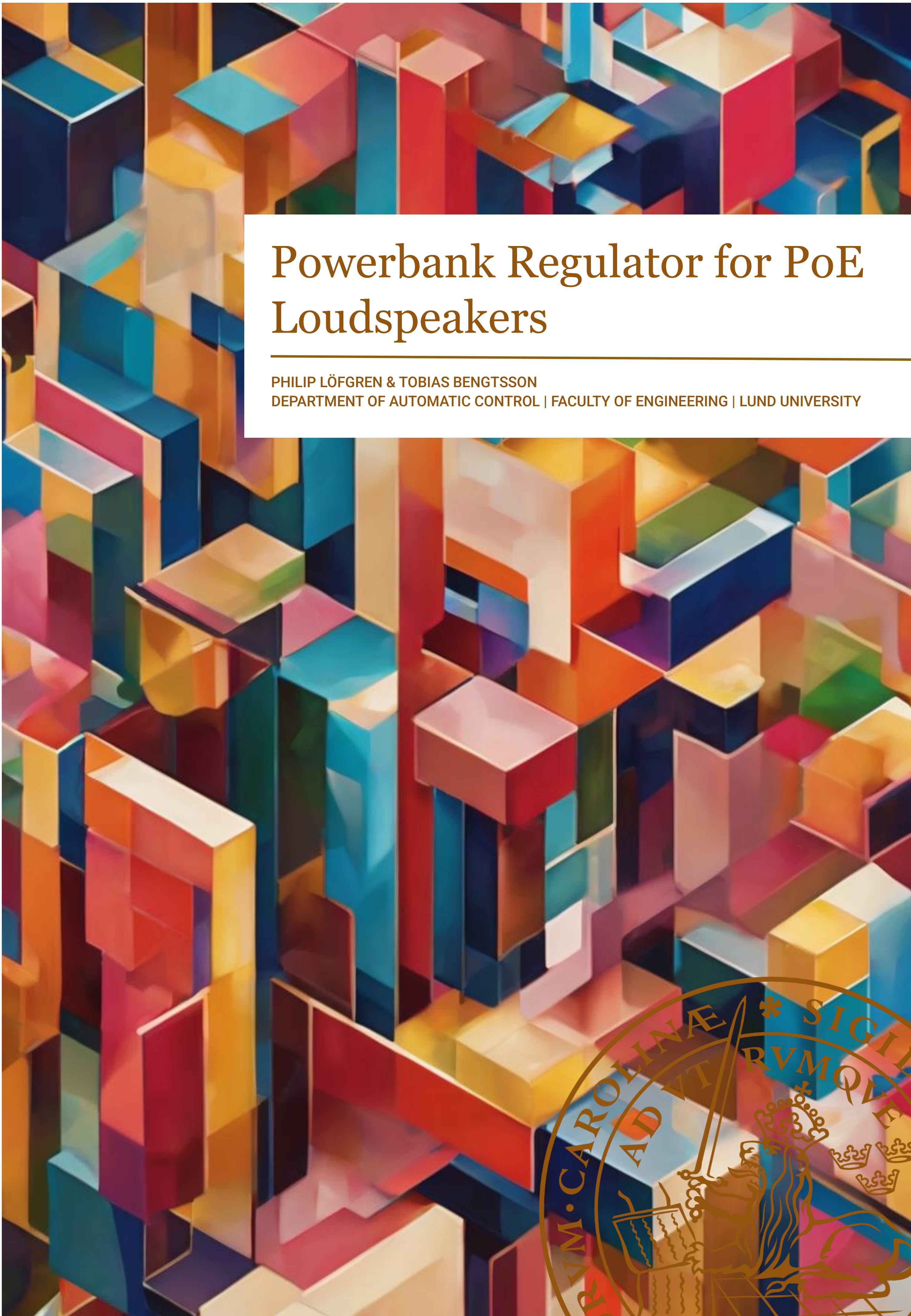


Powerbank Regulator for PoE Loudspeakers

PHILIP LÖFGREN & TOBIAS BENGTSSON
DEPARTMENT OF AUTOMATIC CONTROL | FACULTY OF ENGINEERING | LUND UNIVERSITY

Master Thesis

Powerbank Regulator For PoE Loudspeakers

Spring 2024

Philip Löfgren & Tobias Bengtsson

Abstract

This master's thesis project explores the development and implementation of a powerbank for Power over Ethernet (PoE) driven loudspeakers, addressing the inherent limitations of output power imposed by PoE Type 1. PoE, while advantageous for simplifying installations and reducing wiring costs, restricts the maximum power delivery to connected devices. This constraint poses significant challenges for audio applications, particularly in handling high amplitude peaks without causing clipping and distortion.

To overcome these limitations, the thesis introduces an innovative solution involving a supplementary powerbank added to the loudspeaker's amplifier. The powerbank is intended to provide additional power during peak audio demands, ensuring consistent and high-quality sound output. A critical component of this system is the Power multiplexer (MUX), which intelligently selects the power source, either the PoE or the powerbank, based on real-time analysis of the audio input.

An algorithm developed for the Power MUX monitors the audio signal, predicting power needs and seamlessly switching to the powerbank when the input data indicates potential clipping or power insufficiency. This adaptive power management strategy not only mitigates the limitations of PoE but also enhances the overall performance and reliability of PoE-driven loudspeakers.

The prototype and testing demonstrate the effectiveness of this approach, highlighting its potential in various environments where high-clarity audio and simplified installation are paramount. This thesis contributes to the advancement of PoE technology in audio systems, offering a viable solution to extend the capacity of PoE-driven loudspeakers beyond their conventional power constraints.

Acknowledgements

We, the authors of this thesis, would like to extend our sincerest gratitude to Axis Communication Lund for allowing us the opportunity to write this thesis and providing valuable support, expertise and equipment all throughout the semester. A special thanks to our supervisors at Axis, Eric Liljeblad and Christoffer Forsberg, for always taking the time to assist, discuss and answering questions. We would also like to thank our academic supervisor Charlotta Johnsson for providing valuable guidance and support throughout the project and Anders Dunkars at Texas Instrument for providing both insight and components.

Contents

Acronyms

AP	Audio Precision
BMS	Battery Management System
CDA	Class-D Amplifier
CCCV	Constant Current - Constant Voltage
DAC	Digital to Analog Converter
dB	Decibel
DSP	Digital Signal Processing
EVM	Evaluation Module
IP	Internet Protocol
IoT	Internet of Things
MOSFET	Metal-Oxide-Semiconductor Field-Effect Transistor
MUX	Multiplexer
PAPR	Peak-to-Average Power Ratio
PCB	Printed Circuit Board
PD	Powered Device
PoE	Power over Ethernet
PSE	Power Sourcing Equipment
PWM	Pulse Width Modulation
RMS	Root Mean Square
SFC	Sequential Function Chart
SMPS	Switched-Mode Power Supply
SOC	State Of Charge
SoC	System on Chip
SPL	Sound Pressure Level
THD	Total Harmonic Distortion
THD+N	Total Harmonic Distortion plus Noise

1 Introduction

In recent years, Power over Ethernet (PoE) technology has emerged as a potent solution for delivering both power and data over a single Ethernet cable. This innovative technology has found widespread application in various fields, from networking devices to security cameras, and, more excitingly, to audio systems. PoE offers the advantage of simplifying installation, reducing costs, and providing flexibility in the deployment of networked devices, including speakers.

The integration of PoE technology into speaker systems opens up new possibilities for audio distribution, both in residential and commercial environments. Traditionally, speaker systems have relied on separate power sources and audio sources, often resulting in complex wiring schemes and limitations in placement. With PoE, these limitations are alleviated, as speakers can now be powered and controlled through the same Ethernet cable used for data transmission.

However, since PoE is a relatively new technology the power delivery has not yet fully caught up with the demands of most commercial powered equipment and it is often not able to deliver as much power as a standard system would. And as it is well known; with not so great power comes not so great responsibility, and thus, audio PoE systems in particular are not given the opportunities they might deserve.

Therein lies the motivation behind this project, to explore the possibility to increase the available output power of a PoE powered loudspeaker without stepping up to a newer class of PoE which can supply more power. Instead, the loudspeakers will be supplied with additional power by an ”external” powerbank, such as a battery, which is recharged during time periods where the loudspeaker demands are lower than the available PoE power.

To begin with, the structure of a PoE system was explored. This was done in order to find where power was lacking and where more power could easily be delivered, and most importantly, how it could be delivered without compromising any functionality or performance. The basic structure of a PoE system is shown in Figure 1 and in principle it functions as follows: PoE can deliver both data and electrical power via an Ethernet cable. The power delivered by the PoE to the powered device (PD) is usually converted to a desired voltage as well as isolated and regulated to ensure proper functionality. The digital data signal, in this case an audio signal, is delivered to a DSP which processes the audio signal to desired levels and converts it from a digital signal to an analog. The analog signal is then delivered to an amplifier, which within the scope of this work will from now on be a Class D Amplifier, but in reality it could be any amplifier class. The amplifier is powered from the PoE and amplifies the analog audio signal, which is then delivered to a speaker.

It is also important to note that not all power available at the PD side of the PoE is directly used by the speaker system. There are usually many more subsystems that require power, such as processing units, so the total power available to the amplifier and speaker is substantially lower than the PoE can deliver.

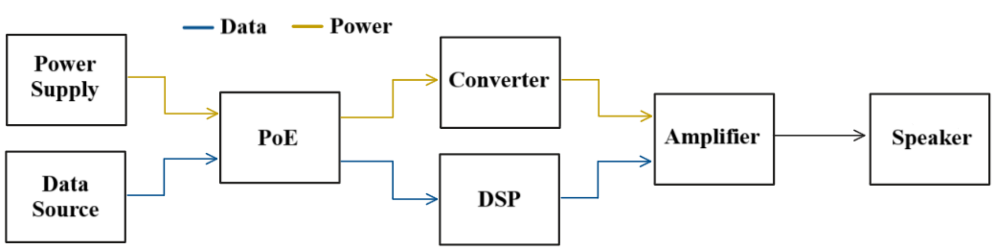


Figure 1: A simplified block schematic of the different components in a Power over Ethernet speaker system. A digital data signal is sent through the PoE along with power from a power supply. The power is generally converted via a flyback converter and delivered to any power demanding units in the PD, in the figure this is the amplifier, while the digital data is sent to a DSP which performs digital filtering and later converts it to an analog signal. The amplifier receives both power and the analog signal, and amplifies the signal which it then delivers to the speaker.

2 Methodology

2.1 Double Diamond

This thesis work will be carried out according to the "Double Diamond" methodology, which consists of two phases and four sub-phases. The two phases can be described as "solve the right problem" and "solve the problem right". These phases are each divided into two sub-phases where the first is meant to research and broaden the view on a topic and the second is for focusing the work towards a more specific path and solution. The four sub-phases are often called "Discover", "Define", "Develop" and "Deliver". The first sub-phase, "Discover", will be used to explore the subject by studying the underlying topics that form the basis of the work, such as speaker systems, PoE systems, power consumption in amplifiers, types of powerbanks, power combining, etc. The next sub-phase, "Define," aims to identify the primary problem based on the insights gained in the previous sub-phase and determine which specific issue the work will address. In the next sub-phase "Develop", perspectives are again broadened and potential solutions to the problem are explored and researched. All potential solutions are then evaluated and the most promising one is then tested and refined in the final sub-phase of the work to ultimately "Deliver" a final solution. [1]

As such, the report will be structured accordingly. The next part of the report is theoretical and informative in order to explore where any power related problems can arise and which can be solved by the addition of a powerbank. It is also meant to describe the purpose and behavior of the individual components of a PoE system and speaker system for the reader. It offers a brief description of each subsystem, advanced enough for the reader to comprehend wherein the "faults" and "problem" lies, as well as to help understand how the presented solution is intended to improve future products.

The subsequent section will then define what problem the work will intend to solve and what objectives and restrictions are considered, allowing the section after that to explore potential solutions to the problem. The reader then be introduced to components and subsystems in order to comprehend the final solution, which will be presented and evaluated in the final sections.

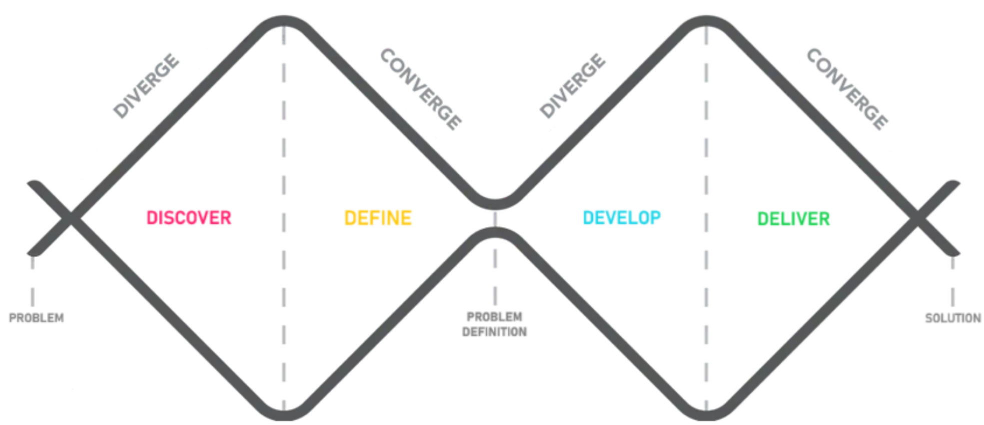


Figure 2: Illustration of the Double diamond method with the different sub-phases. The broadening of the sub-phases "Discover" and "Develop" symbolizes a wider perspective, and the narrowing of the "Define" and "Deliver" sub-phases symbolizes focusing and commitment.

3 Theory

3.1 Power over Ethernet

Power over Ethernet (PoE) has since the first standardization by IEEE in the year 2003 emerged as a transformative technology, revolutionizing the way power network devices are developed and deployed. It facilitates the simultaneous transmission of data and electrical power over a single Ethernet cable, simplifying infrastructure and providing greater flexibility in device placement. PoE is widely employed in applications such as IP cameras, wireless access points, VoIP phones, speaker systems and IoT devices [2].

PoE functions by injecting a low-voltage direct current (DC) into the Ethernet cable alongside the standard data signals. This enables network devices to receive both power and data through an unified cable, eliminating the need for separate power sources. The primary components of a PoE system include the Power Sourcing Equipment (PSE), such as PoE switches or injectors, that deliver power to connected Powered Devices (PD) [2].

PoE leverages the standard twisted pair cables used in Ethernet networks for power delivery, see Figure 3. The power is transmitted over two pairs of wires in the Ethernet cable, either in unison with the data in pairs 1/2 and 3/6 or separately in the spare pairs 4/5 and 7/8 [2]. The power transfer method used and amount of power delivered depends on the specific PoE standard and class, with higher classes capable of delivering more power to meet the diverse needs of various devices.

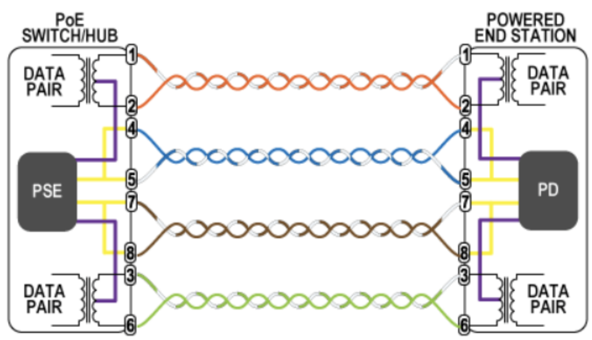


Figure 3: The interface between two Ethernet equipment. Figure from [3].

