Home theater system design

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Háskólinn í Reykjavík

5.12.2016 Amplifier, HDMI

Amplifier, HDMI

Magnari, HDMI

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lokud

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**date**

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Guðlaugur Ingi Marteinsson  
Bachelor of Science
I dedicate this to my family.
Acknowledgements

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Preface

This dissertation is original work by the author, Guðlaugur Ingi Marteinsson.
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List of Abbreviations

AC  Alternating current
ADC  Analog-to-Digital Converter
ARC  Audio return channel
BSc  Bachelor of Science
CEA  Consumer Electronics Association
CEC  Consumer Electronics Control
CMOS  Complementary metal–oxide–semiconductor
DAC  Digital-to-Analog Converter
DC  Direct current
DDC  Display Data Channel
EDID  Extended Display Identification Data
HDCP  High-bandwidth Digital Content Protection
HDMI  High-Definition Multimedia Interface
I2C  Inter-Integrated Circuit
I2S  Inter-IC Sound Bus
IC  Integrated circuit
IDE  Integrated development environment
LCD  Liquid-crystal display
Misc  Miscellaneous
MSB  Most Significant Bit
MSc  Masters of Science
PCB  Printed circuit board
PCM  Pulse-code modulation
PWM  Pulse-width modulation
RGB  Red, Green and Blue
S/PDIF  Sony/Philips Digital Interface Format
SCK  Continuous serial clock
SCL  Serial Clock Line
SD  Serial data
SDA  Serial Data Line
SMPS  switch-mode power supply
THD  Total harmonic distortion
THD+N  Total harmonic distortion + Noise
TV  Television
TWI  Two Wire Interface
VESA  Video Equipment Standards Association
VGA  Video Graphics Array
WS  Word select
## List of Symbols

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<td>Magnetic flux</td>
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<tr>
<td>(C^\circ/W)</td>
<td>Heat per watt</td>
<td>(C^\circ/W)</td>
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Chapter 1

Introduction

The goal of this project is to design and build a home theatre system, research different ways of extracting the audio signal and amplifying it. The system should be able to revive signal from either HDMI, optical, coaxial or analog stereo and convert it into surround sound. The system should also have buttons for control and an LCD screen to display information.

The design process will be based on the fact that evaluation boards can be bought from the producer or a third party in order to hasten the design process so measurements can be started as soon as possible.

The main problem that the system should be able to solve is to amplify the audio signal from a TV or any audio source that has small speakers into bigger speakers without decreasing the audio quality.

1.1 Research Goal

The goal of this project is to design and build a home theatre system and research different ways of extracting the audio signal. The system should be able to revive signal from either a HDMI, optical and analog stereo and convert it into surround sound. The system should also have buttons for control and an LCD screen to display information.

1.2 Research Tasks

- Research different ways of extracting audio signal for instance.
  - HDMI
  - Optical
  - Coaxial
  - Analog
- Research an exactable level for THD in a amplifier compared to what is on the market.
- Research different classes for amplifying audio signal and choosing the most efficient one. This is to minimise the cost of material and size of the design.
  - Class A

\(^1\)See appendix B.4 details on surround sound
– Class B
– Class AB
– Class D

• Research an exactable power level per channel for an amplifier compared to what is on the market.

• The system should be able to operate at ambient temperature, 25°C without the aid of a cooling fan. The cooling fan can in some cases be heard which is not good for a device intended for good audio quality.

• Find an acceptable display for the project.

• Build a home theatre system or a working prototype. In case there is only a prototype built than the PCB designs will be sent out to create professional PCBs.

• Take measurements from the audio path and compare the signals to datasheets and known values.

1.3 Main risks of the project

There are numerus risks involved when designing an building a home theatre system as are outlined here below.

• Time and running out of it.
  – unforeseen events. There are a number of unforeseen events that could happen for that reason a time plan is necessary in order to complete the project on time.

• Availability.
  – Not getting parts on time or not at all.
  – Parts available only in large quantities.

• Proof of concept
  – Instruments not accurate enough. In this project one of the goal is to measure a low THD signal and some instruments in the school may not have the capability.

1.4 Thesis time plan

Below is the time plan goal set for this thesis in order to complete the project in time. This was needed as one of the main risks discussed in chapter 1.2 is running out of time.

• Week 1 - 5. Research methods of extracting audio and amplifying them. This has to be finished as soon as possible so components can be ordered in time. Start the outline on the thesis.

• Week 6 - 10. Design circuits based on datasheets and start making PCBs. This has to be finished so measurements can begin. Continue write the thesis.

• Week 11 - 15. Finish measurements and finish writing the thesis.
1.5 Thesis Outline

The thesis start with going into the theoretical background in chapter 2 where different methods of audio extraction, amplifying, audio manipulation, power supplies, microprocessors and displays are researched.

The thesis will then go in to the design phase of the project in chapter 3. In appendix B the key terms and concepts are discussed in detail.

The thesis will then compare the measurements of gain, THD, heat, efficiency and known signals to the datasheets and concepts.
Chapter 2

Theoretical Background

In this chapter are the theoretical backgrounds on all the parts of the home theatre system design and choices made from them. For this a Pioneer amplifier VSX-830-K/-S will be used as a baseline for comparison.\[1\]

2.1 Purpose of a home theatre system

What are the benefits of using a home theatre system vs using the stock speakers that come with the TV? The speakers that come with the TV are small compared to the speakers that the home theatre system drives. Why does this matter? The size of a speaker determines how much air the speaker can move. This comes to play for bass sound as most people want to feel the bass coming from for instance an explosion in a movie rather than just hearing it.\[2\]

The size of a speaker also determines it’s bandwidth as by nature a big 15 inch speaker that has a bandwidth of 10-200 Hz, which is usually used for bass, cannot move as fast as a so called treble which is in the range of one to two inches and a bandwidth of 200Hz-20kHz.\[2\]

The home theatre system also delivers sound from all sides immersing the user deeper, see figure 2.1\[1\].

Does this make all media viewing better? No, there are some media like news from the TV which do not benefit much, as the need for big bass sound is not necessary from this system, however it will not make them worse.\[2\]

In short using a home theatre system brings the user closer to the intended media via better sound quality and deeper feel.

2.2 Design overview

Figure 2.2 shows a design overview of the home theatre system, this is a general design for a home theatre, there are ways of doing this differently as some IC may have for instance an audio input and an audio processor built in. The blue line shows the audio path. The green lines indicate connections of controls from the microcontroller. The red lines indicate connections to the power supply.

The audio source indicated can be any number of devices Blu-ray player or media station but in this case it is the digital audio is from a television. The audio input is the way for the

\[1\] See appendix B.4 for explanation on surround sound.
sound to enter the system so choosing this is a vital part as it can limit what devices can be connected. The ways of doing this is are described here below.

- **HDMI**: The HDMI is a wide spread standard most TVs and devices today use the HDMI standard for audio and video communication.[3]

- **Analog**: The analog signal is the oldest type of signal in this is also used in most devices.

- **Optical**: The optical signal is used on many home theatre systems today as a way to get digital audio from TVs in to the amplifier. It has the benefit of not being susceptible to electronic interference.

- **Coaxial**: While not as much used as the optical signal the circuitry needed for this are almost identical.

The LCD shown in figure 2.2 is used to display information to the user. The Microprocessor shown in figure 2.2 is the brain of the system. It receives all the error signals used in the system.
2.3 Choosing an input audio source

The choice of the input signal determines how useful the system will be on the market as a number of devices can only support a few of them so by limiting the system to only one reduces the value. However in this project it is important not to go overboard as time is limited.

2.3.1 HDMI

At the beginning of this project the possibility of using HDMI to extract the audio signal was researched. The VSX-830 has this ability and is thus a valid reason to research. There were two possibilities to choose from a HDMI receiver like ADV7611 or a HDMI transceiver like ADV7623. Both of these circuits are special circuits designed to handle all necessary protocols for using HDMI like EDID, CEC and HDCP.

2.3.1.1 ADV7611

The ADV7611 is a HDMI receiver meaning it has the possibility to receive HDMI signal and extract the audio and video signal using only a few external components, see figure 2.3.

Figure 2.3: Block diagram for CN0282 Evaluation board [3]

Figure 2.3 is a block diagram for the evaluation board CN0282 which uses the ADV7611. SSM2604 is a circuit designed to receive $I^2S$ signal \(^2\) and convert it to stereo signal. SSM2604 uses two 24-bit $\Sigma - \Delta$ ADCs and two $\Sigma - \Delta$ DACs \(^4\). The SSM2604 can either be used as a master or slave, in the case of CN0282 it is used as a slave as it gets its clock signal from the ADV7611. \(^3\)

---

\(^2\)See appendix B.1 for explanation on HDMI communication protocol, EDID, CEC and HDCP.
\(^3\)See appendix B.13 for $I^2S$
\(^4\)See appendix B.12 for ADCs and DACs
CHAPTER 2. THEORETICAL BACKGROUND

The ADV7125 is a triple 8-bit video converter. The ADV7125 uses tree $\Sigma - \Delta$ DACs, one for each of the RGB signals, each has its one output pin. The RGB signal is then fed to the VGA connector. The VGA gets its vertical and horizontal sink pulses from the ADV7611.[5]

The ADuC7020 is the microprocessor for the circuit. ADuC7020 is an ARM based microcontroller.[6]

![Figure 2.4: Block diagram for ADV7611 IC][7]

As can be seen in figure 2.4 the ADV7611 has a CEC controller and EDID repeater controller which are essential for the HDMI to work. As good as the ADV7611 is it has a severe limitation as the video signal is converted to RGB into a VGA connector which is not widely used in modern TV screens.

