing the circuit. The biggest difficulty was choosing the appropriate components from the part available.

Figure 19: Complete Class D Schematic
VI. TESTING

a) Triangle Wave Generator

Figure 20 shows our triangle waveform captured from the oscilloscope. Our triangle wave came out to have a $V_{pp}$ of 7.76V and a frequency of 200 kHz. These values were very close to what we simulated and more than we needed. Our $V_{pp}$ only needed to be around 7V. When we tested the audio signal the voltage seemed to peak out around 3.5V.

![Triangle Waveform](image)

Figure 20: Triangle Waveform
b) Pulse Width Modulation

Figure 21 shows one of the pulse width modulated signal. When the triangle wave and the audio signal intersect a pulse is formed. Due to the delay of the operational amplifier the pulse is offset from where the triangle and audio signal intersect. The triangle in this figure is the actual triangle wave generated by the XR2206 at 200 kHz.

![Figure 21: Pulse Width Modulated Signal](image)

c) Pulse Width Modulation with Dead Time

To avoid having simultaneous conduction of both NMOS’s a non-overlapping pulse circuit had to be design with adjustable dead time. The non-overlapping pulses were generated by comparing the audio signal to the generated triangle wave and also to a DC offset triangle wave, which would introduce the desired dead time by increasing the DC voltage. This gave us the two different pulses and has an adjustable dead time, by adjusting the DC offset we are able to change the dead
time. Below in figure 22 are our non-overlapping pulse signals. These two non-overlapping signals are what would be used to drive the MOSFET’s at the desired frequency.

![Figure 22: Pulse Width Modulation with Dead Time](image)

c) Output Filter Stage

In figure 23 the scope capture represents the frequency response of the designed output filter. The response was simulated by running a frequency sweep with a function generator starting from 20 Hz to about 20 kHz. As shown the cutoff frequency is at or around 20 kHz, which is around 1/6 of the original input. This is ok because most people cannot hear over 18 kHz because of this there is very little audio that will be playing at 20 kHz. From the oscilloscope capture and comparing the amplitude of the signal we can deduce that the designed and implemented Butterworth 2nd order Low pass filter has a cut off at around 20 kHz and is adequate for this application.
Figure 23: Low-Pass Filter Response
VII. CONCLUSIONS AND RECOMMENDATIONS

In doing this project we thought we were picking something that was less time consuming it seemed easy to just build and put this circuit together but we ran into a lot of trouble because of how precise our components needed to be. One thing that would have helped is doing more research into the scope of the project. In the last few months of the quarter we finally decided to look at older students previous senior projects and we saw that they were unable to complete them. Another suggestion is to maybe start with an integrated chip that generates the required triangle wave and then start the design of the circuit. As time passed the triangle wave generation is what took up most of the time and would have been something that could have been designed last. One thing that would have deceased time trouble shooting and design time is to do more research on the components required to make sure they meet the required specification. The comparators Slew rates are an important thing to know and make sure it’s sufficient for the required switching frequency. Lastly just like any project time is of the essence and the earlier you start the better. The MOSFET driving by the comparators output was not tested in this project but from the electrical specification and data sheet information the outputs generated by the LM339 should theoretically be sufficient to drive the MOSFETs.
VIII. BIBLIOGRAPHY

   

   

   
   <http://users.ece.gatech.edu/mleach/ece4435/f01/ClassD2.pdf>.

   

   


   
IX. APPENDICES

a) Schematic

b) Parts List and Cost

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### b) Time Schedule Allocation

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C) Picture of completed Prototype