To ensure interoperability and consistent performance, various PoE standards have been established. The **IEEE 802.3af-2003**, **802.3at-2009** and **802.3bt-2018** standards are widely adopted, defining the maximum power that can be delivered over the Ethernet cable, see Table 1. All PoE compatible PDs can be classified according to a standardized format, ranging from class 0 to 8, which discloses its power requirements [4][5]. In order to determine the PD class and PoE type the PSE establishes a negotiation process between itself and the PD. The negotiation is initiated by a detection phase, with the PSE sending a gentle voltage across the Ethernet cable to ascertain the presence of a PD which signals its existence as well as its power requirements by responding with a signature resistance. Several more stages occur in the negotiation between the PSE and PD to configure and setup the power delivery. Once power begins to flow, continuous monitoring occurs by both PSE and PD keeping a vigilant eye on power parameters, ensuring stability and compliance with the negotiated terms. If there are changes in the PDs power requirements, the negotiation process repeat to accommodate those changes, or abort power delivery entirely.

These systems are more than just classification processes, but also necessary safety features to protect devices and users. Detection mechanisms ensure that only compatible PDs receive power, preventing damage caused by incompatible devices and limits the current supplied during malfunctions, safeguarding devices from potential damage.

Table 1: Overview of PoE Types. Data from [5].

	PoE	**PoE+**	**PoE++**
IEEE Standard version	802.3af (802.3at Type 1)	802.3at Type 2	802.3bt Type 3
Voltage range (at PSE)	44.0–57.0 V	50.0–57.0 V	50.0–57.0 V
Voltage range (at PD)	37.0–57.0 V	42.5–57.0 V	42.5–57.0 V
Maximum power from PSE	15.4 W	30 W	60 W
Power available at PD	12.95 W	25.50 W	51 W
Twisted pairs used	2	2	4

3.2 Flyback Converter

The voltage appearing at the PoE output, the PSE side, has to be converted and regulated in order to ensure reliable and safe power delivery to the connected PD. One commonly employed power converter in PoE applications is the flyback converter [6]. The flyback converter is a type of switched-mode power supply known for its simplicity, cost-effectiveness, and versatility. Its operation involves energy storage in a transformer during the switch-on time and subsequent release of stored energy during the switch-off time. This distinctive energy transfer mechanism allows for voltage regulation, galvanic isolation between primary and secondary side, and ease of implementation in PoE systems.

A simplified topology of a flyback converter is shown in Figure 4. It functions around oppositely polarized inductors to transfer current between the primary and secondary side. When the transistor, Q, is open, the voltage, V_{in}, will be applied to the primary side. Since the voltage is constant the current, i_M, will linearly increase to its peak value. At the secondary side, since the inductors are of opposite polarity and due to the presence of the diode, no current, i_s, will flow. The capacitor, C, however, will store charge and produce a current, i_C, which flows through the load, R_L. When Q is turned off, the current, i_M, will continue to flow but will do so through the only possible path, through the primary inductor, L_p. This will induce a current in secondary inductor, L_s, which will flow through the diode, the capacitor, C, and the load, R_L. In reality, the inductor, L_M, does not exist and the current, i_p, will instead flow and store energy in the air gap and magnetic core of L_s, which is released upon switching off Q.

In order to find the output voltage of the flyback converter one has to start with the magnetizing inductor current. The voltage equation for an inductor is

$$V = L\frac{\Delta i}{\Delta t} \iff \Delta i = \frac{V}{L}\Delta t \tag{1}$$

Where Δi is the change in current, Δt the change in time, L is the inductance of the inductor and V the voltage across the inductor. According to above, when the the transistor, Q, is turned on the current through the inductor will be

$$Q_{on} : \Delta i_{on} = \frac{V_{in}}{L} D_{on} T \tag{2}$$

Where D is the duty cycle and T is the total time. Similarly when the switch is off, the current will be

$$Q_{off} : \Delta i_{off} = \frac{\frac{V_o}{n}}{L} D_{off} T \tag{3}$$

where n is the turns ratio between the primary and secondary side inductor

$$n = \frac{N_s}{N_p} \tag{4}$$

With N_s as the number of turns of the secondary side inductor and N_p the turns of the primary inductor. Now an average steady state assumption can be made, which entails that the ripple on the positive side is the exact same as the ripple on the negative side, the changes in current has to equal zero in order for the current not to increase over time.

$$\Delta i_{on} + \Delta i_{off} = 0 \tag{5}$$

now, plugging in Equation (2) and (3) gives

$$\frac{V_{in}}{L} D_{on} T + \frac{\frac{V_0}{n}}{L} D_{off} T = 0 \Longrightarrow V_{in} D_{on} = \frac{V_0}{n} D_{off} \Longleftrightarrow V_o = V_{in} \frac{D_{on}}{D_{off}} \tag{6}$$

Which then shows that the output voltage is determined by the duty cycle of the switch, Q, as well as the input voltage.

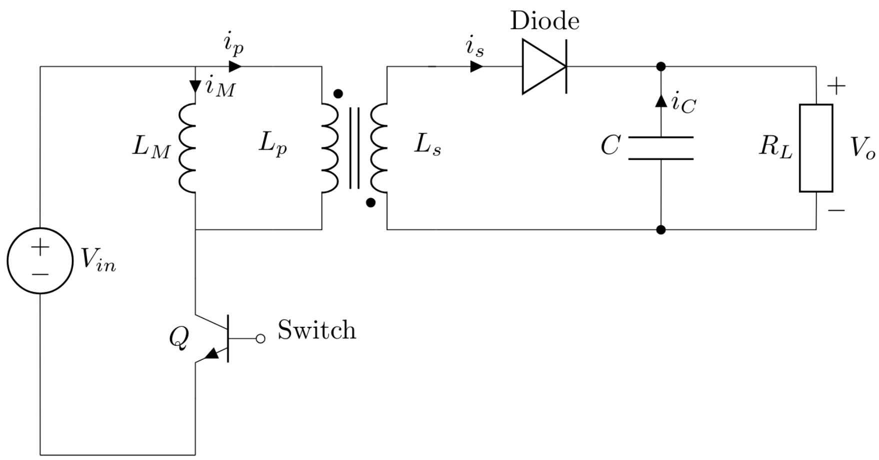


Figure 4: Schematic of a Flyback Converter.

3.3 Digital Signal Processing

After digital data has been transferred to the PD side through the Ethernet cable it has to be handled properly, which is commonly referred to as Digital Signal Processing (DSP). The DSP can have multiple functions and perform several operations depending on the application. In audio applications a DSP can perform tasks such as sample rate conversion, volume/gain control, bit depth adjustment, clocking and synchronization, compression and data formatting. Exactly what happens in a DSP varies even within the same application and it is therefore difficult to precisely define the role of a DSP in all audio systems. Several important functions are however very relevant to this project and will thus be described below.

3.3.1 Dynamic Range Compression

Dynamic range is the difference between the largest and smallest amplitudes in a signal. Compression of the dynamic range means to amplify the weak parts of the signal while decreasing the peaks. In audio applications this serves the purpose of equalizing the volume of the output to avoid some sounds overpowering others as well as increasing the overall amplitude of the signal [7]. Nonetheless, if the compression is too aggressive, the signal will become flat, resulting in the sound losing its natural dynamic and appearing as ”boring” or ”inauthentic” [8].

A common way to measure dynamic range is with crest factor, or in extension, Peak to Average Power (PAPR). Crest factor is the value of the peak power of a signal divided by the Root Mean Square value

(RMS), or the difference between the two in decibel, see Figure 5, while PAPR is the square of the crest factor [9].

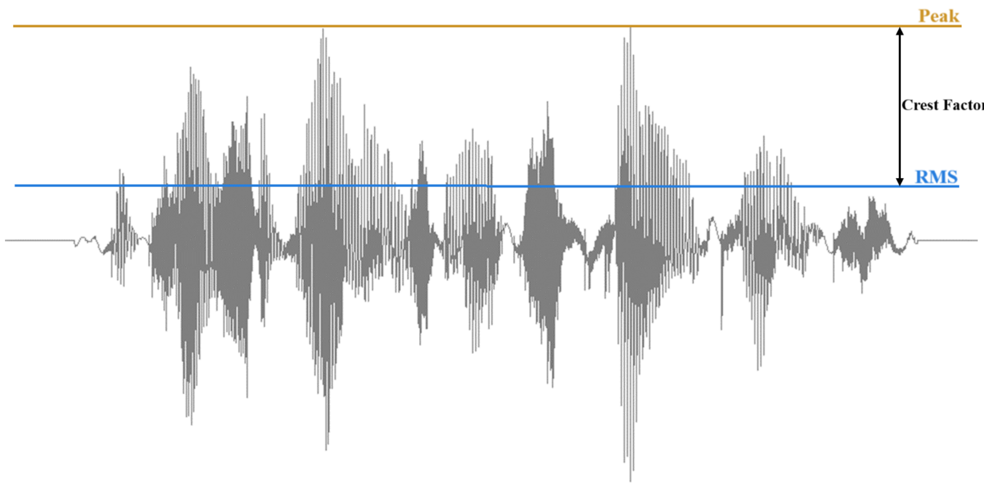


Figure 5: Crest factor, the difference between peak amplitude and RMS.

3.3.2 Volume/Gain Control

Before leaving the DSP and the following Digital to Analog Converter (DAC), the amplitude of the signal must be adjusted according to the specified requirements. When discussing the amplifier in the upcoming sections, this step is vital for ensuring proper signal reconstruction post amplification. This is due to the need to compare the signal with a generated triangle wave, ensuring that the signal's amplitude does not exceed that of the triangle wave, as will be explained below.

3.4 Class-D Amplifier

This work will only consider Class-D amplifiers (CDA), this is because they are very common in low power applications. CDAs, also known as digital amplifiers or switching amplifiers represent a significant advancement in recent audio amplifier technology. Unlike traditional linear amplifiers, which operate by continuously varying input signals to produce amplified output, CDAs employ Pulse-Width Modulation (PWM) techniques to efficiently amplify audio signals. This approach offers several advantages in terms of efficiency, size, and power handling capabilities [10].

One of the key advantages of CDAs is their high efficiency. Traditional linear amplifiers dissipate a significant amount of power as heat, especially when amplifying high-power audio signals. In contrast, CDAs operate in a binary fashion, where the output transistors are either fully conducting or fully non-conducting, minimizing power dissipation and maximizing energy efficiency. In theory, the efficiency of CDAs is 100%, however due to non-ideal switching transistors the actual efficiency is typically around 80-90%. The high efficiency not only reduces power consumption but also allows CDAs to operate at higher power levels without overheating, making them ideal for applications where power efficiency is critical, such as PoE devices where the output power is very limited, or high power applications [10].

3.4.1 Pulse Width Modulation

To begin understanding how a CDA works, one has to start with the heart of it; the concept of Pulse-Width Modulation. PWM is a modulation method to where information is stored in the width of pulses in a pulse train. This is achieved by feeding a comparator with two signals, one is the original signal with information to be modulated, in this case an analog audio signal, and the other is a high frequency triangle waveform generated by a local oscillator. The comparator compares these two signals and outputs either "high" or "low", depending on which of the two inputs has the larger amplitude at that instant. If the audio signal is larger than the triangle wave, then the comparator outputs ”high”, and if the triangle wave is larger than the audio signal, the comparator outputs ”low”. Since the triangle wave is periodic, the amplitude of the audio signal will determine the length of time the comparator sustains one output, and thus, the amplitude of the audio signal controls the width (or more precisely, the duty cycle) of the pulses, effectively encoding the amplitude of the input signal [11], see Figure 6.

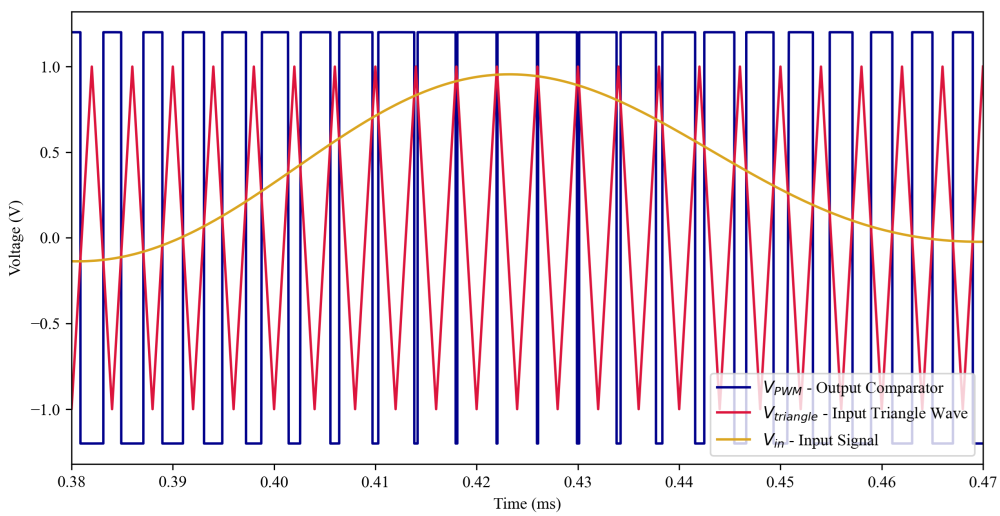


Figure 6: PWM signal. The audio signal, V_{in}, is compared within the amplitude of the triangle waveform, $V_{triangle}$. The comparator output, V_{PWM}, is a pulse train with varying duty-cycle depending on the amplitude of V_{in}.

3.4.2 Power Stage

The PWM signal, now representing the audio waveform in digital form, is then fed into a power stage consisting of two or four pairs of output transistors, often Metal Oxide Semiconductor Field Effect Transistors (MOSFETs), configured in a push-pull arrangement, see the simplified CDA circuit in Figure 7 below. These transistors rapidly switch between fully on (saturation) and fully off (cut-off) states in response to the PWM signal, amplifying and duplicating it, hopefully without distorting the pulses. These pulses then pass through an LC low pass filter which reconstructs the analog input signal by simply filtering out the high frequencies present in the pulses while preserving the low frequency audio signals encoded in the duty cycle. The instant output voltage can be described by the following relation [12]:

$$V_{out} = \frac{W_{pulse}}{T_{pulse}} V_{supply} \tag{7}$$

Where W is the width and T is the period. The fraction, $\frac{W}{T}$, is the duty cycle which will vary between 0% and 100%, and thus the output voltage will vary between 0 and V_{supply}.

While not depicted in Figure 7, most CDAs incorporate a feedback network connecting the output stage to the comparator. This feature aims to mitigate Total Harmonic Distortion (THD) caused by non-linearity and dead-time inherent in the open-loop configuration of the output stage. [11]

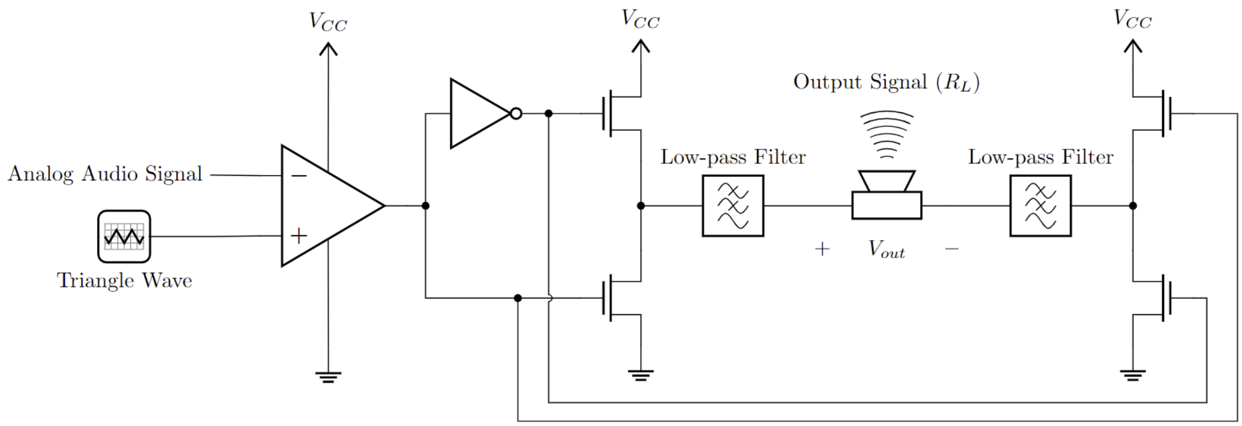


Figure 7: Simplified schematic of a Full-Bridge Class-D Amplifier.