2.3.1.2 ADV7623

The flaws of the ADV7611 can be fixed by using ADV7623. The ADV7623 is a HDMI transceiver, meaning it can receive and transmit the HDMI signal. The key element of the ADV7623 is the fact that it can internally extract the audio signal via six audio outputs, see figure 2.5. The ADV7623 also has the ability to receive the audio signal from the TV via the ARC input. There are a number of extra features in this circuit and it would be perfect for this project however there is a flaw, the ADV7623 has no evaluation board and can only be ordered in large quantity.[8]

2.3.2 Analog

The analog signal is widely used however as the market goes more and more in to the digital age it become less and less used as the sound from the HDMI is not outputted from the TV via analog. However the VSX-830 uses this ability and it is a thus a valid contender for the audio input.[1] The analog signal can be taken from the SCART connector on a TV.

2.3.3 Optical and coaxial

The optical or TOSLINK and the coaxial signals both use PCM signals which can be fed in to a SPDIF Transceiver to change the signal in to a $I^2S$ which is then fed into a circuit designed to change that in to analog audio.[5] The pros of using optical vs coaxial is that the optical does

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5 See appendix B for information on $I^2S$ and PCM.
not pick up electromagnetic interference which can affect the audio quality. Both are widely used making them a great choice for this project. The VSX-830 uses this ability and it is a thus a valid contender for the audio input.[1]

2.3.4 Conclusion

In conclusion after viewing the pros and cons of these different methods it is clear that HDMI cannot be used as it does require licenses for use.\(^6\) The analog audio can be used but is limiting towards digital media leaving only the optical and coaxial which offer the best solution. The reason for building this amplifier instead of buying the VSX-830 is that the VSX-830 is a big design and has a lot of features that are not needed and tend to complicate things for people that are so not good with new technology.

2.4 Choosing an audio processor

When choosing an audio processor some requirements on what that processor should me able to perform need to be outlined as seen below.

- Receive audio data and convert to surround.
- Control Volume level.
- Control bass volume for speakers.
- Control bass volume for bass speaker.

\(^6\)See appendix B on HDMC.
• Control treble volume.
• Control balance.

2.4.1 Conclusion

As can be seen in chapter 2.3 the ADV7623 could have been an audio processor for this project however the HDMI option cancelled due to the fact that it cannot be used without a licence\(^7\).

Do to this a chip known as uPC1892 was chosen. The uPC1892 is a matrix surround sound processor. There are a lot of audio processors on the market however the uPC1892 can fulfil all of the requirements set for the audio processor while having a simple design for the circuitry.[9]

\(^7\)see appendix B.1
2.5 Choosing an amplifier

This section discusses the different classes used to amplify audio signal, their pros and cons and in the end an amplifier is chosen. After some research on what is on the market, like pioneer VSX-830, it was clear that the amplifier should have a THD lower than 1% at 1 kHz and be able to output power in the range of 100-200w. This will become the base for future reference.[1]

2.5.1 Class A amplifier

The class A amplifier was one of the first ones created. Its basic design uses one transistor to amplify the audio signal. In the class A amplifier the current is constantly running this is because the operating point is set at the half way point between the transistor being fully open and fully close as can be seen in figure 2.6.

The strength of this amplifier is that because the operating point is set at the halfway point it does not suffer from crossover distortion meaning the audio is less affected thus keeping audio quality high.

![Class A amplifier circuit and operating point](image)

Figure 2.6: Class A amplifier circuit and operating point [10]

The weakness of this design is also caused by the placement of the operating point, as the current is constantly running through the transistor it causes it to waste power on heat forcing the design to use larger heat sinks do dissipate that heat to keep that transistor working at optimal heat according to its specks.

The efficiency of the A Class amplifier is around 30% meaning that it is not a good candidate for this project as the extra power would be wasted on heat and not audio power.[10]

2.5.2 Class B amplifier

The B class amplifier was the next design of amplifiers. The reason for the design of this amplifier was to increase the efficiency of the A class amplifier. The way this is achieved is by using two transistors in a push pull configuration. In a push pull configuration each transistor handles 50% of the audio signal as can be seen in figure 2.7. The strength of this amplifier is the increase in efficiency vs the class A. The efficiency of the class B amplifier is at best around 50% which is better than the class A but still waists a lot of energy and adds a new weakness as the transistors share in the load of the signal and while one is open the other is closed.[10]

---

[8] See appendix B on detail for crossover distortion.
CHAPTER 2. THEORETICAL BACKGROUND

The weakens of this design is that it has a dead zone where no transistor is open this zone is called a crossover distortion and is caused by the construction of the transistor as it needs to have a signal larger than 0.7V between base and emitter. This has to be addressed so the audio quality does not suffer. An example of this can be seen in figure 2.8. In this circuit two diodes are used to combat the base-emitter voltage drop by raising the voltage on the base, this configuration is called a class AB amplifier.

2.5.3 Class AB amplifier

A class AB amplifier is not a class in its own right but a combination of the class A and the class B amplifier. The class AB design is to have two transistors in a push pull configuration but while in the class B where each transistor is on 50% of the time they now are open a little longer in order to combat the crossover distortion, see figure 2.8. The efficiency of the class AB amplifier can go as high as 60%.

2.5.4 Class D amplifier

The Class D amplifier often called a digital amplifier because of how the signal coming in to the gate on the mosfet transistors is converted in to PWM using a saw tooth signal and a comparator.

---

9 See appendix B on detail for crossover distortion.
10 See appendix B on detail for crossover distortion.
2.5. CHOOSING AN AMPLIFIER

see figure 2.9. By changing the signal in to PWM\textsuperscript{11} the efficiency can be increased drastically compared to the other classes. This is achieved due to the construction of the mosfet transistors, as the main heat is when the operational point is between fully open and fully closed. By using the class D amplifier the efficiency can go as high as 90\%.\cite{10}

\begin{figure}[h]
\centering
\includegraphics[width=0.7\textwidth]{class_d_amplifier.png}
\caption{Basic construction of a class D amplifier \cite{10}}
\end{figure}

2.5.5 Conclusion

In conclusion after viewing the pros and cons of these different amplification and the difference in efficiency, as seen in figure 2.10, it is clear that the class D amplifier has the highest efficiency and is the simplest class for circuit design as most class D amplifiers are come in IC package needing only a few misc components to work.

\textsuperscript{11}See appendix B.9 on detail on PWM
2.6 Choosing an power source

There are a few different ways to design a power supply, one way is to use a SMPS which has a high efficiency but causes a lot of interference on the power line which can affect the audio quality. The interference is caused because of how the SMPS is designed, a SMPS uses high frequency to turn on and off a transistor on the primary winding. By doing this the efficiency of the transformer is increased however as discussed it creates high frequency distortion on the power line which will affect the audio quality.[11]

The other way is to use AC power supplies which may not have as good efficiency but make up for it by having less if none interference on the power line.

2.6.1 Conclusion

While the SMPS has better efficiency it is also a much more complex design which cannot be accomplished in the time given for this project and adding noise on the power line making the AC power supply a more suitable choice for this project.

2.7 Choosing a display

For this project there were few different displays where viewed the main contenders where the vacuum tube display and the LCD. A vacuum tube display is an old design that has been used in many old systems. The vacuum tube displays do look great however there is a catch and that in order to get them to work a special driver circuit is needed. This can complicate the design an affect the time schedule.[12]
2.8. CHOOSING A MICROCONTROLLER

The other way is to use a LCD like the Hitachi HD44780. The Hitachi LCD is a 2x16 character display that only needs to connect 6 pins to a microcontroller to work not counting the power pins.[13]

2.7.1 Conclusion

While the vacuum tubes do look good they are not a good fit for this project as time is of the essence, therefore the Hitachi HD44780 LCD is the choice for this project.

2.8 Choosing a microcontroller

The choice of microcontroller is greatly determined by the signals it needs to receive and send. The microcontroller needs to be able to receive the following signals see table 2.1.[14]

2.8.1 Conclusion

There are a number of microcontrollers capable of handling these kinds of signals however the ATmega family offers a simple design for the project. The Atmega328p and the Atmega2560 are both great however the atmega2560 is the only one capable of providing this volume of pins.
<table>
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<td>3.2</td>
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<td>LCD D7 pin</td>
<td>2.7</td>
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<td>7</td>
<td>O</td>
<td>MS1. uPC1892 mode selector</td>
<td>3.3</td>
</tr>
<tr>
<td>8</td>
<td>I</td>
<td>DIR9001 Error indicator</td>
<td>3.2</td>
</tr>
<tr>
<td>9</td>
<td>O</td>
<td>Input selector input</td>
<td>3.2</td>
</tr>
<tr>
<td>10</td>
<td>O</td>
<td>Input selector input</td>
<td>3.2</td>
</tr>
<tr>
<td>11</td>
<td>O</td>
<td>LCD enable pin</td>
<td>2.7</td>
</tr>
<tr>
<td>12</td>
<td>O</td>
<td>LCD RS pin</td>
<td>2.7</td>
</tr>
<tr>
<td>22</td>
<td>I</td>
<td>TAS5613_1: SD</td>
<td>3.4</td>
</tr>
<tr>
<td>23</td>
<td>I</td>
<td>TAS5613_1: RESET</td>
<td>3.4</td>
</tr>
<tr>
<td>24</td>
<td>I</td>
<td>TAS5613_1: OTW</td>
<td>3.4</td>
</tr>
<tr>
<td>25</td>
<td>I</td>
<td>TAS5613_1: OTW1</td>
<td>3.4</td>
</tr>
<tr>
<td>26</td>
<td>I</td>
<td>TAS5613_1: OTW2</td>
<td>3.4</td>
</tr>
<tr>
<td>27</td>
<td>I</td>
<td>TAS5613_1: CLIP</td>
<td>3.4</td>
</tr>
<tr>
<td>28</td>
<td>I</td>
<td>TAS5613_1: READY</td>
<td>3.4</td>
</tr>
<tr>
<td>29</td>
<td>I</td>
<td>TAS5613_2: SD</td>
<td>3.4</td>
</tr>
<tr>
<td>30</td>
<td>I</td>
<td>TAS5613_2: RESET</td>
<td>3.4</td>
</tr>
<tr>
<td>31</td>
<td>I</td>
<td>TAS5613_2: OTW</td>
<td>3.4</td>
</tr>
<tr>
<td>32</td>
<td>I</td>
<td>TAS5613_2: OTW1</td>
<td>3.4</td>
</tr>
<tr>
<td>33</td>
<td>I</td>
<td>TAS5613_2: OTW2</td>
<td>3.4</td>
</tr>
<tr>
<td>34</td>
<td>I</td>
<td>TAS5613_2: CLIP</td>
<td>3.4</td>
</tr>
<tr>
<td>35</td>
<td>I</td>
<td>TAS5613_2: READY</td>
<td>3.4</td>
</tr>
<tr>
<td>36</td>
<td>I</td>
<td>TAS5613_3: SD</td>
<td>3.4</td>
</tr>
<tr>
<td>37</td>
<td>I</td>
<td>TAS5613_3: RESET</td>
<td>3.4</td>
</tr>
<tr>
<td>38</td>
<td>I</td>
<td>TAS5613_3: OTW</td>
<td>3.4</td>
</tr>
<tr>
<td>39</td>
<td>I</td>
<td>TAS5613_3: OTW1</td>
<td>3.4</td>
</tr>
<tr>
<td>40</td>
<td>I</td>
<td>TAS5613_3: OTW2</td>
<td>3.4</td>
</tr>
<tr>
<td>41</td>
<td>I</td>
<td>TAS5613_3: CLIP</td>
<td>3.4</td>
</tr>
<tr>
<td>42</td>
<td>I</td>
<td>TAS5613_3: READY</td>
<td>3.4</td>
</tr>
<tr>
<td>ADC0</td>
<td>A</td>
<td>Volume</td>
<td>3.3</td>
</tr>
<tr>
<td>ADC1</td>
<td>A</td>
<td>Bass</td>
<td>3.3</td>
</tr>
<tr>
<td>ADC2</td>
<td>A</td>
<td>Treble</td>
<td>3.3</td>
</tr>
<tr>
<td>ADC3</td>
<td>A</td>
<td>Balance</td>
<td>3.3</td>
</tr>
<tr>
<td>ADC4</td>
<td>A</td>
<td>Bass external</td>
<td>3.3</td>
</tr>
</tbody>
</table>

Table 2.1: Microcontroller pin requirements and Decryptions
Chapter 3

Home theatre system design

3.1 Circuit design overview

This chapter will go deeper into the design of each phase of the circuit designs and show case what should be kept in mind while designing the system.

Figure 3.1: Block diagram of the design overview. The blue line is the audio path the red line are error signals and the black lines are the signals from the microprocessor

3.2 Audio input

As discussed in chapter 2.3 be for any signal can be amplified a way of receiving it has to be chosen. As a choice of optical and coaxial has been taken a circuit has to be found that can receive the PCM signal and then convert that to a signal that can be used without affecting the audio quality. The circuit chosen after research can be seen in appendix D.1.

An example of the audio path from that circuit can be seen in figure 3.2.
CHAPTER 3. HOME THEATRE SYSTEM DESIGN

3.2.1 74HC00

This circuit uses the 74HC00 which is a 2 input NAND gate to choose between the optical and coaxial. The circuit choses between these signals by using the truth table of the NAND gates. One gate is set up as an inverter to turn a high signal in to a low signal and vice versa. A switch changes the state in this circuit but a microcontroller can be used to do this as well. The only thing to keep in mind when choosing a NAND gate like this one is that the transition time is as low as possible in order to handle the incoming signal. The transition time on the 74HC00 is 7ns.

3.2.2 DIR9001

After the signal has goes through the 74HC00 the signal enters the circuit DIR9001 which is a 24 bit Digital Audio Interface receiver. The DIR9001 can receive signal in the range of 28 kHz to 108 kHz [16]

The signal received at the RXIN pin is a biphase encoded. The FMT0 and FMT1 are control pins which control how the audio signal out is formed. In this circuit the signal is set

---

\[1\] See appendix B.2 for detail on biphase encoded signals
3.2. AUDIO INPUT

for 24 bits $I^2S$ with the MSB first.

PSCK0 and PSCK1 are control pins which control the frequencies for the SCKO, BCKO and LRCKO pins. However in this circuit all clocks come from the PLL source as the CKSEL pin is set to ground.

According to the data sheet the RSN control pin has to be set to ground. /RST or reset has to be connected to 5v for the circuit to work.[16]

The outputs in this circuit are SCKO which stands for system clock output. LRCKO determines if the data being sent at that moment is for the left channel or the right. The BCKO is the clock for the audio data. And the DOUT pin is the output delivering 16-24bit audio data. Figure 3.5 shows the output from DIR9001.
CHAPTER 3. HOME THEATRE SYSTEM DESIGN

3.2.3 PCM1793

After the signal has been changed in I²S by the DIR9001 it can be received by the PCM1793. The PCM1793 is a CMOS stereo DAC with all the circuitry needed to receive and convert the I²S stream to stereo. See figure 3.6. [17]

In this configuration the FMT0 is set high the FMT1 is set low and the FMT2 is high this sets the audio data at 24bits.

3.2.4 OPA1234

The D/S and filter circuit is shown in figure 3.6 is shown in figure 3.7. This is a digital low pass filter needed to filter out unwanted frequencies. The opamp used for this is the OPA2134 which is a high performance audio operational amplifier. This is needed as there might be some noise in the signal which needs to be filtered out.
Figure 3.7: PCM1793 low pass filter circuit [17]
CHAPTER 3. HOME THEATRE SYSTEM DESIGN

3.3 Audio processor

3.3.1 uPC1892

The audio processor chosen in chapter 2.4 was the uPC1892. The diagram for the uPC1892 can be seen in figure 3.8. As can be seen it can receive stereo signal and converting it in to a left and right output, rear output and a L+R output which is the bass output.[9]

![Figure 3.8: Block diagram for uPC1892 IC][9]

Pins 7 and 8 are control pin that determine if the mode in which the signal is presented. See table 3.1. The mode setting sets the signal through unfiltered or through a series of phase shifters and a matrix encoder designed for the desired affect.[9] The added benefit of using the uPC1892 is that it only uses five potentiometers and by using dual gang potentiometer one can be used for the uPC1892 while the other can be used for the microcontroller to display the relevant data to the LCD.

Table 3.1: Mode select for uPC1892 [9]

<table>
<thead>
<tr>
<th>Mode / Code</th>
<th>MS1</th>
<th>MS2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>L</td>
<td>L</td>
</tr>
<tr>
<td>Music</td>
<td>H</td>
<td>L</td>
</tr>
<tr>
<td>Movie</td>
<td>L</td>
<td>H</td>
</tr>
<tr>
<td>Simulated</td>
<td>H</td>
<td>H</td>
</tr>
</tbody>
</table>
3.4 Amplifier

3.4.1 TAS5613

As decided in chapter 2.5 the amplifier class chosen for this project is a D-class amplifier and as stated in chapter 1.2 the amplifier needs to be able to handle 100-200 watts and have a THD lower than 1%. In order to accomplish this the TAS5613 was chosen, see figure 3.9. The TAS5613 promises a 0.03% THD at 1 watt into 4Ω and has the possibility of 150 watts per channel. The reason the TAS5613 was chosen instead of the TPA3251 was the availability of evaluation boards. The TAS5613 has an efficient power rating above 90%.

The unique design of the TAS5613 allows for fewer voltages in order to control needing only a few misc components.[18]

![Diagram explaining basic connections and use for TAS5613](image)

Figure 3.9: Diagram explaining basic connections and use for TAS5613 [18]
As can be seen in figure 3.9 the TAS5613 offers the ability to send signals back, be it errors or conformation of ready. A list of signals from the TAS5613 can be seen in table 3.2. [18]

**Table 3.2: TAS5613 signal table [18]**

<table>
<thead>
<tr>
<th>Pin</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SD</td>
<td>Shutdown signal, open drain, active low</td>
</tr>
<tr>
<td>RESET</td>
<td>Device reset Input active low, requires $47k\Omega$ pull up resistor to VREG</td>
</tr>
<tr>
<td>OTW</td>
<td>Over temperature warning signal, open drain, active low.</td>
</tr>
<tr>
<td>OTW1</td>
<td>Over temperature warning signal, open drain, active low</td>
</tr>
<tr>
<td>OTW2</td>
<td>Over temperature warning signal, open drain, active low</td>
</tr>
<tr>
<td>CLIP</td>
<td>signal is indicating that the output is approaching clipping. The signal can be used to either an audio volume decrease or intelligent power supply controlling a low and a high rail.</td>
</tr>
<tr>
<td>READY</td>
<td>Normal operation; open drain; active high</td>
</tr>
</tbody>
</table>

### 3.5 Power supply

The main role of the power supply is in fact to supply all the circuitry the required voltages. It is important to know first which circuits will be used. In table 3.3 has an outline on which circuits are used and their respective voltages.