3.5 Loudspeaker Anatomy

Loudspeakers as a concept are presumably well known to the reader, however some basics of loudspeakers has to be covered in order to comprehend future problems. In a nutshell, loudspeakers are devices that convert electrical signals into sound waves, in other words, an electroacoustic transducer. The basic elements required in an electroacoustic transducer is a motor, which converts electrical energy into mechanical movement, a diaphragm, which converts the mechanical movement into acoustic waves and a suspension which, as the name suggests, suspends the diaphragm and allows it to oscillate as desired [13], see Figure 8.

The most common type of transducer used today is an electrodynamic transducer [13], and it is the type of transducer of which this work will focus upon. In this type of transducer the motor consists of a tightly wound coil of wire, known as a voice coil, that sits within the magnetic field of a permanent magnet. When an electrical signal is applied, current flows through the voice coil, creating a varying magnetic field, which interacts with the static magnetic field supplied by the magnet, resulting in a time varying force that mechanically moves the voice coil.

Attached to the voice coil and following its movement is the diaphragm. The most common type of diaphragm used is a cone shaped paper diaphragm [13]. As the diaphragm moves, it compresses and rarefies the air in front of it, creating sound waves that correspond to the original electrical signal applied to the voice coil.

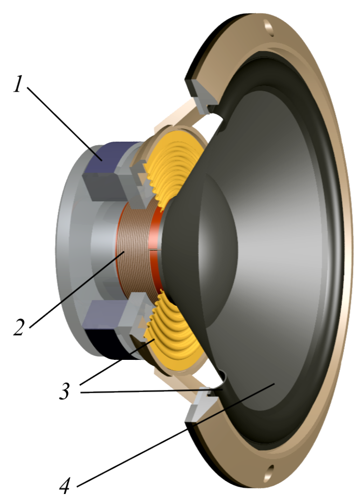


Figure 8: Cut through of an electrodynamic transducer with a cone shaped diaphragm. 1: The permanent magnet. 2: The voice coil. 3: The suspension. 4: The diaphragm. Image from [14].

The frequency of the electrical signal determines how fast the voice coil moves back and forth, thus controlling the pitch of the sound produced. The amplitude (or intensity) of the electrical signal controls the amplitude (loudness) of the sound waves. However, the signal will not be perfectly reproduced over the entire frequency spectrum because of the complex mechanics of the enclosed speaker system. Every mass-spring type of system has a resonant frequency, f_s, as will an electrodynamic transducer system. At this frequency the coil will vibrate at its full range and speed and generate its highest counter-electromotive force, the electrical impedance of the speaker will thus be at its maximum. Beyond this frequency the impedance will decrease rapidly until it reaches its local minimum from which the impedance will increase with frequency [15]. This behaviour can be understood by considering the voice coil as an inductor with its impedance formula as

$$Z = j\omega L = j2\pi f L \tag{8}$$

where L is the inductance of the coil and f is the frequency. The electrical impedance of an electrodynamic transducer can therefore be visualized as in Figure 9 below.

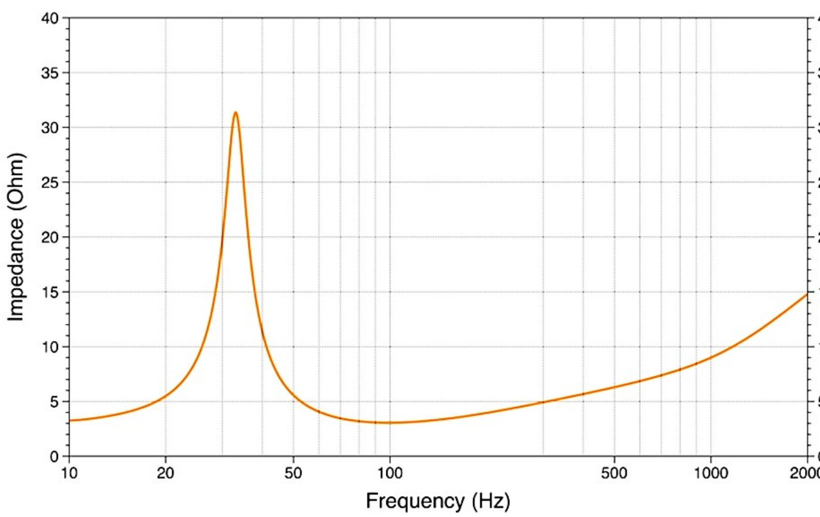


Figure 9: The usual appearance of the impedance versus frequency curve for an electrodynamic transducer. The peak is at the resonant frequency of the system, f_s. Image from [16].

3.6 Sound

A loudspeaker's purpose is to produce sound, and while the theory behind a loudspeaker's acoustic reproduction is relatively complex, the basics of a sound generation is quite straightforward. At its core, acoustic sound is the result of mechanical disturbances in a medium, typically air, which travel in the form of longitudinal waves. The disturbances are swift periodic fluctuations in pressure, both positive and negative, in relation to the static pressure, see Figure 10. As most periodic fluctuations, these waves exhibit characteristics such as wavelength, frequency, amplitude, and harmonic content, all of which contribute to the auditory experience [17]. It is beneficial to explain some of these physical concepts briefly in order to attain some understanding of figure of merits for speakers to eventually be able to evaluate the results of the project.

To begin with, the zones of pressure which serve as the foundation of sound can be measured in terms of Pascal (Pa). The very softest sound a human can perceive is in the order of 20 µPa, while the very loudest sounds, that easily shatters an eardrum, can reach beyond 20 kPa. It is terribly inefficient to use this linear number system for the intensity of sound. Instead, a logarithmic system is often used, known as the decibel (dB). A Bel, in words, is *the logarithm of the ratio of two powers*, or, more commonly, expressed as decibel:

$$\mathrm{dB} = 10 \log \frac{p^2}{p_0^2} = 20 \log \frac{p}{p_0} \tag{9}$$

Where p is the measured pressure and p_0 is the reference pressure. The two pressures are as mentioned in the unit of Pascal, and they are thus squared in order to be expressed as a power. Using two logarithmic laws, the square can be moved outside the logarithm to simplify the formula. When the softest perceivable sound, i.e 20 µPa, is used as the reference pressure and the measured pressure is expressed in its RMS value, the decibel measurement is known as Sound Pressure Level (SPL). SPL is a very common figure of merit for any speaker, often measured through the entire auditory frequency span, 20 Hz to 20 kHz, which provides a speaker's frequency response curve. A human can thus perceive sounds from 0 dbSPL up to and above the threshold for pain at around 140 dBSPL, where normal speech levels are at roughly 60 dBSPL [18], see figure 11.

Another integral figure of merit is Total Harmonic Distortion (THD). Acoustic sound from complex sources (such as speakers and amplifiers) often contains harmonics, which are additional frequencies that are integer multiples of the fundamental frequency, 2f, 3f, 4f, and so on. THD is a crucial metric in evaluating the fidelity, or purity, of audio reproduction systems. It quantifies the extent to which a system introduces harmonic components not present in the original signal. THD is expressed as a percentage, calculated by dividing the RMS of the harmonics by the RMS of the original signal:

$$\mathrm{THD} = \frac{\sqrt{V_{2f}^2 + V_{3f}^2 + V_{4f}^2 + ...}}{V_f} \tag{10}$$

THD is a non-standardized specification, the above is the IEEE definition, and is therefore often unreliable to be used in comparisons between manufacturers. But measurements performed in a the same setup and a closed system can give valuable information on the distortions. Another, more reliable, specification is Total Harmonic Distortion plus Noise (THD+N):

$$\mathrm{THD{+}N} = \frac{\sqrt{V_{2f}^2 + V_{3f}^2 + V_{4f}^2 + ... + N}}{V_f} \tag{11}$$

In this measurement the noise from system sources such as ground loop hum or outside sources such as high frequency interference is taken into account in addition to the distortion [19].

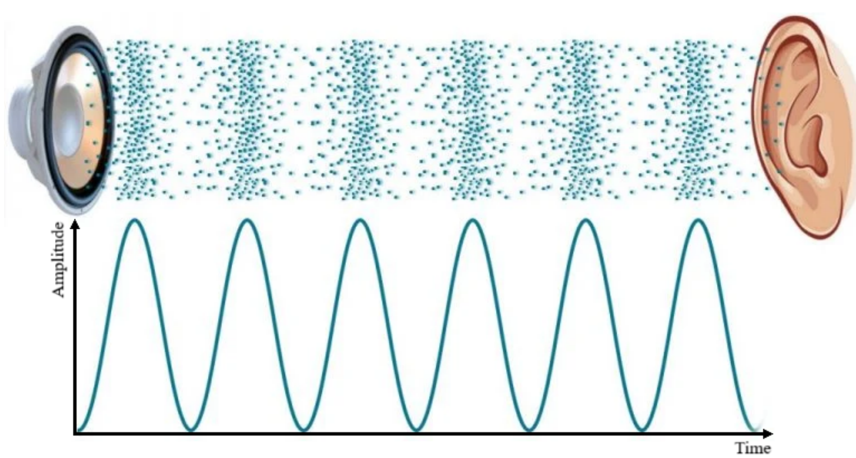


Figure 10: An interpretation of how sound can be visualized. A speaker produces periodic zones of high and low pressure which can be perceived the human auditory system. The pressure zones can be measured as waves of amplitude in time.

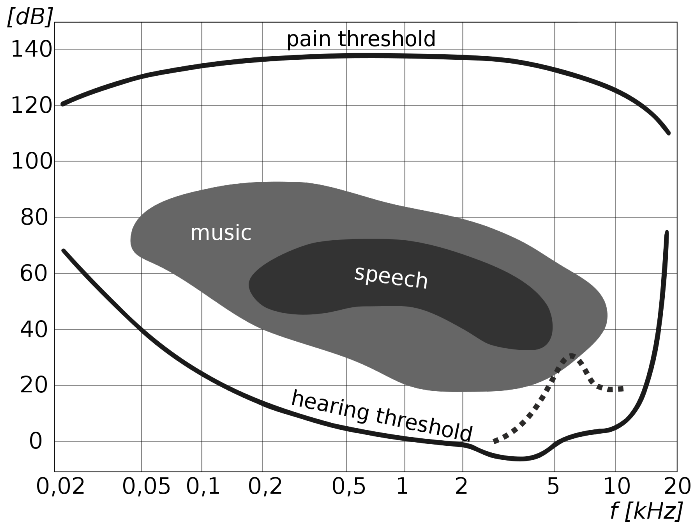


Figure 11: Human hearing range in frequency and intensity. Figure from [18].

3.7 Boost Converter

A Boost Converter (BC) or step-up converter is a type of DC-DC switching converter. It works by increasing the voltage from the input to the output while simultaneously decreasing the current to keep the power consumption constant.

A simplified schematic of a boost converter can be seen in Figure 12. It contains an inductor, a diode, a capacitor and a switch (implemented as a MOSFET in this case).

To step-up the voltage the transistor alternates between conduction and not. When the transistor is on, current will flow through the inductor and transistor, see Figure 13a. The transistor is then turned off, see left half of Figure 13b. Because the inductor, L, has an inherent tendency to resist changes in current it cannot abruptly stop the current flow. The response from the inductor is that a potential difference, v_L, is formed over the inductor in addition to the input voltage, V_{in}. This produces a current that flows through the diode to the capacitor, C, and load, R_L, and charge is stored in the capacitor. When the transistor is again turned on the diode will be reverse biased, which means that the capacitor can only discharge through the load, see right half of Figure 13a. [20], [21]. It is important to note that a boost converter will consume small amounts of power during operation, which must be taken into consideration.

The duty cycle, D, determines the partial time, T_{ON}, when the switch is on (short circuited) from the total time, $T = T_{ON} + T_{OFF}$ [22][23].

$$D = T_{ON}/T \tag{12}$$

In the first case, when the transistor is on, the diode can be considered to be an open circuit. The transistor will be on for a time of $T_{ON} = \Delta t = DT$ and if V_L is the voltage over the inductor L, Kirchhoff's Voltage Law gives:

$$V_L = L\frac{di_{in}}{dt} = L\frac{\Delta i_{in}}{\Delta t} = V_{in} \qquad \Rightarrow \qquad (\Delta i_{in})_{ON} = \left(\frac{V_{in}}{L}\right)\Delta t = \left(\frac{V_{in}}{L}\right)DT \tag{13}$$

In the second case, when the switch is turned off (open circuit). The total time when the switch is turned off is $T_{OFF} = \Delta t = (1 - D)T$. Neglecting the voltage drop across the diode, KVL gives:

$$V_L = L\frac{di_{in}}{dt} = L\frac{\Delta i_{in}}{\Delta t} = V_{in} - V_{out} \quad \Rightarrow \quad (\Delta i_{in})_{OFF} = \left(\frac{V_{in} - V_{out}}{L}\right)\Delta t = \left(\frac{V_{in} - V_{out}}{L}\right)(1-D)T \tag{14}$$

Over a complete cycle, the net charge of the inductor L must be equal to zero, which gives:

$$(\Delta i_{in})_{ON} + (\Delta i_{in})_{OFF} = 0 \quad \Rightarrow \quad \frac{V_{out}}{V_{in}} = \frac{1}{1-D} \tag{15}$$

From Equation (15) the output voltage from the BC can be controlled by the duty cycle D of the switch. Due to variations in load and input voltage over time, BCs typically incorporate a feedback mechanism that adjusts the duty cycle to maintain a consistent and steady output voltage. Be aware that these calculations disregard the voltage drop across the diode, which slightly alters the conditions.

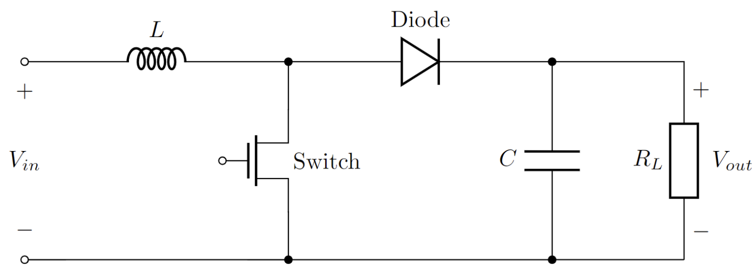
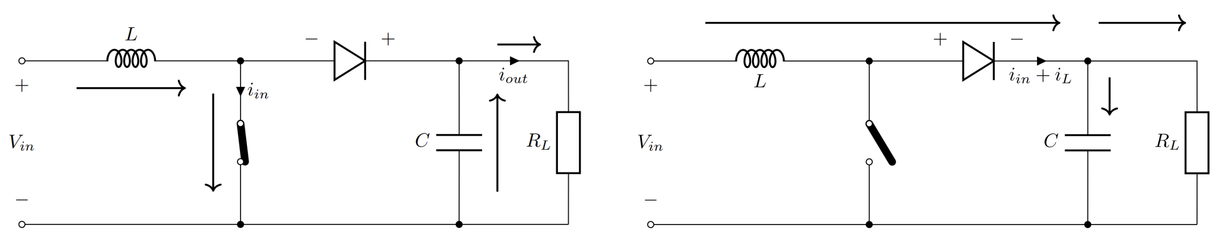


Figure 12: Schematic of a DC-DC Boost Converter. The boost converter performs a voltage step-up and simultaneously decreases the current.