**Table 3.3: Voltage table for the requirements of all the circuits**

<table>
<thead>
<tr>
<th>Circuit</th>
<th>Voltage</th>
</tr>
</thead>
<tbody>
<tr>
<td>TAS5613</td>
<td>12v and 36v</td>
</tr>
<tr>
<td>DIR9001</td>
<td>3.3v</td>
</tr>
<tr>
<td>74HC00</td>
<td>-0.5 - 7v</td>
</tr>
<tr>
<td>PCM1793</td>
<td>3.3v and 5v</td>
</tr>
<tr>
<td>OPA2134</td>
<td>±2.5V to ±18V</td>
</tr>
<tr>
<td>uPC1892</td>
<td>5v and 12v</td>
</tr>
<tr>
<td>ATmega2560</td>
<td>5V</td>
</tr>
</tbody>
</table>

As can be seen in table 3.3 the highest voltage needed comes from the TAS5613, this is gotten from the datasheet of the TAS5613.

As concluded in chapter 2.6 the design uses AC power supply design meaning one of the secondary winding needs to have a winding ratio of $10/1$ thus giving out 24 volt AC. By using a full wave rectifier and a capacitor bank the $36V$ dc is achieved.

$$V_{dc} = \sqrt{2} \times V_{ac}$$  \hspace{1cm} (3.1)

It was decided to use a Toroid transformer as it has a greater efficiency than a normal iron core transformer. The toroid achieves the greater efficiency as the dome shape uses the magnetic flux in the core better. The rest of the voltages use different secondary windings as can be seen in figure 3.10.

As the right transformer did not arrive in time a little power supply had to be made in order to provide the circuit power with all the relevant voltages. See figure 3.11.
3.6 Microcontroller

3.6.1 ATmega2560

As discussed in chapter 2.8 there is a need to provide over 33 digital pins and 5 analog pin. The ATmega2560 has over 54 digital input and output pins and 16 analog pins. For this reason the ATmega is a clear choice for the project.\[14\]

The ATmega2560 is a 8 bit microcontroller that uses RISC architecture. The RISC architecture or Reduced Instruction Set Computer uses compact but powerful commands instead of using more specialized commands as can be seen in many other types of microcontrollers.\[19\]

The main tasks of the Microcontroller can be see below.

1. Receive errors from DIR9001 and TAS5613
2. Control LCD
3. Receive button commands
4. Shut down amplifier if mayor errors occur.

To program the ATmega2560 it was desired to use the Arduino IDE \[14\] as all the libraries are present and will shorten the prototyping time. However as the libraries are closed and a mix of C and C++ and there is no way to tell what the microcontroller is doing.
Figure 3.11: Schematic for the power transformer

However, there is time the there are plans to use Eclipse IDE which can allows the AT-mega2560 to be programed in C which allows for faster programs and more control.[21] However, there is not much need for fast programs as the program is set to continuously monitor the input pins and respond to what those pins states.

Figure 3.13 shows the flowchart for the microcontroller as indicated earlier it reads the inputs for error.
Figure 3.12: Arduino Atmega2560 pin mapping [20]
Figure 3.13: Home theatre system microcontroller flowchart
Chapter 4

Simulations and measurements

4.1 Simulations

Simulations where maid in LTspice[22] of the output filter of the TAS5613 as all other circuits cannot be simulated as they are specially designed circuits that cannot be simulated by LTspice. This simulation will then be compared to the graph made from the transfer function in appendix B.10.

Figure 4.1: TAS5613 output filter simulations made in LTspice
4.1.1 Conclusion

Comparing the bode plot from figure 4.1 to the plot made with the transfer function in appendix B.10 it is apparent that the transfer function and simulations are the same and should result in a drop of -30dB for the $F_{PWM}$. 
4.2 Measurements

4.2.1 Efficiency

The efficiency measurements were done by measuring the current and voltage supplied to the circuit and compared to the power output of the amplifier. Test conditions 1kHz sine wave with an amplitude of 2V provided by KENWOOD AG-203 oscillator. Output load is HSC-100-4R7 is a 4.7Ω 100w resistor.

Current from the power supply is 1.79A and the voltage is 35.6V. Using this the watts are calculated as 63.7W. The output power is derived from figure 4.9 and using the equation 4.1.[23]

\[ P = \frac{V^2 \times 0.3535^2}{4.7\Omega} \tag{4.1} \]

Using this method the power is calculated as 52.7W. To find the efficiency equation 4.2.[23]

\[ n = \frac{P_{in}}{P_{out}} \times 100 \tag{4.2} \]

Using this the efficiency is calculated as 82.7% which is good however 7.3% from the target however this is caused from other components in the evaluation board. Using this results in future work a more efficient amplifier can be constructed.[23]
4.2.2 Power supply

As can be seen in figure 4.2 the output voltage of the full wave rectifier is $35.6V$ with a ripple of around $1.5V$. The ripple can be lessened by adding more capacitors however is does not seem to effect the quality of the output signal and thus does not merit more capacitors.

Figure 4.2: Power supply measurements and quality test
4.2.3 Audio input

Test conditions 1kHz sine wave from the optical output from media station Verbatim model 47543. The signal is a PCM signal. The verbatim 47540 does have the ability to send a signal that is RAW instead of PCM however the DIR9001 does not appear to understand this signal.

Figure 4.3: PCM measurements taken at the optical input

Figure 4.3 shows the PCM signal taken from the $V_{out}$ pin of the optical input. As can be seen there is some over shoot in the signal however there is not much that can be done with this as this is the signal as clean as can be from the audio source.

Figure 4.4: Signal measurements comparison on coaxial and optical
Figure 4.4 shows the PCM signal taken from the optical input (yellow) and coaxial input (purple). Comparing these signals it is clear that the optical signal is a far greater digital signal as the optical looks more like a square than the coaxial.

Figure 4.5: Signal measurements taken at RXIN pin of DIR9001. Showing the PCM signal

Figure 4.5 shows the PCM signal taken from the RXIN pin of the DIR9001. Comparing this to the signal at the optical input it is clear that the signal has not been affected by the circuitry.

Figure 4.6: Signal measurements taken at Optical in (yellow) LRCKO(blue), BCKO(purple) and DOUT(turquoise) pins of DIR9001

Figure 4.5 shows the 24 bit $I^2S$ signal taken from the DIR9001 and the PCM input signal. This signal shows that the 1kHz signal is being sent to the right channel and the left channel at
a frequency of \(44.0\, kHz\) seen at the LRCKO. By comparing this to the \(I^2S\) theory it is clear that the signal is exactly like the theory.

![Figure 4.7: analog signal from the PCM1793](image)

Figure 4.7 shows the analog signal from the output pins of the PCM1793. The signal is having connectivity problem due to the fact that the circuit prototype is on a bread board. By doing FFT measurements on the output signal we could get the frequency response of the circuit however due to the connectivity issues the FFT cannot be done.

This signal is then be sent to the OPA2134 circuit for filtering. The measurements planned for this part of the circuit where not possible due to time constraints of the impending deadline. However just by hooking up headphones to the output pins of the PCM1793 showed that there is audio signal there that needs to be filtered out.

### 4.2.4 Amplifier

Test conditions \(1kHz\) sine wave with an amplitude of \(2V\) provided by KENWOOD AG-203 oscillator. Distortion factor \(400Hz-20kHz\), 0.1% or less. Output load is HSC-100-4R7 which is a \(4.7\Omega\) 100w resistor. Unfortunately there was not enough time to find an \(8\Omega\) resistor or resistors with a big enough watt rating to fully test the amplifier in time. Power to the circuit is from a 600VA with a full wave rectifier and a \(20mF\) 63V capacitor bank. Oscilloscope RIGOL DS1104.

Figure 4.8 shows the signal from the KENWOOD AG-203 oscillator and FFT measurements made on it. The oscillator can do FFT measurements however it cannot show the THD level. However using the graph the THD was calculated to be \(7.8\%\) using equation B.2.

This is a lot more than expected however the facilities in Háskólinn í Reykjavík does not have an audio analyser meaning that this cannot be proven by measurements. There is also a lot of noise in the signal that could be affecting the results.

This signal will however be used and compared to the signal coming out of the amplifier to see the changes in the signal.
Figure 4.9 shows the gain and THD measurements made on the output signal. By using the gain formula it is calculated that the gain is $26.9\,\text{dB}$.\footnote{See chapter C.3} An by using equation B.2 the THD is calculated to be around $0.107\%$ which is much lower than the target set for this project which was $1\%$.

\footnote{See chapter C.3}
4.2. MEASUREMENTS

Figure 4.10 shows the output signal of the amplifier over a broader bandwidth than in figure 4.9. By viewing the graph is shows a small spike at 50kHz which is 20kHz lower than the transfer function and simulations dictate. This big a difference means that the evaluation board does not have the exact same components as the manufacturer specifies in its datasheet.
4.3 Results

From the measurements it is clear that a number of goals that were set for this project were made. See below.

- Have output power between 100-200w. The TAS5613 can give power up to 150w per channel.

- Design an amplifier with a THD lower than 1%. As seen in the results for graph 4.9 the THD is 0.107% at 1kHz at 52.6 watts.

There were a few measurement that where planned for this project that where not done due to a lack of time, these include the signal to noise ratio in a digital signal showing that for a 16 bit signal it should be around 96dB and for a 24 bit signal it should be 144dB.

And THD measurements on the output sinus wave from the OPA2134.
Chapter 5

Discussion

This chapter outlines what was a success and what aspects of this project where failed and what could have been done to fix these problems. This is then compared to the research tasks and main risks of the project.

5.1 Summary

5.1.1 Successes

1. The optical and coaxial input chosen for this project where a success. However there are some connection problems at present however they are due to the construction of the board and would go away once a professionally PCB has been made.

2. Have a THD lower than 1%. According to calculations the THD of the amplifier is 0.107% however the only fail is that the equipment in the electronic lab only has the option to make a Fast Fourier Transformation but not output the THD or THD+N, however this was overcome by using math. By further research it became clear that there are special mustimeters that are used for this purpose like the Keithley 2015 Series: THD and Audio Analysis Multimeter.

3. Design a home theatre that is efficient in amplifying the signal. As it stands the amplifier has an efficiency rating of 82.7% which is close to what a class D amplifier can do the remaining 7.3% can a somewhat because by a voltage regulator on the evaluation board meaning in future design the efficiency rating might get better.

4. Have a power rating per channel between 100-200w. The TAS5613 amplifier chosen for this project has a power rating of 150w per channel.

5. Build a prototype. There is a working prototype that can deliver sound to one speaker for a few moments at a time. At the moment this is due to the construction of the PCB and bad connections. This will be remedied in the final product as the prototype is just a proof of concept.

6. Comparing the signals. By comparing the signals of the measurements to the data sheets we clearly see that they match exactly for what was expected.

In the fails of this project where caused by many of the thins described in the main risk chapter 1.3
5.1.2 Fails

1. Time and running out of it.

- Able to operate at room temperature. The TAS5613 amplifier was designed to be able to operate at ambient temperature, however there was not enough time to do a full test on the amplifier as is needs to operate for a long time to get adequate results. But in current tests the heat rise of the heat sink was negligible.
- Not getting parts in time.
  - The audio processor did not arrive in time for the dead line. However it will still be used for the final product when it arrives.
  - The transformer required for this project did not arrive in time for the dead line. However it will still be used for the final product when it arrives.
- Unforeseen events. There were a number of unforeseen event that did arise.
  - Some classes in school did take more time than anticipated.
  - Parts got lost in mail.
  - Not able to get a high enough wattage dummy load in time for the deadline.

2. Proof of concept.

- As discussed in the success part while the amplifier does show a low THD there is no way of accurately proofing the success of this due to lack of equipment.

5.2 Conclusion

While there where a few mishaps in this project there are a lot more things that did go well and the only reason that stops this project form becoming a complete product is time. The circuits will be sent out to a PCB manufacturer that should reduce the errors in connectivity. The power supply will be redesigned to reduce cost in parts. All of these things will not take a lot of time as all of the circuit schematics and the bill of materials are complete.
Bibliography


Appendix A

Code

1 // Includes

3 #include <LiquidCrystal.h>

5 // initialize

7 LiquidCrystal lcd(12, 11, 5, 4, 3, 2);

9 // Pin define analog

11 const int Volume = A0;
  const int Bass = A1;
13 const int Treble = A2;
  const int Balance = A3;
15 const int Bassext = A4;

17 // Pin define digital
  const int InputSelectorOutput = 1;
19 const int MS2uPC1892ModeSelector = 6;
  const int MS1uPC1892ModeSelector = 7;
21 const int DIR9001ErrorIndicator = 8;
  const int InputSelectorInput1 = 9;
23 const int InputSelectorInput2 = 10;

25 const int TAS56131SD = 22;
  const int TAS56131RESET = 23;
27 const int TAS56131OTW = 24;
  const int TAS56131OTW1 = 25;
29 const int TAS56131OTW2 = 26;
  const int TAS56131CLIP = 27;
31 const int TAS56131READY = 28;
  const int TAS56132SD = 29;
33 const int TAS56132RESET = 30;
  const int TAS56132OTW = 31;
35 const int TAS56132OTW1 = 32;
const int TAS56132OTW2 = 33;
const int TAS56132CLIP = 34;
const int TAS56132READY = 35;
const int TAS56133SD = 36;
const int TAS56133RESET = 37;
const int TAS56133OTW = 38;
const int TAS56133OTW1 = 39;
const int TAS56133OTW2 = 40;
const int TAS56133CLIP = 41;
const int TAS56133READY = 42;

//pin states
int VolumeValue = 0;
int BassValue = 0;
int TrebleValue = 0;
int BalanceValue = 0;
int BassextValue = 0;
int OldVolumeValue = 1;
int OldBassValue = 1;
int OldTrebleValue = 1;
int OldBalanceValue = 1;
int OldBassextValue = 1;
int InputSelectorOutputState = 0;
int MS2uPC1892ModeSelectorState = 0;
int MS1uPC1892ModeSelectorState = 0;
int DIR9001ErrorIndicatorState = 0;
int InputSelInput1State = 0;
int InputSelInput2State = 0;
int TAS56131SDState = 0;
int TAS56131RESETState = 0;
int TAS56131OTWState = 0;
int TAS56131OTW1State = 0;
int TAS56131OTW2State = 0;
int TAS56131CLIPState = 0;
int TAS56131READYState = 0;
int TAS56132SDState = 0;
int TAS56132RESETState = 0;
int TAS56132OTWState = 0;
int TAS56132OTW1State = 0;
int TAS56132OTW2State = 0;
int TAS56132CLIPState = 0;
int TAS56132READYState = 0;
int TAS56133SDState = 0;
int TAS56133RESETState = 0;
int TAS56133OTWState = 0;
int TAS56133OTW1State = 0;
int TAS56133OTW2State = 0;
int TAS56133CLIPState = 0;
int TAS56133READYState = 0;

// the setup routine runs once when you press reset:
void setup() {
  // initialize serial communication at 9600 bits per second:
  Serial.begin(9600);
  lcd.begin(16, 2);

  pinMode(InputSelectorOutput, OUTPUT);
  pinMode(MS2uPC1892ModeSelector, OUTPUT);
  pinMode(MS1uPC1892ModeSelector, OUTPUT);
  pinMode(DIR9001ErrorIndicator, INPUT);
  pinMode(InputSelectorInput1, INPUT);
  pinMode(InputSelectorInput2, INPUT);

  pinMode(TAS56131SD, INPUT);
  pinMode(TAS56131RESET, OUTPUT);
  pinMode(TAS56131OTW, INPUT);
  pinMode(TAS56131OTW1, INPUT);
  pinMode(TAS56131OTW2, INPUT);
  pinMode(TAS56131CLIP, INPUT);
  pinMode(TAS56131READY, INPUT);

  pinMode(TAS56132SD, INPUT);
  pinMode(TAS56132RESET, OUTPUT);
  pinMode(TAS56132OTW, INPUT);
  pinMode(TAS56132OTW1, INPUT);
  pinMode(TAS56132OTW2, INPUT);
  pinMode(TAS56132CLIP, INPUT);
  pinMode(TAS56132READY, INPUT);

  pinMode(TAS56133SD, INPUT);
  pinMode(TAS56133RESET, OUTPUT);
  pinMode(TAS56133OTW, INPUT);
  pinMode(TAS56133OTW1, INPUT);
  pinMode(TAS56133OTW2, INPUT);
  pinMode(TAS56133CLIP, INPUT);
  pinMode(TAS56133READY, INPUT);

  void loop() {
    DIR9001ErrorIndicatorState = digitalRead(DIR9001ErrorIndicator);
    int InputSelectorInput1State = digitalRead(9);
    int InputSelectorInput2State = digitalRead(10);
    

APPENDIX A. CODE

TAS56131SDstate = digitalRead(TAS56131SD);
TAS56131RESETState = digitalRead(TAS56131RESET);
TAS56131OTWState = digitalRead(TAS56131OTW);
TAS56131OTWState = digitalRead(TAS56131OTW1);
TAS56131OTW2State = digitalRead(TAS56131OTW2);
TAS5611CLIPState = digitalRead(TAS5611CLIP);
TAS56131READYState = digitalRead(TAS56131READY);

TAS56132SDstate = digitalRead(TAS56132SD);
TAS56132RESETState = digitalRead(TAS56132RESET);
TAS56132OTWState = digitalRead(TAS56132OTW);
TAS56132OTW1State = digitalRead(TAS56132OTW1);
TAS56132OTW2State = digitalRead(TAS56132OTW2);
TAS56132CLIPState = digitalRead(TAS56132CLIP);
TAS56132READYState = digitalRead(TAS56132READY);

TAS56133SDstate = digitalRead(TAS56133SD);
TAS56133RESETState = digitalRead(TAS56133RESET);
TAS56133OTWState = digitalRead(TAS56133OTW);
TAS56133OTW1State = digitalRead(TAS56133OTW1);
TAS56133OTW2State = digitalRead(TAS56133OTW2);
TAS56133CLIPState = digitalRead(TAS56133CLIP);
TAS56133READYState = digitalRead(TAS56133READY);

// Input Control

if(InputSelectorInput1State == HIGH)
{
    InputSelectorOutputState = HIGH;
}
if(InputSelectorInput1State == LOW)
{
    InputSelectorOutputState = LOW;
}