(a) Closed: Current flowing from v_{in} through L, simultaneously C is discharged through R_L.

(b) Open: The combined current from source V_{in} and L is flowing towards R_L and charging C simultaneously.

Figure 13: Schematic of the two operating modes.

3.8 Regulation

In order for the Power Amplifier utilize the powerbank effectively and operate properly, the new power circuit containing the powerbank must be regulated. This regulation depends completely on how the powerbank is integrated in to the circuit. Both the power output from the powerbank and the power input (charging) of the powerbank must be regulated accordingly.

3.8.1 Regulating on a switching circuit

By utilising a switch, the circuit can select power supplies individually. In the case of this project the power supplies are the PoE and the powerbank. The switch chooses the source depending on an applied control signal $u(t)$. Because the switch only switches between two outputs $y(t)$ the control signal also only needs to toggle between two values. A regulator that works according to this principle is the ON/OFF-regulator [24]. Figure 14 shows the relationship between the control signal $u(t)$ and the output signal $y(t)$ for an

ON/OFF-regulator implemented with a hysteresis. The hysteresis helps to avoid oscillations around the switchover point and improve stability.

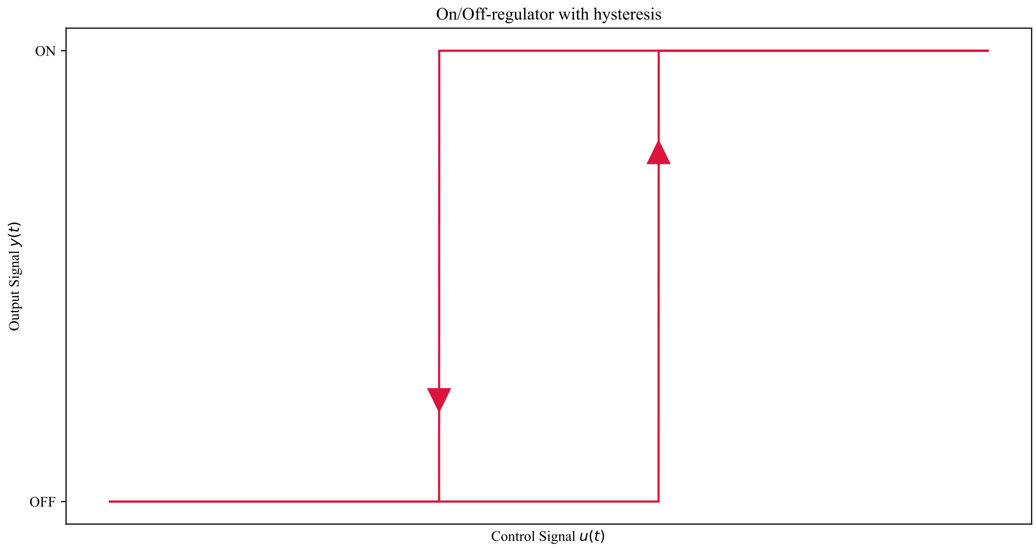


Figure 14: Operation of ON/OFF-regulator with hysteresis.

Due to the short peaks in audio signals a pure ON/OFF regulator might not work for this project, because it would require very fast switching. An alternative implementation of the ON/OFF-regulator might be required.

3.8.2 Regulating the charger

While utilising a switching circuit that switches between the PoE and powerbank, the charging process of the powerbank must be regulated properly. The power required by the charger must be supplied from the PoE. When the powerbank is actively driving the circuit, it might not be possible to charge it simultaneously. When the PoE alone is powering the circuit, It is crucial to ensure that the charger is not drawing power needed by the PA, but rather utilizing any surplus power that the PA is not using, that is: $P_{\text{charger}} \leq P_{\max} - P_{\text{PA}}$. The power currently being utilized by the PA (P_{PA}) can be estimated by measuring the current flowing into it.

Developing an appropriate regulator for the charger is a complex task that exceeds the scope of this project, given the time constraints. However, a potential alternative solution utilizing the charger will be presented, see Section 6.1.3.

3.9 Powerbank Types

To increase the power output by switching to a new source, the new source must be able to deliver more power than the PoE on its own. This places certain demands on the type of powerbank and the technology. In addition to this, the powerbank may be required to charge/discharge at a certain rate depending on how often the power source is switched without significantly reduce its lifetime. Also, every battery has a rating which determines how fast the battery can be charged or discharged, putting further constraints on choice of chemistry.

3.9.1 Rechargeable Battery Technologies

Ni-MH:
The Nickel Metal Hydride battery is a battery technology with a chemical reaction similar to the older Nickel Cadmium technology, but with higher capacity and without memory effect [25], [26]. The memory effect in rechargeable batteries occurs when regularly recharging a battery after partial discharge leads to a gradual reduction in its maximum energy capacity. However, the life expectancy is rather low in harsh conditions and they generate more heat than Ni-Cd during charging which increases the charging time. The nominal voltage of Ni-MH cells are 1.2 V.

Li-Ion:
There are various Lithium Ion Battery technologies based on different chemistries [25]. Typically, Li-Ion batteries exhibit a very high charge density, surpassing that of Ni-MH batteries. They do not experience memory effect and do not need to be fully discharged before being charged again. It can undergo numerous charge and discharge cycles before its lifespan is diminished. The nominal voltage of Li-Ion batteries is notably high, ranging from 3.2 to 3.8 V depending on the specific chemistry. Among these, Lithium Iron Phosphate (LiFePO4) batteries boast the highest current rating. Improper handling of Li-ion batteries can lead to complications, for example, discharging a cell below approximately 2.4 volts can trigger irreversible oxidation reactions that degrade capacity. Similarly, forcing current into a fully charged cell can also diminish its capacity. Li-ion batteries are also slightly more expensive and has higher internal resistance compared to Ni-based batteries.

Supercapacitors:
For fast charge and discharge cycles, supercapacitors are a more efficient alternative to batteries [27]. Supercapacitors have larger capacitance and energy storage capacity than normal capacitors, but lower than batteries and can deliver charge much faster than batteries. A downside with supercapacitors is their generally lower nominal voltage and still significantly lower capacity compared to batteries, leading to rapid voltage fluctuations during charge and discharge. This characteristic makes them unreliable and difficult to employ in situations where stable power delivery to a system for an unspecified duration is needed.

3.10 Charger

To use rechargeable batteries without manually removing them and charging them, a charger must be implemented into the product. How the charger works, and which technique it utilizes, depends on the battery type and how it will be used.

3.10.1 Constant Current - Constant Voltage

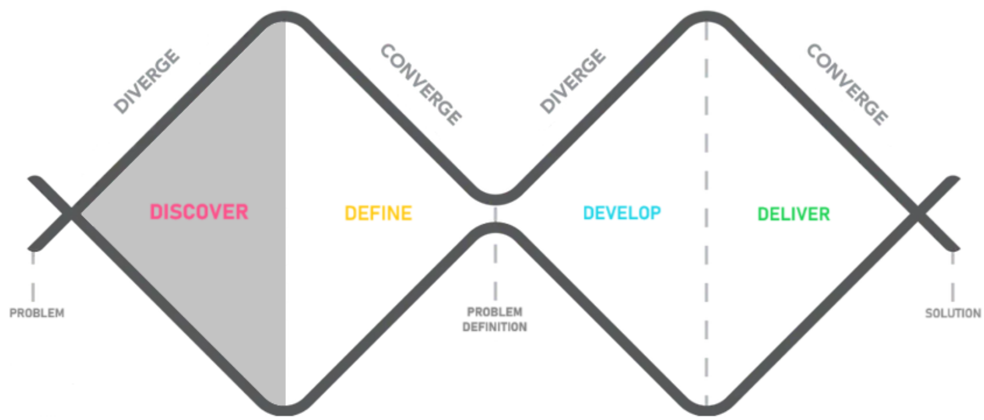
The Constant Current - Constant Voltage (CCCV) technique is a very useful technique for charging Li-ion batteries. The charging process begins with a trickle-charging stage with a low current ($\sim$ 0.1C). If the battery is very drained this initial step is particularly important. The battery is then supplied with a constant current (CC) where the voltage of the battery increases. When a voltage specified by the battery manufacturer is reached the charging is switched to constant voltage (CV). Here the battery is supplied with a constant voltage while the charging current into the battery is gradually decreasing. The charging stops when the current falls below a predefined limit. [28]. The CCCV technique contributes to a longer battery life and it helps to protect the battery from overcurrent and overvoltage. As a comparison, Ni-MH is normally not charged with CCCV and requires a more complicated charging technique [29].

3.11 Battery Management System

A Battery Management System (BMS) is a smart system that monitors many different parameters of a battery like State-Of-Charge (SOC), State-of-Health and more. The final product will probably require an implementation of a BMS but these can be very complicated and due to the limited time of this thesis project we won't delve into them extensively.

4 Discovery of project direction

In the "Discover" sub-phase, the subject is examined from a broad perspective, allowing for the identification of related problems.



4.1 Identifying problems

During the first phase of the thesis work the aim was to identify what different types of power related problems are present in a PoE driven class-D amplifier speaker system. This was done by researching PoE systems, CDAs and loudspeakers, much of which is presented in the theory section. Many simulations and measurements were also performed in order to more thoroughly understand where issues can appear. Four different types of power related problems have during the "Discover" phase been identified in CDAs and will be described below. These are:

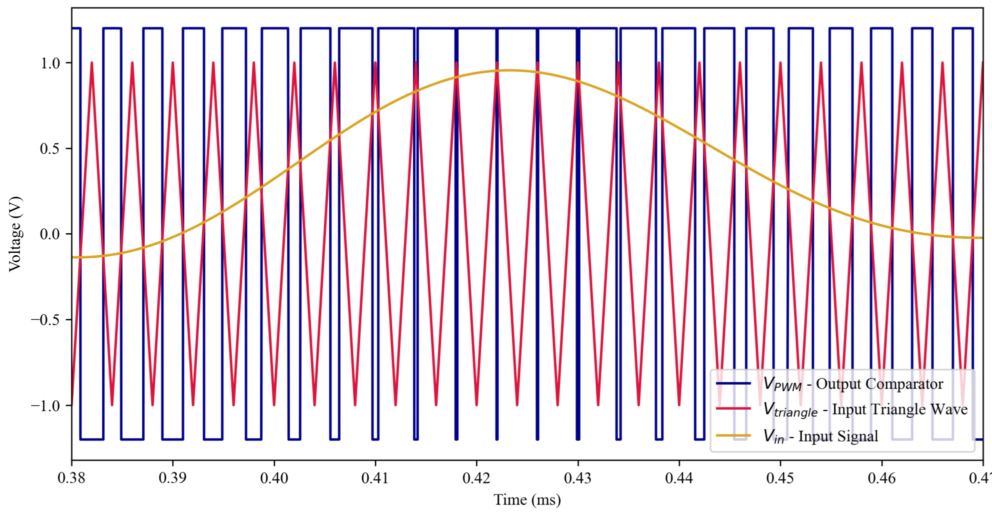
- Clipping of output signal due to mismatched PWM inputs.
- Compression of output signal due to high crest factor on input signal.
- Clipping of output signal due to power supply ripple.
- Clipping of output signal due to low impedance on load.

4.2 Clipping of output signal due to mismatched PWM inputs

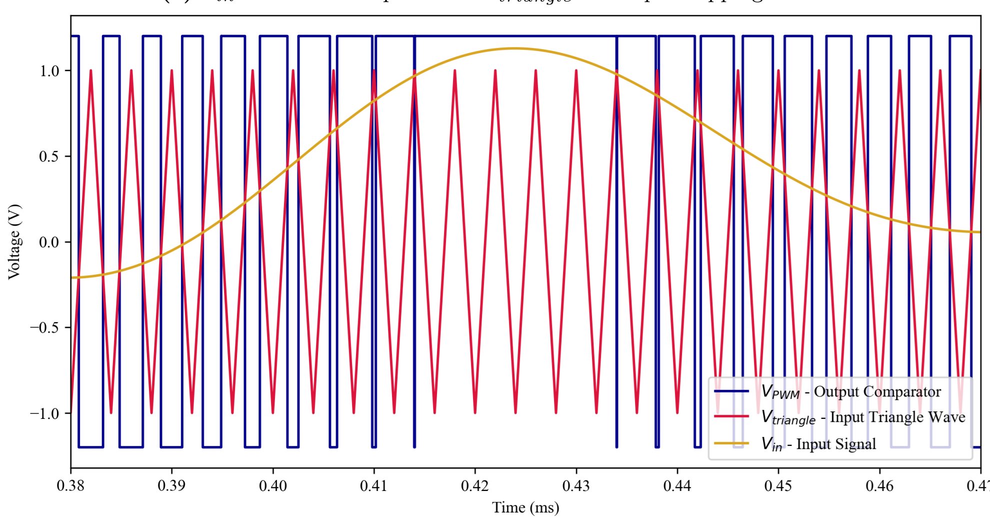
As described in Section 3.4, the CDA functions by amplifying a PWM signal generated by a comparator. Within the width of the pulses lies information on the amplitude of the input signal. When the amplitude of the input signal is higher than the triangle wave the comparator outputs high, and when the amplitude is lower, it outputs low, creating pulses where the width reveals how long the input has been larger than the triangle wave, see Figure 15a.

An issue that theoretically can occur is that the maximum amplitude of the input signal is larger than the maximum amplitude of the triangle wave. This will cause the comparator to overlook important information regarding the amplitude change of the input signal and effectively cut the output peak of the amplifier, see Figure 15b.

However, while this is a very real issue that can occur, it is also very simple to avoid. Essentially the only thing that is required to avoid any faults originating in the PWM is to set a hard limit on the amplitude of the output signal of the DAC (input to PWM) to ensure it never exceeds the maximum amplitude of the triangle wave.



(a) V_{in} within the amplitude of $V_{triangle}$. No input clipping occurs.



(b) V_{in} has a greater peak amplitude than $V_{triangle}$. Input clipping occurs! Notice that the pulse maintains its high value for too long.

Figure 15: If the input signal, V_{in}, has greater peak amplitude than the maximum value of the triangle wave, $V_{triangle}$, the output PWM signal, V_{PWM}, from the comparator will be saturated at the peaks and information from the signal will be lost. The signal will be clipped after exiting the power stage.

4.3 Compression of output signal due to high crest factor on input signal

A related issue to previous Section 4.2 has to do with how the input signal is limited before it reaches the amplifier. As explained in Section 3.3.1, many audio DSPs compress the audio signal to some extent, usually depending on the crest factor. In most cases this is beneficial, allowing a higher output volume without interfering with the natural dynamics of the sound. But some signals are inherently very dynamic and posses a high crest factor, such as speech. In these cases, if the compression is too aggressive, the speech will sound flat, and it is more beneficial to allow the high peaks remain, which either results in a lower overall volume, or demands a higher output power. In Figure 16 a compressed audio signal (yellow) is compared to its non-compressed version (blue). The problem arises when a signal is inherently very dynamic. Then the high peaks limits the amplification of the low peaks, or the high peaks will be clipped. Because of this, the signal must either be compressed before amplification, or the high peaks needs an increased maximum power ceiling.