// Error signals

if( TAS56131SDstate == LOW && TAS56131OTWState == LOW &&
    TAS56131OTW1State == LOW || TAS56131OTW2State == LOW )
{
    lcd.print("Error temp");
}
if( TAS56131SDstate == LOW && TAS56131OTWState == LOW &&
    TAS56131OTW1State == HIGH || TAS56131OTW2State == HIGH )
{
    lcd.print("Error temp");
}
if( TAS56131SDstate == LOW && TAS56131OTWState == HIGH &&
    TAS56131OTW1State == HIGH || TAS56131OTW2State == HIGH )
{
    lcd.print("Error temp");
}
{  
lcd.print("Error temp");
}

183 if( TAS56131SDstate == HIGH && TAS56131OTWState == LOW && (←
   ~TAS56131OTWState == LOW || TAS56131OTW2State == LOW) )
{
    lcd.print("Overtemperature warning 2");
    // here the amplifier should be turned off future work
    if( TAS56131SDstate == HIGH && TAS56131OTWState == LOW && (←
       ~TAS56131OTWState == HIGH || TAS56131OTW2State == HIGH) )
    {
        lcd.print("Overtemperature warning 1");
    }
}

195 if( TAS56132SDstate == LOW && TAS56132OTW1State == LOW && (←
    ~TAS56132OTWState == LOW || TAS56132OTW2State == LOW) )
{
    lcd.print("Error temp");
}

199 if( TAS56132SDstate == LOW && TAS56132OTW1State == LOW && (←
    ~TAS56132OTWState == HIGH || TAS56132OTW2State == HIGH) )
{
    lcd.print("Error temp");
}

203 if( TAS56132SDstate == LOW && TAS56132OTW1State == HIGH && (←
    ~TAS56132OTWState == LOW || TAS56132OTW2State == LOW) )
{
    lcd.print("Overtemperature warning 2");
    // here the amplifier should be turned off future work
    if( TAS56132SDstate == HIGH && TAS56132OTW1State == LOW && (←
       ~TAS56132OTWState == HIGH || TAS56132OTW2State == HIGH) )
    {
        lcd.print("Overtemperature warning 1");
    }
}

217 if( TAS56133SDstate == LOW && TAS56133OTW1State == LOW && (←
    ~TAS56133OTWState == LOW || TAS56133OTW2State == LOW) )
{
    lcd.print("Error temp");
}
if( TAS56133SDstate == LOW && TAS56133OTW1State == LOW && (TAS56133OTWState == HIGH || TAS56133OTW2State == HIGH) )
{
    lcd.print("Error temp");
}

if( TAS56133SDstate == LOW && TAS56133OTW1State == HIGH && (TAS56133OTWState == HIGH || TAS56133OTW2State == HIGH) )
{
    lcd.print("Error temp");
}

if( TAS56133SDstate == HIGH && TAS56133OTW1State == LOW && (TAS56133OTWState == LOW || TAS56133OTW2State == LOW) )
{
    lcd.print("Overtemperature warning 2");
    // here the amplifier should be turned off future work
}

if( TAS56133SDstate == HIGH && TAS56133OTW1State == LOW && (TAS56133OTWState == HIGH || TAS56133OTW2State == HIGH) )
{
    lcd.print("Overtemperature warning 1");
}

//Volume control
int VolumeValue = analogRead(Volume);
VolumeValue = map(VolumeValue, 0, 1023, 0, 64);
if(VolumeValue != OldVolumeValue)
{
    lcd.print("Volume");
    lcd.print(VolumeValue);
    VolumeValue = OldVolumeValue;
}

//Bass control
BassValue = analogRead(Bass);
BassValue = map(BassValue, 0, 1023, 0, 64);
if(BassValue != OldBassValue)
{
    lcd.print("Bass");
    lcd.print(BassValue);
    OldBassValue = BassValue;
}

//Treble control
int TrebleValue = analogRead(Treble);
TrebleValue = map(TrebleValue, 0, 1023, 0, 64);
if(TrebleValue != TrebleValue)
{
    lcd.print("Treble");
    lcd.print(TrebleValue);
    OldTrebleValue = TrebleValue;
}

// Balance control

BalanceValue = analogRead(Balance);
    BalanceValue = map(BalanceValue, 0, 1023, -16, 16);
if(BalanceValue != OldBalanceValue)
{
    lcd.print("Treble");
    lcd.print(BalanceValue);
    OldBalanceValue = BalanceValue;
}

// Bassext control

BassextValue = analogRead(Bassext);
    BassextValue = map(BassextValue, 0, 1023, 0, 64);
if(BassextValue != OldBassextValue)
{
    lcd.print("BassextValue");
    lcd.print(BassextValue);
    OldBassextValue = BassextValue;
}
Appendix B

Key terms & concepts

B.1 HDMI

In modern times large-screen HDTVs have achieved widespread acceptance. In response to this many producers of home theaters have started designing devices that include HDMI in order to simplify things for the consumers. A few of these devices are soundbars and audio/video receivers. These devices enhance the users experience with better audio while complementing the HDTV video performance.[24]

The HDMI uses transition-minimized differential signaling lines to carry video, audio and data. The HDMI also uses a display data channel in order to exchange EDID, CEC and HDCP. The HDMI DDC lines also carry The current HDMI 1.4 protocol also has an ARC channel which is for audio return from a TV.[3]

B.1.1 EDID

EDIE or Extended Display Identification Data is the first bit of data transmitted in the HDMI communication protocol. The EDID is a 128 byte long for the VESA or Video Equipment Standards Association and 256 byte long if the CEA-861 or Consumer Electronics Association is added. The EDIE describes what capabilities a display has, as seen in table B.1. All data for the EDID are sent over the DDC lines using $I^2C$ protocol1.[3]

In some cases there may be a need to convert VGA signals into HDMI. In those cases some changes need to be made to the EDID in order for it to work. These changes are outlined in table B.2.

B.1.2 CEC

The CEC or consumer electronics control channel is a single-wire communication designed to enhance the abilities of a home theatre system. One such ability is to have one remote control turning on all devices with a single push of a button. As the HDMI continues to evolve this ability becomes more and more in need. [25]

---

1 See chapter B.14 for details on $I^2C$
APPENDIX B. KEY TERMS & CONCEPTS

Table B.1: EDID byte descriptions [3]

<table>
<thead>
<tr>
<th>Address</th>
<th>Bytes</th>
<th>Description</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>00h</td>
<td>8</td>
<td>Header: (00 FF FF FF FF FF FF FF 00)h</td>
<td>Mandatory fixed block header</td>
</tr>
<tr>
<td>08h</td>
<td>10</td>
<td>Vendor and product identification</td>
<td></td>
</tr>
<tr>
<td>08h</td>
<td>2</td>
<td>ID manufacturer name</td>
<td>Three compressed ASCII character code issued by Microsoft®</td>
</tr>
<tr>
<td>12h</td>
<td>2</td>
<td>EDID structure version and revision</td>
<td></td>
</tr>
<tr>
<td>12h</td>
<td>1</td>
<td>Version number: 01h</td>
<td>Fixed</td>
</tr>
<tr>
<td>13h</td>
<td>1</td>
<td>Revision number: 03h</td>
<td>Fixed</td>
</tr>
<tr>
<td>18h</td>
<td>1</td>
<td>Feature support</td>
<td>Features such as power management and color type. Bit 1 should be set to 1.</td>
</tr>
<tr>
<td>36h</td>
<td>72</td>
<td>18 byte data blocks</td>
<td></td>
</tr>
<tr>
<td>36h</td>
<td>18</td>
<td>Preferred timing mode</td>
<td></td>
</tr>
<tr>
<td>48h</td>
<td>18</td>
<td>Detailed timing #2 or display descriptor</td>
<td></td>
</tr>
<tr>
<td>5Ah</td>
<td>18</td>
<td>Detailed timing #3 or display descriptor</td>
<td></td>
</tr>
<tr>
<td>6Ch</td>
<td>18</td>
<td>Detailed timing #4 or display descriptor</td>
<td></td>
</tr>
<tr>
<td>7Eh</td>
<td>1</td>
<td>Extension block count N</td>
<td>Number of 128-byte EDID extension blocks to follow.</td>
</tr>
<tr>
<td>7Fh</td>
<td>1</td>
<td>Checksum</td>
<td>1-byte sum of all 128 bytes in this EDID block shall equal zero.</td>
</tr>
<tr>
<td>80...</td>
<td></td>
<td>Block map or CEA extension</td>
<td></td>
</tr>
</tbody>
</table>

B.1.3 HDCP

For a number of types of devices a certain type of licence is required, a so called HDCP. HDCP or High-bandwidth Digital Content Protection is a security standard designed to protect movies and other media from theft. HDCP is in fact the main reason on why the option of using the HDMI was cancelled as the licences required are expensive and not being able to connect to a device using HDCP would limit the design of the project. There are a number of ways to bypass the HDCP however they are not stable which is not a desirable option. [26]
### B.2 Biphase Encoding

Biphase encoding is a variation on polar encoding which is good for combating synchronisation problems. Polar encoding works on being symmetrical around 0 volts, the RS-232D interface used this code. Biphase encoding works by changing the signal in the middle of the bit intervals. This mid-interval change is perfect for synchronisation.

![Figure B.1: An example of polar encoding](image)

Biphase encoding works by changing the signal in the middle of the bit intervals. This mid-interval change is perfect for synchronisation.

![Figure B.2: An example of biphase encoding](image)

### B.3 Gain

Gain is an audio term denoting the amplitude, or level, of an electrical signal. It is represented in dB. The dB unit is used to represent the gain in an easier way as a double of power in watts is
only 6dB increase, see equation B.1 \[23\]

\[
gain = 10 \log \left( \frac{V_{out}}{V_{in}} \right)^2 \quad (B.1)
\]

This is also useful as the human ear works on a log scale and it makes larger numbers more relatable to the user.\[30\]
B.4 Surround sound

Surround sound is a technique for enriching the sound reproduction quality of an audio source by adding additional audio channels instead of using only front right and front left channels. By adding these channels the audio will come from all directions, see figure B.3.

The setup in figure B.3 is of a so called 5.1 set up meaning that there are five speakers and one base speaker. [2]

By adding these extra audio channels the experience of watching a movie is greatly enriched as the viewer can immerse himself more into the movie for instance when an explosion occurs in the movie behind the camera that sound will come from the speakers behind them.[2]

![Figure B.3: Surround sound setup [2]](image)

There are a number of different methods used today to process the audio signal into surround sound. The most popular is Dolby Pro-Logic which decodes the audio signal into each channel giving a better audio quality. [31]

A different kind of audio manipulation is to use passive components as can be seen in figure B.4. The method is simple however it is not as good as the Dolby Pro-Logic one as it indiscriminately amplifies and filters the signal while the Dolby Pro-Logic can detect minor changes in the audio signal.[2]

B.5 Bandwidth

The bandwidth of an amplifier is often based on the bandwidth of the human ear, for there is no reason to amplify sound that the human ear cannot hear. The human ear can detect on average sound from 20Hz to 20 kHz when people are young and healthy. However the 20 kHz will go decrees with age. In order to keep good audio quality there must be no tops or falls over 0.5dB throughout the bandwidth.[30]

B.6 Noise and Distortion

When the subject of audio quality is addressed it is often mentioned the THD and THD+N. A typical amplifier has a THD value lower than 1%.
The THD of a signal is a measurement of the harmonic distortion present and is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency as can be seen in equation B.2. Sometimes there is also some noise in the signal to then equation B.3.

\[ THD = \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + \ldots + V_n^2}}{V_1} \times 100 \]  

\[ THD + N = \sum_{i=2}^{\infty} \frac{\text{Harmonics + Noise}}{\text{fundamental}} \]  

When designing an Home theatre system it is imperative to keep the THD as low as possible to keep the audio signal unchanged however in some applications like a guitar amplifiers a high THD may be needed to change the audio signal from the guitar. As a low THD in this project is a requirement it is crucial to find circuits and components that offer lower THD in the design.

Figure B.5 shows an input signal with no THD and how it has changed on the output where it now has a THD of 31% which is a far cry from the target of under 1%.

In figure B.6 the same signal is shown however now noise has been introduced. It is apparent that the signal has become much more distorted from what the input signal is and that value is then shown as THD+N and is around 32.1%.
B.7. DAMPING FACTOR

Audio amplifiers, with a few very special exceptions, approximate to perfect voltage sources in that they aspire to have zero output impedance across the audio band. The result of this that the amplifier output is unaffected by by the load, so that the frequency-variable impedance of loudspeakers does not give an equally variable frequency response.[30]

The damping factor is calculated using equation B.4 where $R_{\text{load}}$ is the impedance of the speaker and the $R_{\text{out}}$ is the output impedance of the amplifier.[30]

$$Damping\ factor = \frac{R_{\text{load}}}{R_{\text{out}}} \quad (B.4)$$

But what does a high damping factor mean. Take a speaker if you were to give a short signal in to the speaker and have a low damping factor the cone of the speaker would continue to resonate for a while as the cone in the speaker has mass and does not stop immediately. However if you were to have a $R_{\text{out}} = 0.5\Omega$ and a speaker with $R_{\text{load}} = 8\Omega$ you would have a damping factor of 160. This would imply that the impulse had been dampened by 160 however this is not the case as the resonance of a loudspeaker unit depends on the total resistance in the circuit.[30]
The total series resistance is the sum of the speaker resistance, the cables and the amplifier output impedance. The values will be typically 7.0, 0.5, and 0.05Ω, so the amplifier only contributes 0.67% to the total, and its contribution to speaker dynamics must be negligible.[30]

Comparing the 160 damping factor to a 1600 one seems like a big step as it is a 10 bigger however the human ear would not hear the difference in the signals. The output impedance in a class D amplifier if usually done through negative feedback and cannot be controlled.[30]

### B.8 Crossover distortion

As can be seen in figure B.7 there is a dead zone when the sine wave passes through 0v. This is called crossover distortion. This happens because the transistor which are BJT transistors need the base to be above 0.7v or below -0.7v depending if they are NPN or PNP transistors. This only happens when transistors are used in a push pull configuration.[10]

![Figure B.7: An example of the effects of crossover distortion on a clean input signal](image)

There are a few ways to combat this. One way is to bias the transistors so that they are open a little longer than 50% of the time, that configuration is called an class AB amplifier, see figure B.8. In this configuration the two diodes are used to negate the voltage drop over the base-emitter.[10]

![Figure B.8: Circuit example of counter measurement against crossover distortion](image)

The resistors in figure B.8 are used to bias the diodes so Q1 and Q2 don’t experience a crossover distortion.[11]
B.9 Pulse width modulation

PWM is the basis for the class D amplifier to ensure that the mosfet transistors work only at the points of fully open and fully closed. The basis of the PWM is to use a type of an operational amplifier called a comparator. A comparator compares the signal on one input to the other and swings the output from V+ to V- depending on which input is higher.[11]

As can be seen in figure B.9 on one input the audio signal is fed while on the other is a saw tooth signal. The saw tooth signal is well above the audio signal on the audio spectrum. The saw tooth is typically over 100 kHz and can go as high as 1 MHz.

In figure B.10 an example of the output from the comparator is shown. It visually describes how the saw tooth signal changes the audio signal in to a PWM signal.[11]

B.10 Filtering

In class D amplifier when the signal has been amplified the signal is still a square wave this is not good as a square wave has a lot of unwanted frequencies. This means that the signal has to
be filtered before if can be sent to the speaker. The filter used to filter out the amplified signal is a low-pass filter, meaning low frequencies pass through but high frequencies are filtered out. The frequencies that are filtered out are dependent on the design of the filter.

In the case of the class D amplifiers the saw tooth signal used is often set high or in the 100 kHz or above so that the signal can be filtered better out. For the circuit chosen in this project is a second order LC filter where L stand for C stands for capacitors. The TAS5613 has a 400kHz $F_{PWM}$. By using a Z transformation a transfer function can be determined, see equation B.5.[11]

\[
\frac{V_{out}}{V_{in}} = \frac{s(C_3R) + 1}{s^3LR(C_1C_3 + C_2C_3) + s^2L(C_1 + C_2 + C_3) + s(C_3R) + 1}
\]

(B.5)

As can be seen in figure B.11 the filter cut of frequency is set at 70 kHz. The sharp spike should not have an effect on the sound quality as it is far beyond the human hearing.\(^2\) Figure B.11 also shows that the 400 kHz frequency has gone down -30dB.

![Bode Diagram](image)

Figure B.11: Bode plot for the output filter of the TAS5613

---

**B.11 Heat sink**

A heat sink is a way to transfer heat from an amplifier or other devise by increasing the area which comes into contact with the surrounding air. This is needed as a lot of circuits waist a lot

\(^2\)See chapter B.5 for details on bandwidth.
of power in to heat and are unable of riding themselves of it. Equation B.6 is used to calculate the size of the heat sink. First there is $T_J$ which is the maximum temperature that the junctions in the circuit can handle before being destroyed.[34]

$$T_J = P(R_{JC} + R_{CH} + R_{HA}) + T_A$$ \hspace{1cm} (B.6)

Next there is $R_{JC}$ or Thermal Resistance Junction to case which is given in $C^\circ/W$ in the datasheet. $C^\circ/W$ stands for how many degrees the heat goes over ambient temperature or $T_A$ which is 25$^\circ$ Celsius. $R_{JC}$ stands for the thermal resistance from the junctions to the case. The $R_{CH}$ is the thermal resistance from the case to the heat sink. There is a possibility to lower the $R_{CH}$ using thermal paste or to use so called Mica sheets which have to ability to lower the $R_{CH}$ and insulate the case to the heat sink electrically.[34]

The $R_{HA}$ stand for the thermal resistance between the heat sink and the air around it. The $R_{HA}$ can be lower by using fans to increase the flow of air around the heat sink. There are in the datasheets for heat sinks tables for calculations if fans are to be used in the design of the product. The $R_{HA}$ is also a good indicator on how big a heat sink is. P stands for maximum power planned to be used in a circuit. [34]

As can be seen in appendix C.1 the thermal resistance required from the heat sink needs to be equal or lower than 2.196$C^\circ/W$.[34]

**B.12 Sigma delta**

Sigma delta is a type of ADC or DAC. Sigma delta is often shown as the greek letters $\Sigma - \Delta$. $\Sigma - \Delta$ works on the idea of sampling the signal on a 1 bit resolution on a high frequency, a method called over sampling.

Next in the $\Sigma - \Delta$ process is to take a fixed number of bits and averaging them out. Once this has happened the signal goes in to a digital filter which then sends the signal to the output, see figure B.12.[35]

![Sigma delta block diagram](image-url)
B.13 I2S

When I2S or inter-IC sound was introduced, it was to establish the standard of how audio would be handled on new devices that were coming on the market in order to avoid communication problems. I2S uses three lines to send audio data from transmitter to receiver.