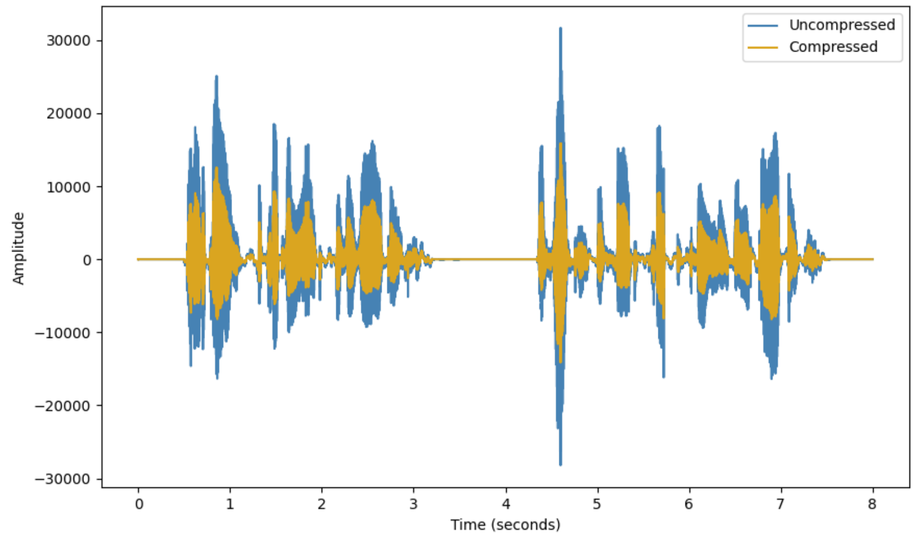


Figure 16: Compression of an audio signal.

4.4 Clipping of output signal due to power supply ripple

Since a CDA operates between supply voltage and ground, the amplifier output will be susceptible to deviations in power supply voltage [30]. Any fluctuations in supply voltage will propagate to the output through the pulse train and appear as ripple at the amplified pulse train peaks. Figure 17 shows a simulation where a simple (but exaggerated) sine wave is added to the supply voltage as ripple. The pulses will peak at the ripple and not the preferred supply voltage. This sinusoidal appearance will then appear at the audio output as well, attenuating and amplifying the signal corresponding to the ripple. However, ripple does not necessarily appear as a sine wave.

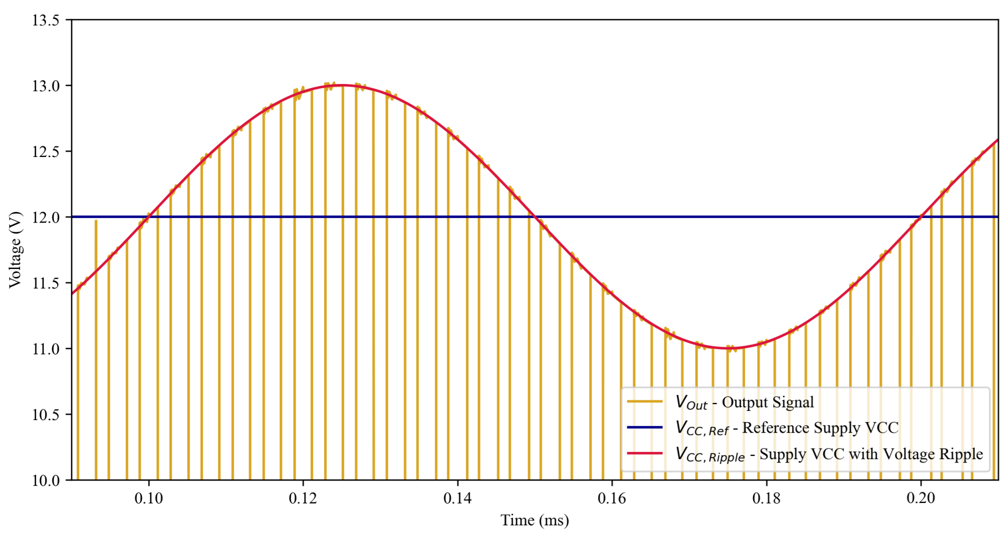


Figure 17: Voltage ripple on the supply rail (Exaggerated).

4.5 Clipping of output signal due to low impedance on load

An ideal voltage source will always deliver the required amount of current to keep its voltage constant, this current can thus be infinitely high. In reality, no such sources exist, and the current is always limited to the maximum power capacity of the source. The effect of this becomes apparent when the voltage source is connected to a very low load. Due to the limited power capacity, the voltage source will not be able to deliver the amount of current required and as a result the voltage provided by the source will drop. This becomes apparent from ohm's law in Equation (16) below.

$$P_{max} = UI_{max} \quad \Rightarrow \quad U = I_{max}Z\downarrow \quad \Rightarrow \quad U\downarrow \tag{16}$$

As seen, if the impedance drops while the current is limited and can not increase, the voltage will drop instead.

In a loudspeaker the impedance is not constant and will vary with the amplitude and frequency of the signal, as was discussed in Section 3.5 and can be seen in Figure 9. If the total impedance becomes too low, the supply voltage feeding the amplifier might not be able to deliver the required amount of current. Because of this the rail voltage will drop and the signal will be clipped.

Figure 18 shows a simulation of how the output voltage is distorted (soft clipped) when the current is limited for a 1 kHz sinusoidal signal compared to a signal with an ideal source.

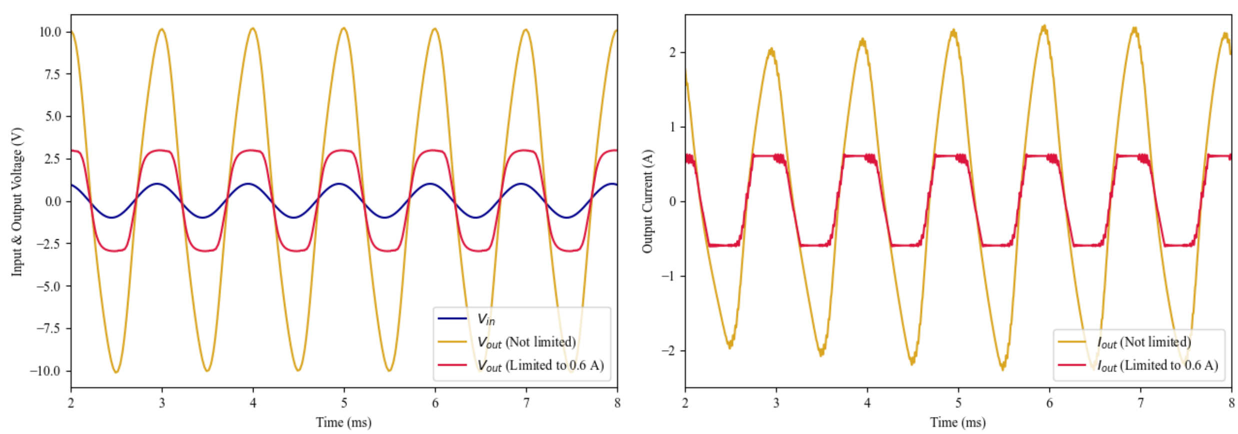
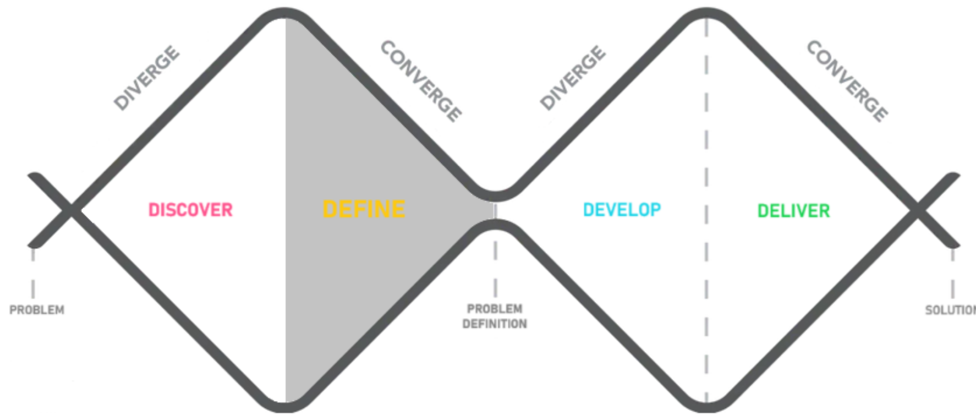


Figure 18: Effect of limited current on the output voltage and output current (right) compared to the unlimited output.

5 Definition of project focus, goals and restrictions

In the "Define" sub-phase the main problem of the project is defined and the focus, restrictions and goals are established.



5.1 Project focus

In the previous section most of the power related issues that can arise in a CDA were explored. While all of them are relevant, a deliberate decision was made to focus exclusively on mitigating the issue related to the "clipping of output signals resultant from low impedance on the load" within the scope of this project. This choice was informed by the significant impact that resolving this issue can have on speaker system's performance. Additionally, time constraints necessitate focusing efforts on addressing only one concern rather than attempting to tackle all identified issues simultaneously.

The "clipping of output signal due to mismatched PWM inputs" issue is assumed to be solved already within the DSP of a complete audio system by limiting the output signal amplitude from the DSP. A similar argument can be made for the "power supply ripple" issue; The reduction of power supply ripple is an important topic for the quality of audio products. The whole project could be spent investigating and lowering power supply ripple, however it is assumed that a complete audio system has the necessary hardware and software in place to reduce the power supply ripple enough, hence this will not be the focus of the thesis. The "compression of output signal due to high crest factor" is a relevant topic for the project, but more of a secondary benefit than a primary focus. If the available power at the amplifier can be increased, then it is possible to improve on the software which compresses the audio to allow for a more dynamic reproduction.

5.2 Goals

The primary goal of the project will, as mentioned, be to increase the available power at the amplifier whenever the load temporarily requires a higher power. In its simplest form this could be done by simply running the amplifier from a powerbank source with higher output power than available from the PoE, and then charge the powerbank from the PoE whenever the speaker is not in use, as illustrated in Figure 19a and done in [31].

However, a more challenging approach would be to regulate the secondary source to only be available at the time instances when a higher output power is required, i.e. only increase the power at the peaks of the output signal, as seen in Figure 19b. There are several ways this could potentially be done, many of which are explored in the following section.

In order to evaluate the developed system, some figure of merits have to be measured, and have not necessarily improved, but at least not worsened. A few of the desired goals are:

- Higher possible SPL while driven by secondary source.
- Unaffected THD while driven by secondary source.
- "Seamless" transition between power supplies.
- Functioning algorithm for switching.
- No audible clicks, pops or other artifacts.

Other important merits certainly exists but will not be considered as important as the above. These are:

- Functioning algorithm for charging.
- Lifetime
- Cost
- Size

5.3 Restrictions

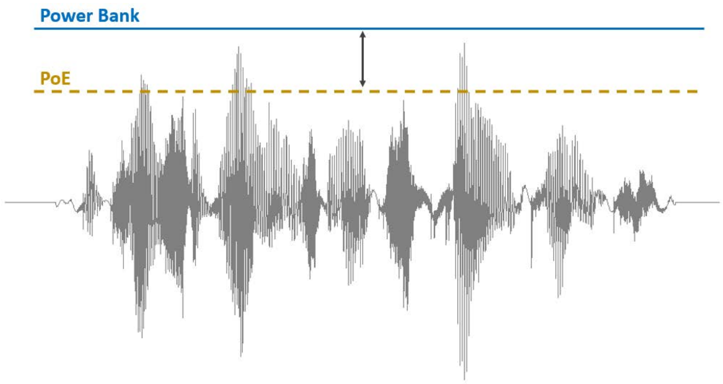
The scope of the work has to be narrowed with regards to the limited time frame. Therefore, some restrictions are imposed, many of which have already been mentioned previously in the report. The restrictions are the following:

- The work will only consider Class-D amplifier systems.
- The work will only consider PoE Type 1 Class 3 systems.
- The available power to the audio amplifier from the PoE will be limited to 7W.
- All measurements will be made on a Texas Instruments TPA3129D2 amplifier.
- All final measurements will be done using Audio Precision audio analyzer system.
- No custom made components/circuits.

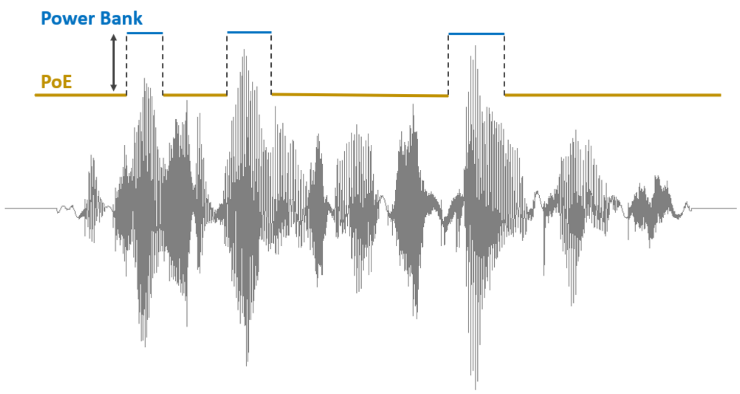
As previously mentioned loudspeakers powered by PoE have limited power capacity depending on the specific PoE-class. As can be seen in Table 1, the lowest class (Type 1) can deliver a maximum power of about 12.95 W to the PD. This work will be directly targeting applications that utilizes Type 1 PoE in order to narrow the scope, but can in theory be applied to higher PoE classes as well.

In addition to this limited power capacity, the speaker and its sub components can not take advantage of the full power range, since in most PoE systems, there are other electronic parts that require power as well. For this work it is assumed that roughly half of the power is partitioned for other components of the PD, leaving 7 W available for the amplifier to use. This is an important limitation to set in order to evaluate how much more power can be delivered in the end.

All components used in the prototype will be bought from a supplier, already mounted on evaluation modules (EVM) to save time. These EVMs are meant to only evaluate components and not to use for real testing or demanding use. They can therefore put some constraints on the prototype, mostly by affecting measurements since EVMs are not meant to be used in conjunction and are not matched nor optimized for the application. In a finished product it would be greatly beneficial to produce a custom Printed Circuit Board (PCB) for all components, which is not within the scope of this project.



(a) Use powerbank for entire audio signal.

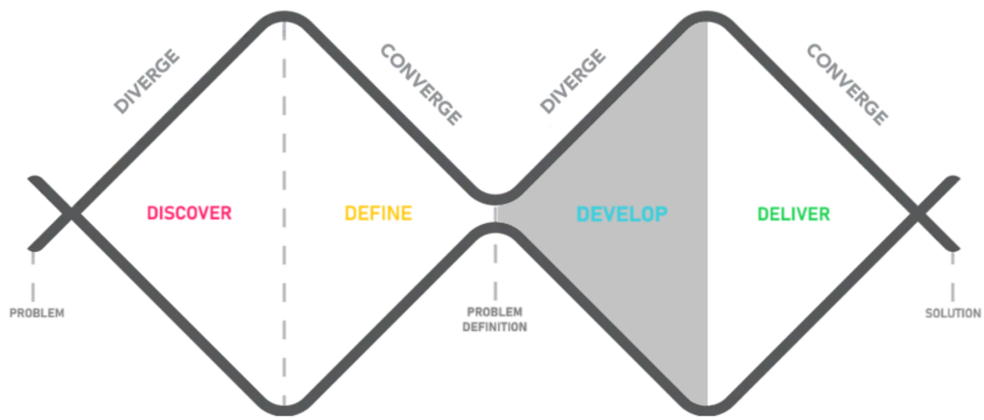


(b) Switch between sources when the audio requires higher output power.

Figure 19: Two ways to switch between sources.

6 Development of a potential solution

During the "Develop" phase, different methods and solutions are examined and tested. This section is mostly a chronological summary of the development process, ideas, trials and errors, measurements and theory.



6.1 Potential solutions

The very first course of action after localizing and defining the problem was to draft potential solutions or improvement. The basic concept of solving the power issue is to simply add a powerbank to the amplifier that can supply the amplifier with the extra required power, see Figure 20.

However, it is of course not as easy as just connecting a battery to the supply. Power combining requires advanced circuitry, and it can be done in a multitude of ways. Three principally different ways have been identified as potential solutions and are explored below. These three are **OR-ing**, **Switch** and **Hybrid**.

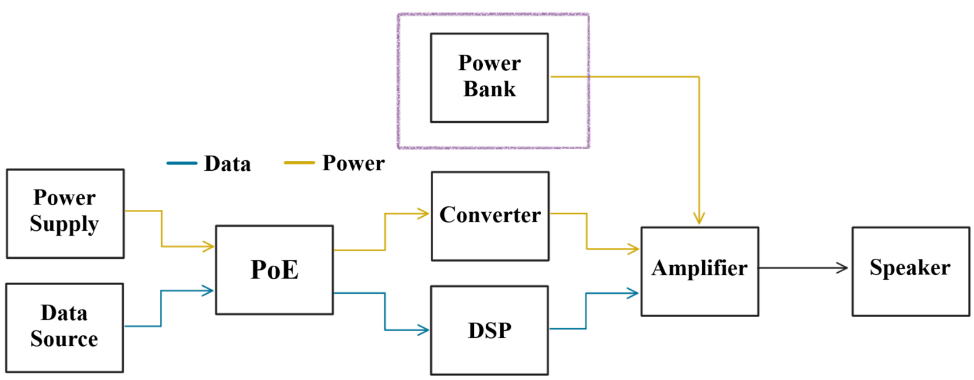


Figure 20: Powerbank solution block diagram.

6.1.1 OR-ing

OR-ing in electronics refers to the practice of combining multiple power supplies to enhance reliability and ensure uninterrupted operation of circuits, especially in applications where reliability is critical. This technique involves connecting power supplies in parallel to share the load and provide redundancy, see Figure 21, meaning if one power supply fails, others can continue to provide power [32], [33].

One common method to achieve OR-ing is by using power ORing diodes, typically Schottky diodes, placed on the output side of each power supply, see Figure 22. These diodes serve as a barrier between the power supplies, preventing a failed or shorted supply from disrupting the operation of others. If a power supply fails, its voltage drops to zero, and without protection, it could potentially short-circuit other connected supplies. The power OR-ing diodes prevent this by isolating each power supply from the others.

The advantage of using Schottky diodes lies in their fast reverse recovery time and lower forward voltage drop compared to conventional diodes, which ensures efficient operation. Unfortunately, diodes present an inherent issue with a voltage drop of approximately 0.5 V, leading to an increase in dissipated power. An alternative approach involves employing MOSFETs to create so-called ideal diodes [34].

Load sharing is another crucial aspect facilitated by OR-ing in electronics. When multiple power supplies are connected in parallel, the load is distributed among them based on factors like the forward voltage drop of the power OR-ing diodes and the voltage difference between the supplies. Essentially, the majority of the current is drawn from the supply with higher voltage and lower forward diode voltage drop [35].

In scenarios where increased current capacity is required, redundant power supplies with active load sharing capabilities are preferred. Active load sharing involves communication between the power supplies to adjust their output voltages, ensuring that each supply delivers the same current. This way, the load is evenly distributed among the supplies, maximizing efficiency and reliability.

Applying this to a speaker system fed by PoE and a powerbank in parallel seems plausible and effective, but it does come with some challenges. In order for a passive load sharing OR-ing setup to work the PoE would need to supply a higher voltage than the powerbank which could restrict the power output of the powerbank. Charging the powerbank also presents complications, since the two power sources are isolated from each other by the diodes the PoE can not deliver current to the powerbank and complex circuitry and communication between sources would be needed to facilitate charging.

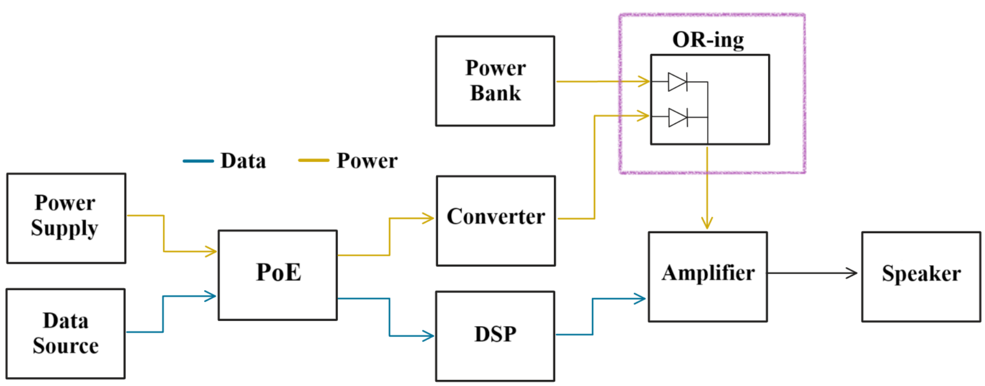


Figure 21: Block diagram of a power combining solution in an OR-ing configuration.

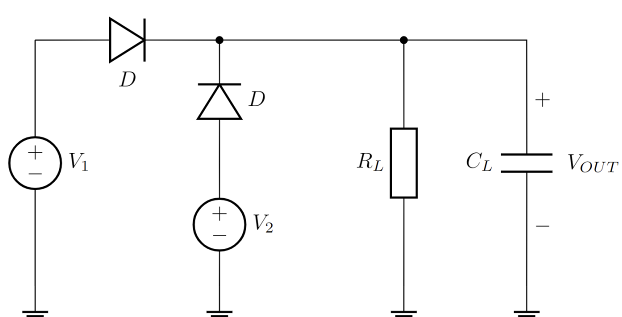


Figure 22: Simple Diode OR-ing

6.1.2 Switch

One of the most basic ways to combine power, or rather select a power source is via a switch, see Figure 23. Mechanical switches, though reliable, are for this application too bulky and slow. Instead, electrical switches would have to be utilized. One type of electrical switch is a Power Multiplexer (MUX), which facilitate the switching or combining of power from multiple sources. Typically, a MUX features two or more inputs for connecting separate power supplies and one output.

When transitioning between two sources, there are two main methods to ensure no reverse current flows through the unused source. The first method, known as "Break-Before-Make," involves completely switching off the supply path from the first source before activating the path from the second source. This prevents any direct path between the two sources during the transition. However, it results in a temporary interruption of power delivery to the load, requiring measures to ensure continuous operation during this period [36].

The alternative approach is a "Make-Before-Break" transition, where the path from the second source is opened before closing the path to the first source. Reverse current into the first source is prevented by other methods, such as OR-ing. This method ensures continuous power supply to the load without any interruption, making seamless switching more straightforward to achieve [36].

A MUX is a possible contender to combine the two power sources. It also provides possibilities to charge the powerbank from the PoE if the switching is configured in a way that connects the powerbank to the PoE when it is switched off. However, two major concerns are present in this potential solution. The first one being the speed of the switchover event; if the MUX can switch to the powerbank fast enough to provide enough power or not. The second concern being if the output voltage drops too low at a switchover, which would in all probability be audible, since a loudspeaker provides the load. Another minor concern is the longevity of a MUX that is switched often and hard.

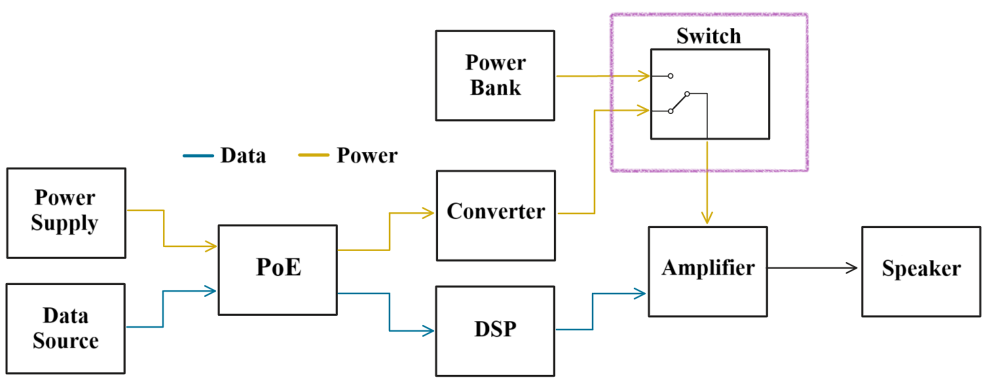


Figure 23: Block diagram of a power combining solution in a switch configuration.

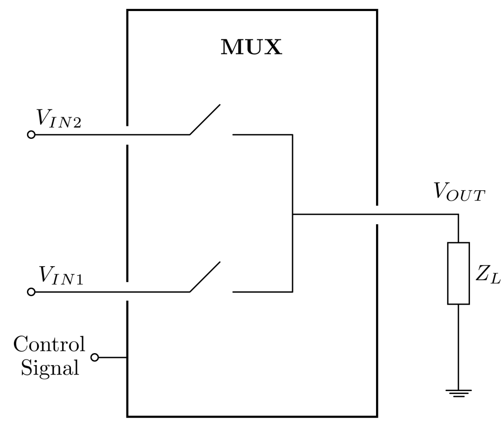


Figure 24: Simple schematic of a power multiplexer (MUX). The output V_{OUT} can be switched between V_{IN1} and V_{IN2} by a Control Signal.

6.1.3 Hybrid

The main idea of the hybrid configuration is to use the PoE and powerbank simultaneously through a powerbank charger, see Figure 25 and 26 below. When the PoE itself cannot deliver enough power to the system, the powerbank contribute with the extra required power that the PoE cannot deliver.

The hybrid solution presents a challenge when working with PoE systems. PoE systems themselves are not power-limited, meaning they will simply shut down if excessive power is drawn. In order to use PoE in a hybrid configuration, a current or power limiter must be included. Otherwise, the PoE will attempt to deliver the excess power instead of allowing the powerbank to handle it, which can lead to shutdowns if too much power is drawn for an extended period.

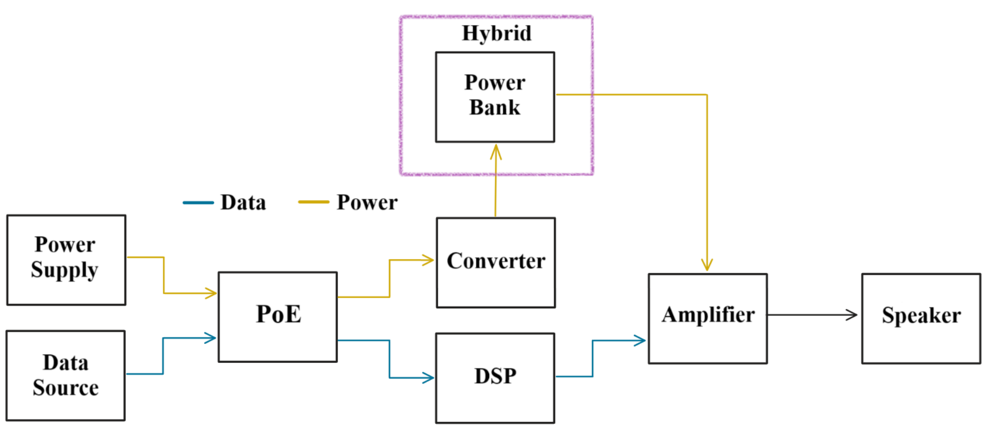


Figure 25: Block diagram of a power combining solution in a hybrid configuration.

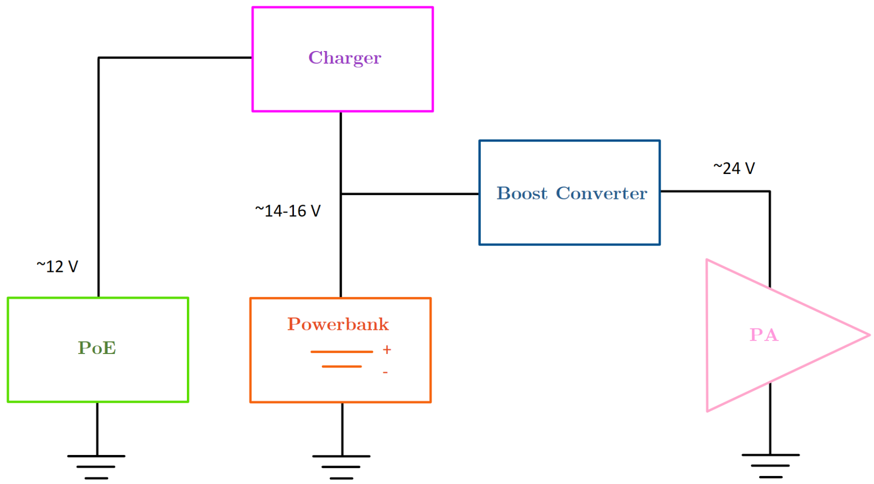


Figure 26: Simplified schematic of the hybrid solution. Power combining using only the charger.

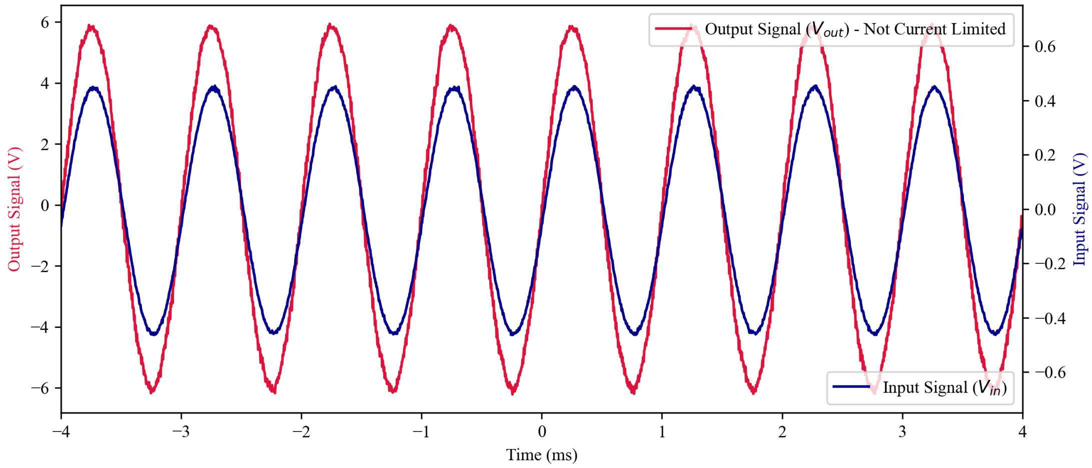
6.2 Switch/MUX prototype

The most promising idea after careful consideration and discussions with supervisors was to utilize a MUX, leading to a decision to prioritize this solution while keeping the option open to explore others if time permitted. This decision was made partly because the solution seemed very reliable and feasible, and partly because it seemed a lot more fun. The initial step involved researching available power multiplexers on the market that fulfilled specific criteria, such as input and output voltage range, switchover time, and voltage drop. Additionally, potential complementary components, such as boost converters and chargers, were investigated. Upon identifying potential candidates for all components, an order was placed. After receiving the components, several preliminary measurements were conducted on each component to assess their performance and determine if they were suitable for the task or imposed any limitations on the solution.