Continuous serial clock (SCK) which is the clock sent from the master. The Word select (WS) line is used to tell if the right or left channel is being transmitted, if one is sent on WS it means that right channel and zero for the left channel. Finally, there is the Serial data or (SD) channel which is for the data being sent. The Works on low voltages as can be seen here below, however, they can all be used on TTL level.[37]

- Outputs
  - $VL < 0.4V$
  - $VH > 2.4V$

- Inputs
  - $VIL = 0.8V$
  - $VIH = 2.0V$

An example of these signals can be seen in figure B.13 these signals will be compared to the final product.

![Figure B.13: I²S signal example][16]

B.14 I2C

I²C which is also known as TWI or Two Wire Interface was developed by Philips in 1982. It was designed for communication on 100 kHz for 7 bit addresses in 1992 however the 7 bits where expanded to 10 bit addresses and the communication speed was increased to 400 kHz. Today the speed can go as high as 5MHz. I²C was designed so many devices on the same bus that uses only 2 wires. SCL or Serial Clock Line is the clock from the Master. Having only one clock helps keep cost down and the communication failure low.

SDA or Serial Data Line is the wire that delivers the data between the master and slave. The I²C communication starts by sending out 8 bit packages from the master to the slave, the first 7 bits are the address and the eighth is the acknowledge bit sent by the slave how has the address. After the slave has acknowledged the master is starts sending out the data package. The package is sent out with the MSB first. See figure B.14 for I²C communication. [38]
The $I^2C$ then sends out or receives more packages until the stop condition is met. The stop condition is when SDA goes from high to low after having received low from SCL. In case of 10 bit addresses the $I^2C$ sends out two packages the first packages sent out is $11110XYZ$, which is not a legal packages for 7 bit address. The $X$ is the MSB of the address, $Y$ is the eight bit of the address and $Z$ is the read or write bit.[38]

Pulse code modulation was originally developed in 1939 as a method for transmitting digital signals over analog communications channels. There are a number of different variants used based on different mathematical techniques for quantization this includes linear, logarithmic, and adaptive. This method was developed in 1939 by the English inventor Alec H. Reeves.[39]

Pulse-code modulation is a method used to digitally represent analog signals. It is the standard that the TOSLINK uses and is thus widely used in modern systems. In a PCM signal stream the analog signal is sampled at a fixed interval, each sample is then quantized to the nearest value depending on the bit depth. The bit depth is how many bits are used to express how many digital values the signal can have for a 16 bit signal there are 65536 different values and in a 24 bit signal there are 16,777,216 values.

The next thing in PCM is the sampling rate. The sampling rate is used inorder to express the time between the each sample. According to the Nyquist theorem the sampling frequency should be at least twice the highest frequency in order to avoid aliasing the signal. see equation B.7.

$$f_s \geq 2 \times f_c$$

(B.7)

The most common way for quantitating the signal audio signal is the linear pulse-code modulation where the quantization levels are linearly uniform. Though PCM is a more general term, it is often used to describe data encoded as LPCM. [40]
Figure B.16: Example of PCM signal [41]
Appendix C

Calculations

C.1 Heat sink

Table C.1: Values for the heat sink from the datasheet of TAS5613 [18]

<table>
<thead>
<tr>
<th>Merki</th>
<th>Gildi</th>
</tr>
</thead>
<tbody>
<tr>
<td>$R_{\Theta JC}$</td>
<td>2,1°C/W</td>
</tr>
<tr>
<td>$T_J$</td>
<td>125,0W</td>
</tr>
<tr>
<td>$T_A$</td>
<td>25</td>
</tr>
<tr>
<td>$R_{\Theta CH}$</td>
<td>1.02°C/W</td>
</tr>
<tr>
<td>$P$</td>
<td>260W</td>
</tr>
<tr>
<td>$P_{Channel}$</td>
<td>130W</td>
</tr>
</tbody>
</table>

When choosing a heat sink it is imperative to know what power needs to be dissipated. For these calculations I will use assume that the power per channel is 130w in to 4Ω, however there are 2 channels per IC so the total power is 260w. The effeminacy of the amp is around 90%. So the power of heat that needs to be dissipated is 26w, see formula C.1.[34]

$$26w = 260w \times 0.1$$ (C.1)

Equation C.2 are all the values form tabel C.1 in to formula for heat sink calculations.

$$125 = 26(1.1 + 1.02 + R_{HA}) + 25$$ (C.2)

Equation C.3 is the heat sink formula converted for $R_{HA}$ without using thermal paste. Thermal paste is used to lower the $R_{\Theta CH}$. [34]

$$R_{HA} = \frac{125 - 25}{26} - (2.1) - 1.02 = 0.646°C/W$$ (C.3)

Equation C.3 is the heat sink formula converted for $R_{HA}$ using thermal paste. As can be seen the thermal paste lowers the $R_{\Theta CH}$ by 70%. [34]

$$R_{HA} = \frac{125 - 25}{26} - (2.1 \times 0.3) - 1.02 = 2.196°C/W$$ (C.4)
APPENDIX C. CALCULATIONS

C.2 Output filter

The following calculations are made for the output of the TAS5613. Equation C.11 is then fed in to matlab in order to create a bode plot.[42]

- \( R = 3.3 \Omega \)
- \( L = 7 \mu H \)
- \( C_1 = 680nF \)
- \( C_2 = 1nF \)
- \( C_3 = 10nF \)

\[
Z_1 = \frac{1}{sC_1} \quad (C.5)
\]
\[
Z_2 = \frac{1}{sC_2} \quad (C.6)
\]
\[
Z_3 = R + \frac{1}{sC_3} \quad (C.7)
\]
\[
Z_X = \frac{1}{\frac{1}{Z_1} + \frac{1}{Z_2} + \frac{1}{Z_3}} \quad (C.8)
\]
\[
Z_L = sL \quad (C.9)
\]
\[
Z_{heild} = Z_L + Z_X \quad (C.10)
\]

\[
\frac{V_{out}}{V_{in}} = \frac{s(C_3R) + 1}{s^3LR(C_1C_3 + C_2C_3) + s^2L(C_1 + C_2 + C_3) + s(C_3R) + 1} \quad (C.11)
\]
According to figure 4.9 for a 4Ω speaker when a 2V sinus signal is set on the input it comes out as 44.5V. These numbers are then put in equation C.12.[23]

\[ Gain = 10 \log \left( \frac{V_{\text{out}}}{V_{\text{in}}} \right)^2 \]  

(C.12)

From equation C.12 we get a gain of 26.94 dB.
Appendix D

Circuit schematics
D.1 Audio input

Figure D.1: Circuit schematic overview for the input stage [43]
Figure D.2: Audio input schematic part 1. Inputs and 74HC00 circuit
Figure D.3: Audio input schematic part 2. DIR9001

Figure D.4: Audio input schematic part 3. PCM1793
Figure D.5: Audio input schematic part 4. Voltages inputs and filter circuit
D.2 Audio processing

Figure D.6: Audio processing schematic part 1. uPC1892
D.2. AUDIO PROCESSING

Figure D.7: Audio processing schematic part 2

Figure D.8: Audio processing schematic part 3
D.3 Amplifier

Figure D.9: TAS5613 evaluation board schematic
D.4 Power supply

Figure D.10: project power supply circuit
Appendix E

PCB layout
E.1 Audio input

- **Note T1:** This design was made for prototyping purposes to simplify the design.
- **Note T2:** Another PCB will be designed and made for the final product.

Figure E.1: Audio input PCB top layout
Figure E.2: Audio input PCB bottom layout
Figure E.3: Audio input PCB layout. Analog output and low pass filter
E.2 Audio processing

Figure E.4: Audio processor PCB layout

- **Note T1:** This design was made for prototyping purposes to simplify the design.
- **Note T2:** Another PCB will be designed and made for the final product.
E.3 Amplifier

Figure E.5: TAS5613 evaluation PCB top layout[18]

- **Note T1:** PVDD decoupling bulk capacitors C60-C64 should be as close as possible to the PVDD and GND X pins. The heat sink sets the distance. Wide traces should be routed on the top layer with direct connection to the pins and Without going through vias. No vias or traces should be blocking the current path.[18]

- **Note T2:** Close decoupling of PVDD with low impedance X7R ceramic capacitors is placed under the heat sink and close to the pins.[18]

- **Note T3:** Heat sink needs to have a good connection to PCB ground.[18]

- **Note T4:** Output filter capacitors must be linear in the applied voltage range preferable metal film types[18]
• **Note B1:** It is important to have a direct low impedance return path for high current back to the power supply. Keep impedance low from top to bottom side of PCB through a lot of ground vias.[18]

• **Note B2:** Bootstrap low impedance X7R ceramic capacitors placed on bottom side providing a short low inductance current loop.[18]

• **Note B3:** Return currents from bulk capacitors and output filter capacitors.[18]
E.4 Power supply

Figure E.7: Power supply circuit used for evaluation

- **Note 1:** This circuit was created as an evaluation power supply as the transformer for this design did not arrive on time and will not be used for the final product.

- **Note 2:** The final design PCB was not designed as the transformer did not arrive in time.
## Appendix F

**Bill of materials**

Table F.1: Bill of materials

<table>
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<th>Surround</th>
<th>Power supply</th>
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