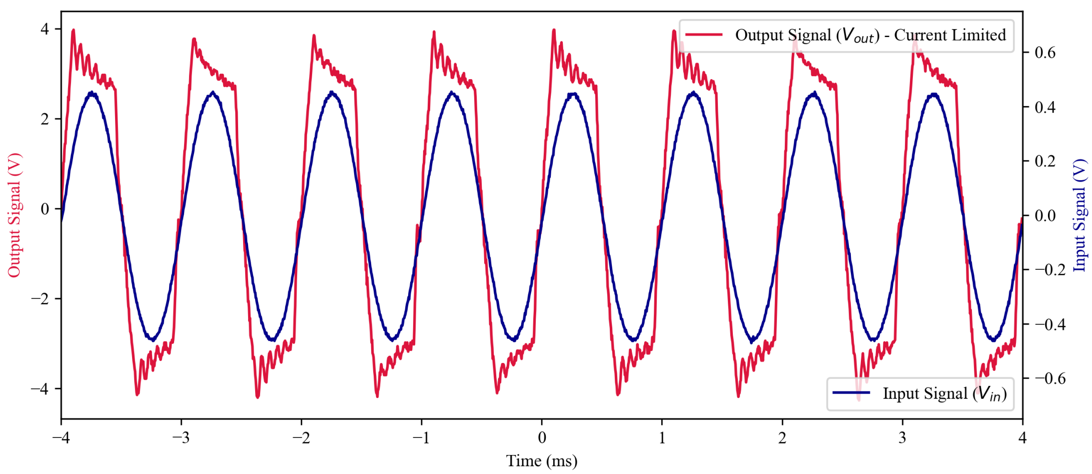
6.2.1 Amplifier

As previously mentioned, all measurements will be conducted using the TPA3129D2 amplifier. This is mainly due to already having access to a unit as well as its corresponding evaluation module. The amplifier was first measured with a static load in order to find the same clippings as was observable in the simulations. This was done by powering the amplifier with a power supply unit (PSU) with variable voltage and current settings. The amplifier was simultaneously fed with a sine wave and an oscilloscope was probing the input and output in order to observe the clippings, see Figure 27a. Just as in the simulations, the current from the PSU was limited while the amplitude of the sine wave was continuously increased until the amplifier began to cut the peaks of the output signal, see Figure 27b.

This clipping effect is the primary problem as presented in Section 4, and its existence is therefore vital to the solution, since without the problem there is no solution. The amplifier is a Class-D amplifier, rated for a maximum supply voltage at 26 V at the power stage and a maximum output power of 30 W to a 8 Ωmono load. Since the maximum output power is 30 W in mono, and the maximum power delivered to the PA is only 7 W, as determined in Section 5.3, the amplifier is in practice able to handle a much higher input power. This is where the powerbank enters the conversation. When the PoE can not deliver the current that the signal requires and the amplifier clips the signal, the amplifier is instead fed by a powerbank which can supply the required current to avoid clippings.



(a) Not current limited - Accurate amplification.



(b) Current limited - Clipping Occurs!

Figure 27: Measured Output V_{out} and Input signals V_{in} from the amplifier (TPA3129D) when the current from the supply is not limited and limited. The power is supplied through the MUX (TPS2121) to the amplifier and the input signal is a 1 kHz sinusoidal.

6.2.2 MUX

In order to switch between the two power supplies a Power Multiplexer (MUX) was used. The MUX used was Texas Instrument TPS2121. It has two inputs, both possible to operate between 2.8 V and 22 V. The MUX can be run in two modes, either with automatic switchover or manual switchover. The automatic mode senses and selects which input source has the highest voltage, while in the manual mode it can be controlled for source selection using an external signal, which makes it a suitable choice for rapid and precise regulation. Since the automatic mode is essentially OR-ing, the MUX will only be run in manual mode for all tests and measurements.

The TPS2121 also has a fast switchover function, which can be activated in both automatic and manual mode. Fast switchover allows for faster transitions between input sources under the right conditions. The measurement in Figure 28 illustrates the switchover time for the MUX between two sources and demonstrates the improvement fast switchover provides when a logic control signal of 1 kHz is applied to the MUX. As seen, it noticeably speeds up the transition from V_{IN1} to V_{IN2}. However, the rate of switching back from V_{IN2} to V_{IN1} remains largely unaffected. The switchover time ultimately depends on the output load, as such it is difficult to determine an exact switchover time. Texas Instruments reports a switchover time (with fast switchover enabled) of 5 µs during peak conditions [37], while the fastest measured switchover time using an oscilloscope and 8 Ω load was closer to 200 µs. This number can surely be reduced, but since only EVMs are used and the load varies with the speaker, it is redundant to try to optimize it. Additionally, as will be seen later, a switchover time in the hundreds of microseconds is fast enough.

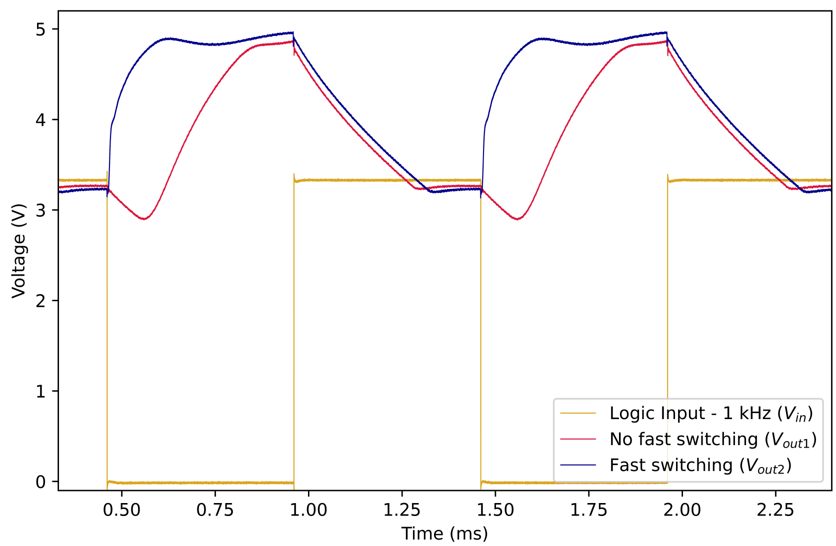


Figure 28: Plot of the MUX output voltages for fast switchover mode (blue) and normal mode (red) when the control signal is a 1 kHz square wave (yellow).

6.2.3 Boost Converter

The task of the MUX is only to switch between the PoE and the powerbank and will output whatever voltage and current the selected source provides. The PoE supplies a fixed voltage of around 12 V to the amplifier, while the powerbank has its own variable voltage. For instance, a powerbank composed of multiple Li-Ion battery cells in series, each at 3.6 V, can output only specific voltage levels like 3.6 V, 7.2 V, 10.8 V, and so forth. Moreover, the powerbank's voltage may fluctuate during use based on the State Of Charge (SOC) and during fast current fluctuations.

These voltage differences between the sources can potentially pose problems and lead to unpredictability when used for audio systems. When the MUX switches between the supplies and the output voltage rapidly drops or increases on the power rail to the PA (Power Amplifier), it can result in audible noises such as "pops" and "clicks."

To address this issue, a Boost Converter (BC) can be positioned after the MUX and before the PA (see Figure 42 in the previous section). The BC raises and stabilizes the voltage to, in this case, 24 V for a range of input voltages. As a result, the PA will always receive a stable 24 V supply regardless of which power source is active. By nearly doubling the output voltage, we increase the total available output voltage to the load. However, it is important to note that the BC reduces the maximal available current for use, which means that the maximal power limit P_{max} is more easily achieved. This is of no concern if we use a powerbank with a high enough power capacity.

The boost converter used is Texas Instruments TPS61377, working in forced PWM mode, and its associated EVM. It has an input voltage range 2.9 to 23 V, and an output voltage range of 4.5 to 25 V, which sits nearly within the operating range of the system. The most important part to evaluate in the BC is whether it drops in output voltage at a switchover event. If this drop is too severe it can cause "clicks" and "pops", exactly what the BC is supposed to negate.

Figure 29 depicts measurements from a switchover event. The blue curve represents the output voltage from the MUX as it switches between two sources ($V_{IN1} = 12$ V and $V_{IN2} = 10.8$ V). The red curve shows the output voltage from the BC. It is evident from the plot that the output from the BC remains relatively stable during the switchover event, with only a small drop of approximately 50-100 mV from 24 V observed, which is an acceptable drop. Tests demonstrated that the small voltage drop has no audible effect on the speaker signal. It can also be noted that the red curve is more magnified than the blue curve, giving the appearance of higher noise levels. While the output of the BC may indeed be slightly noisier, it is not significantly more noisy.

While the BC aids in increasing and stabilizing the output voltage, the fact that it consumes a small amount of power must be taken into consideration. The BC operates with an efficiency of up to 96%, implying that more than 4% of the power may be lost during its operation [38], and also has a quiescent current of 70 µA.

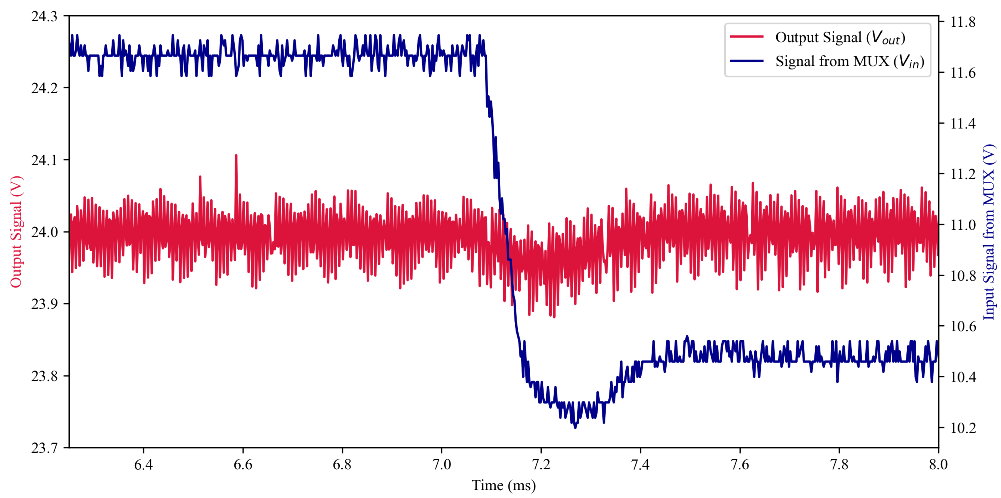


Figure 29: Voltage ripple at the boost converter's (BC) output during a switchover event. The blue curve shows the output of the MUX or input to the BC. The red curve shows the output voltage from the BC.

6.2.4 Switching algorithm

During development and testing of the prototype the switching of the MUX was controlled by an Arduino device through serial communication from Python code. The Python code itself contains an algorithm that sends a signal to the Arduino to toggle a switch depending on an analyzed audio signal. A Raspberry Pi was later set up to replace the Arduino, the only difference being that the Raspberry Pi runs the Python code directly making it more efficient and removes the step of serial communication.

The software algorithm is responsible for analyzing the incoming audio. The audio can both be a live stream from a microphone input or a wave-format audio file depending on how the program is preset. The code divides incoming audio data into chunks which consists of a predefined number of samples but is adjustable and can be increased in millisecond steps. The incoming audio is handled and analyzed one chunk at a time. If any sample in the current chunk (n) exceeds a predetermined threshold, that whole chunk will be marked as a boolean logic "true". This threshold has been an arbitrary value for testing but will require more attention in the future, see Section 6.2.5. The four latest chunks and their associated boolean states are saved in a list so that the previous chunk ($n-1$), the current chunk (n) and two chunks ahead ($n+1$, $n+2$) are available. It's important to note that $n+2$ is the latest analyzed chunk in the algorithm but will not be delivered to the amplifier in two iterations. The algorithm therefore has an inherent delay of two time-chunks in order to have enough time to be processed, something that in all probability will not be noticeable due to the very short size of the chunks. The boolean variables associated with all four chunks (V_{n-1}, V_n, V_{n+1}, V_{n+2}) are then analyzed together in the following logic function, Equation (17). This function determines if the algorithm should send a switch command to the MUX so that it switches to the powerbank as the power supply, or oppositely switch to the PoE.

$$\mathrm{SW}_n = V_{n-1} + V_n + V_{n+1} + \mathrm{SW}_{n-1} \cdot V_{n+2} \tag{17}$$

Here, SW_n is the current state of the powerbank, and SW_{n-1} is the previous, either switched on (true) or off (false). The result of this boolean function generates the next state of the powerbank, whether the MUX selects the powerbank or the PoE.

The key aspect of this logic function is that it guarantees that the chunk preceding every "true" chunk will also be toggled to true, and one chunk after every "true" chunk is also toggled true. Additionally there can never be less than two neighbouring chunks toggled "false". This algorithm was carefully developed in order to ensure that the powerbank is available for the required time as well as to avoid unnecessary switchover that can potentially cause stress in the hardware. In Figure 30 below it is visualized how the algorithm treats the incoming data. The vertical divisions are chunks, 5 ms long and the red dashed line is a threshold value, any input data that exceeds this value will toggle the chunk true, and a green colored chunk is toggled true. The length of each chunk is arbitrary in this case and should be set according to the application and type of signal.

A plot of how the algorithm processes and work with an audio signal can be seen in Figure 30 below.

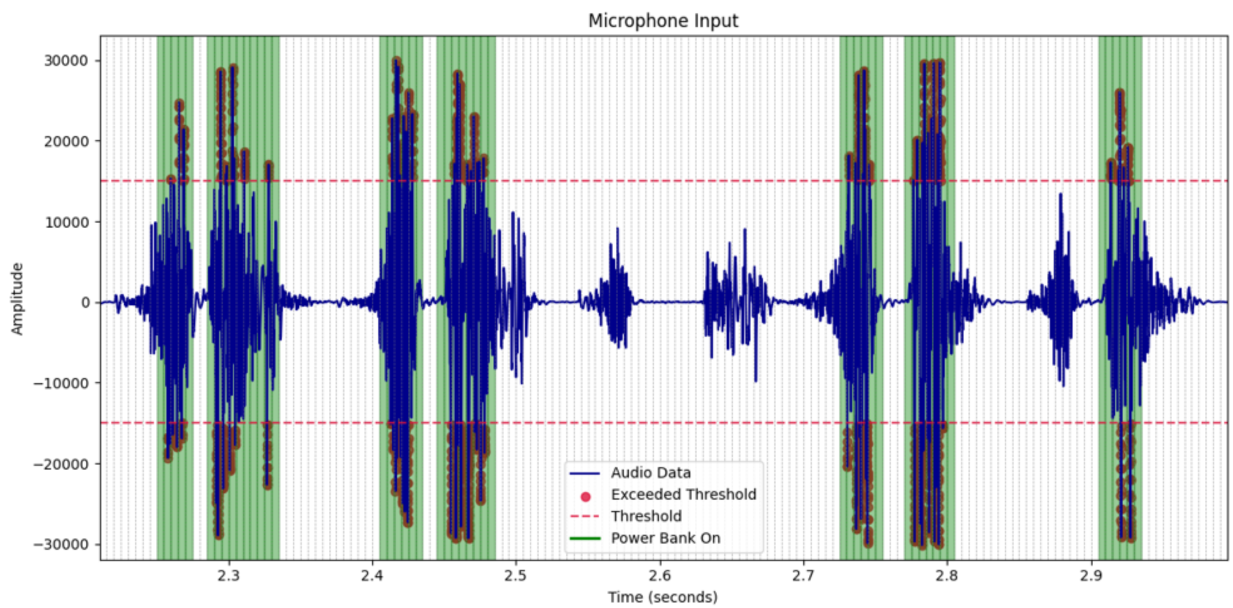


Figure 30: The result of an audio signal that has been analyzed by the algorithm. The small segments on the x-axis corresponds to the chunks. The dashed red line shows the threshold for switching. When the signal amplitude exceeds this threshold the system should switch over and use the powerbank. When this happens the chunk segment turns green.

6.2.5 Trigger signal

An important part of the algorithm above is to determine what triggers a switchover event, or rather what toggles a chunk "true". In the example above and during most of testing a simple threshold value was used. This was set to an arbitrary ±15000 of the input value. The audio file is analyzed as 16 bit data type, and thus has a maximum value of 32767 and minimum value of -32767. A high input value essentially translates to high required output power, and thus, for a real world application, it can be important to know what input value corresponds to what output power to be able to set the limit just below clipping. This is likely to differ among various speakers and thus needs to be calibrated individually for each one. However, in testing it is not as important.

The threshold method to determine whether a chunk is toggled true or not was at first used for it simpleness to implement and use. The idea was to develop the algorithm and triggering signal even further to make a more complex, dynamic and useful switching function. But after testing it has become more and more apparent that, despite the simplicity of the threshold method, it still reigns supreme.

Below, in Figure 31, a few other ways to trigger the chunks are plotted. These were attempts to extract more/other information from the incoming data signal to better determine if the signal warranted a switch to the powerbank. However, it became clear that much of the information was difficult or unnecessary to process and evaluate. For example, the red graph represents the derivative of the envelope, or rather how much the envelope changes between two points. From the graph it looks like a large positive derivative is a good indicator of an incoming sound with high amplitude, but this is not always the case and some peaks can easily be overlooked. Moreover, the RMS and envelope of the signal does not add any additional information or value. Hence, triggering on a threshold value for the data points seems to be the easiest and best way.

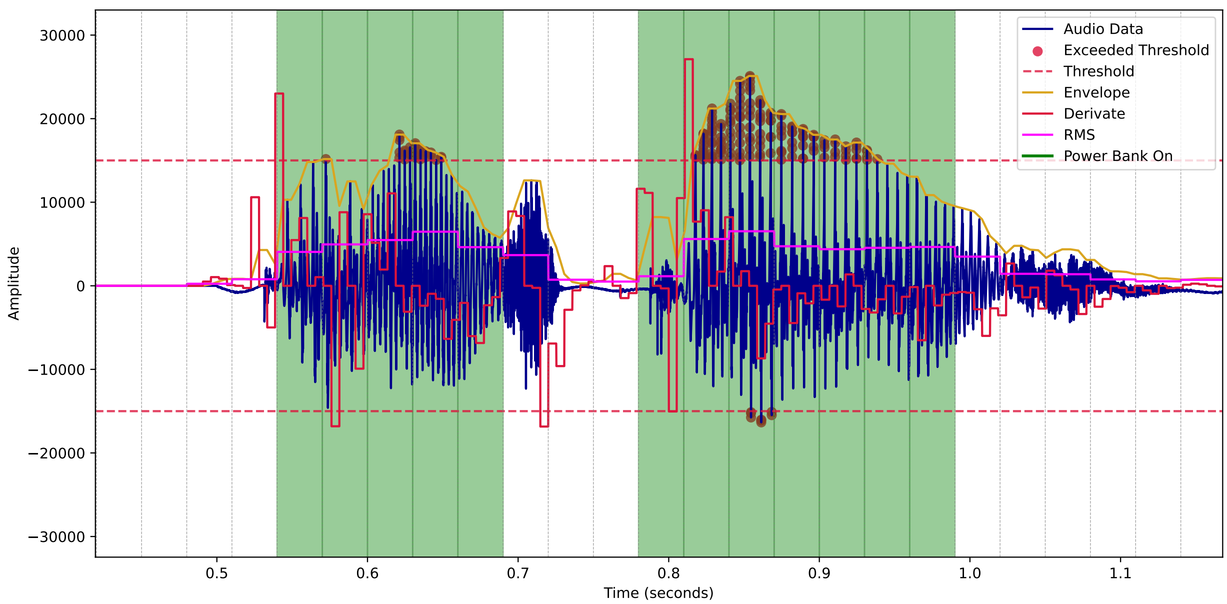
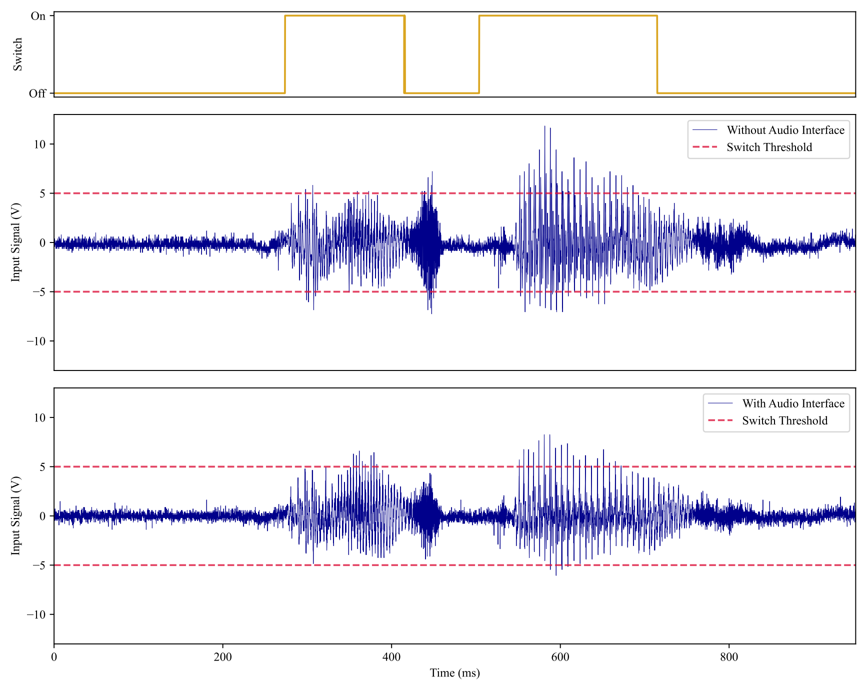


Figure 31: Different ways to handle the input and determine if the powerbank should be on or off.

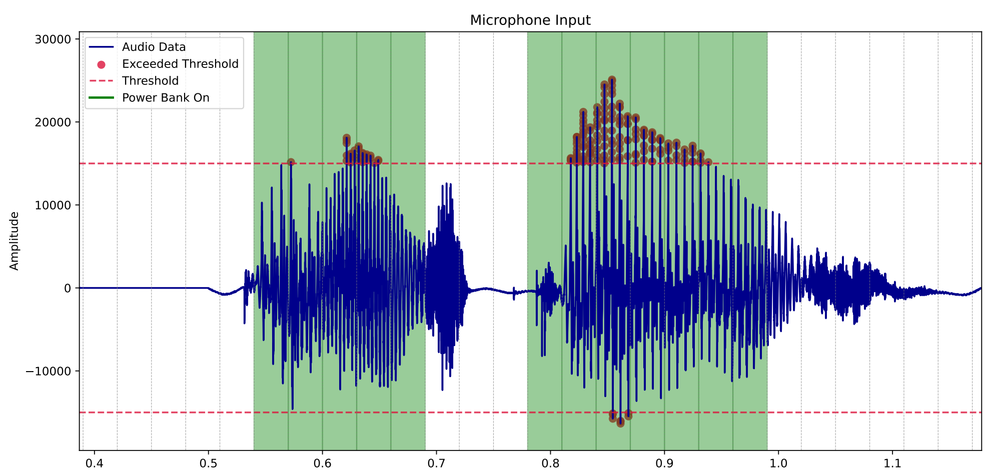
Most measurements were performed using an external audio interface to send the audio signal to the amplifier. Some measurements were also performed using the PCs own internal sound card. While using the internal sound card a significantly longer delay could be observed compared to when using the external audio interface. Furthermore, it was observed that the amplitude of the audio signal exhibited a slight increase in regions where the signals contained more high-frequency components, sometimes higher than the threshold for switching. However, due to the digital analysis of the signal before its conversion to analog audio, these amplitude increases were not detected by the algorithm and no switch occurred, even though it should. In Figure 32, the measurement without an external audio interface shows a peak above the threshold where the algorithm has not toggled to switch.

To address this issue, an attempt was made to improve the algorithm by performing a Fast Fourier Transform (FFT) on each of the chunks and thus analyzing the frequency spectrum at the higher frequencies. The FFT could then be integrated over high frequencies (from 5 kHz to 20 kHz) and compared to a threshold value.

If the integral value of the chunk exceeded this threshold, it indicated the presence of more high frequency components, prompting the algorithm to switch accordingly. This solution solved the issue but did add significant processing time. Although this can probably be much improved, the solution seems redundant in a complete system since the problem can also be solved by using a better DAC in the audio interface. Consequently, it was decided to proceed under the assumption that the speaker system incorporates a sufficiently high-quality DAC to mitigate this issue.



(a) The same analog audio signal played without (middle) and with (lowest) an audio interface (an external DAC). Due to the inferior DA-conversion the audio played without an audio interface has increased amplitude at places where the audio has high frequency content, thus it appears as though the algorithm has missed a peak, while the actual signal looks like the one in (b). The top plot shows where the MUX is toggled to switch.

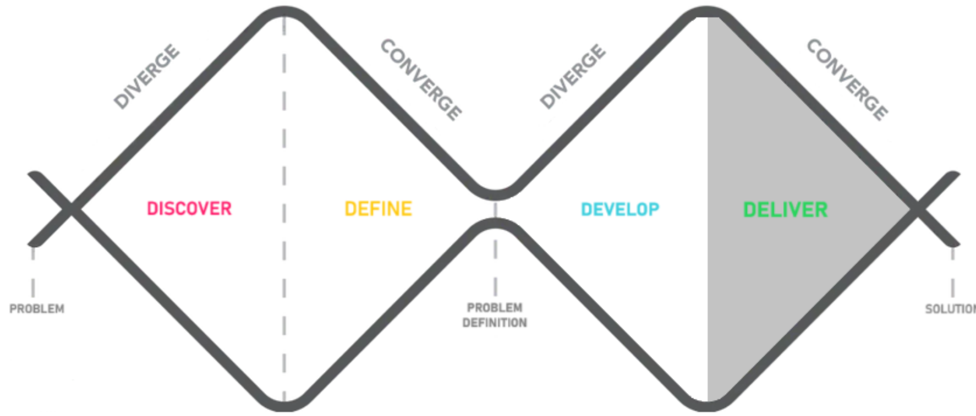


(b) The same digital signal before being DA-converted. It most closely resembles the bottom plot in (a) using an audio interface.

Figure 32

7 Delivery of a prototype

The "Deliver" sub-phase narrows down the solutions found in the previous phase and focuses on one of them to design a prototype.



7.1 Prototype design

Figure 33 presents an overview of all the system components in a complete schematic and illustrates a potential solution. From the PoE, the data and power is directly divided into two separate partial circuits.

In the power circuit, a flyback converter, converts the voltage from the PoE to the appropriate voltage before entering the MUX. The MUX is controlled by an algorithm to switch between the PoE and the powerbank as power a source. Subsequently, a boost converter steps up the voltage and stabilizes the output so that there are no noticeable difference in the potential when the sources are switched. The output of the BC then drives the PA to output an amplified signal to the loudspeaker.

The audio signal to be amplified is delivered from the data output of the PoE. The signal in the data circuit is routed through the DSP, which processes the audio, adjusts its volume and prepares it to enter the amplifier. Additionally, from the DSP the audio signal is extracted and analyzed by an algorithm. The algorithm is a program that analyzes small parts of the audio waveform called chunks and determines if the MUX should switchover to use the powerbank to drive the circuit instead of the limited PoE.

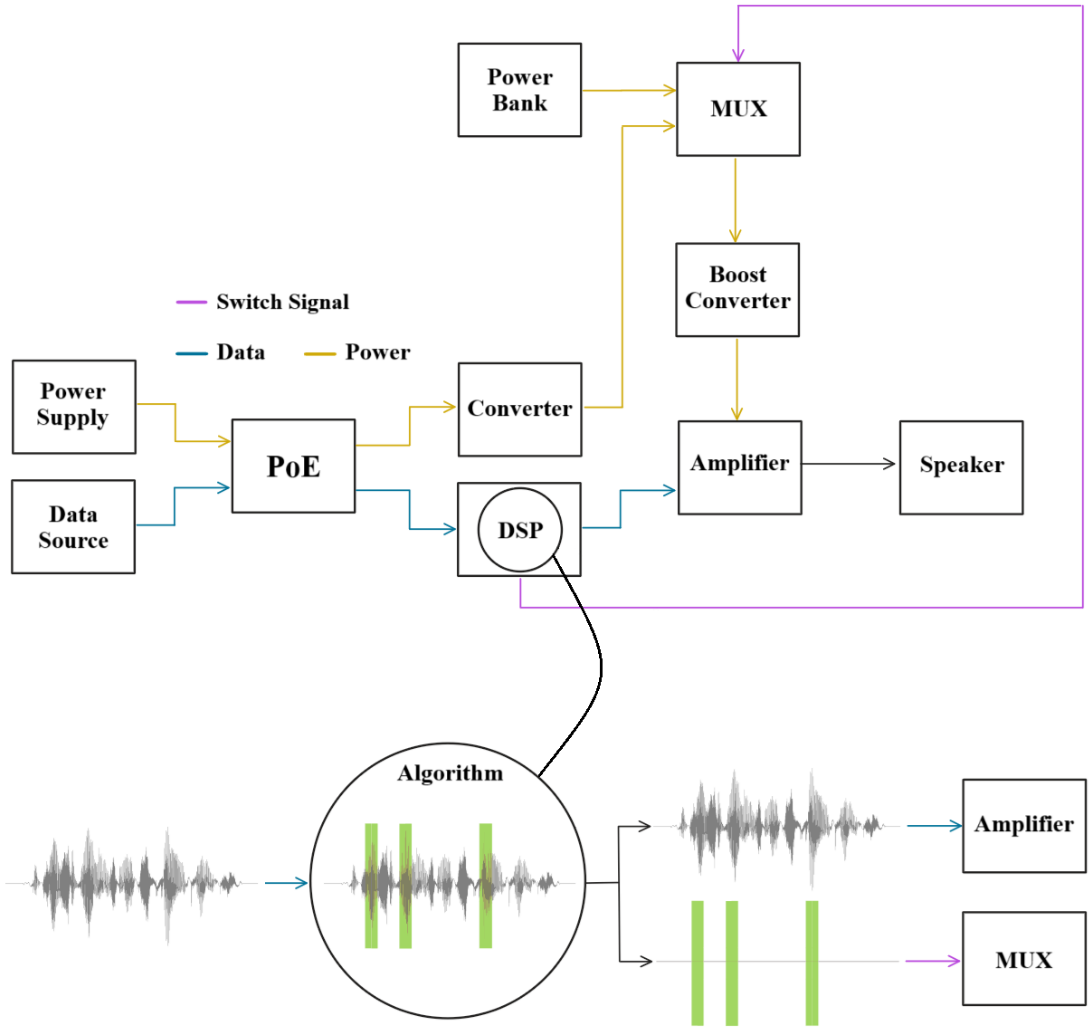


Figure 33: Entire block schematic.

7.2 Prototype measurement results

After each component was measured and evaluated individually, it was time to assemble and measure the complete system. Figure 34 shows the complete testbench setup for the measurements and how the components are connected together. The PoE is connected to IN1 and the powerbank is connected to IN2 at the MUX input. In addition an Arduino, or Raspberry Pi (both have been used), is connected to the MUX which runs the Python script and selects the input depending on the algorithm output. The output from the MUX then goes through the BC to the amplifier and out to the load (speaker). The amplifier is also fed with the same audio signal that the Python algorithm processes. However, when using the Arduino, the signal route for the data signal has more latency than the power route, resulting in a mismatch between the MUX switches and audio signal. Therefore, a delay was added in the code to mitigate this during testing. The latency issue was resolved later by utilizing the Raspberry Pi, which exhibited no signs of latency.

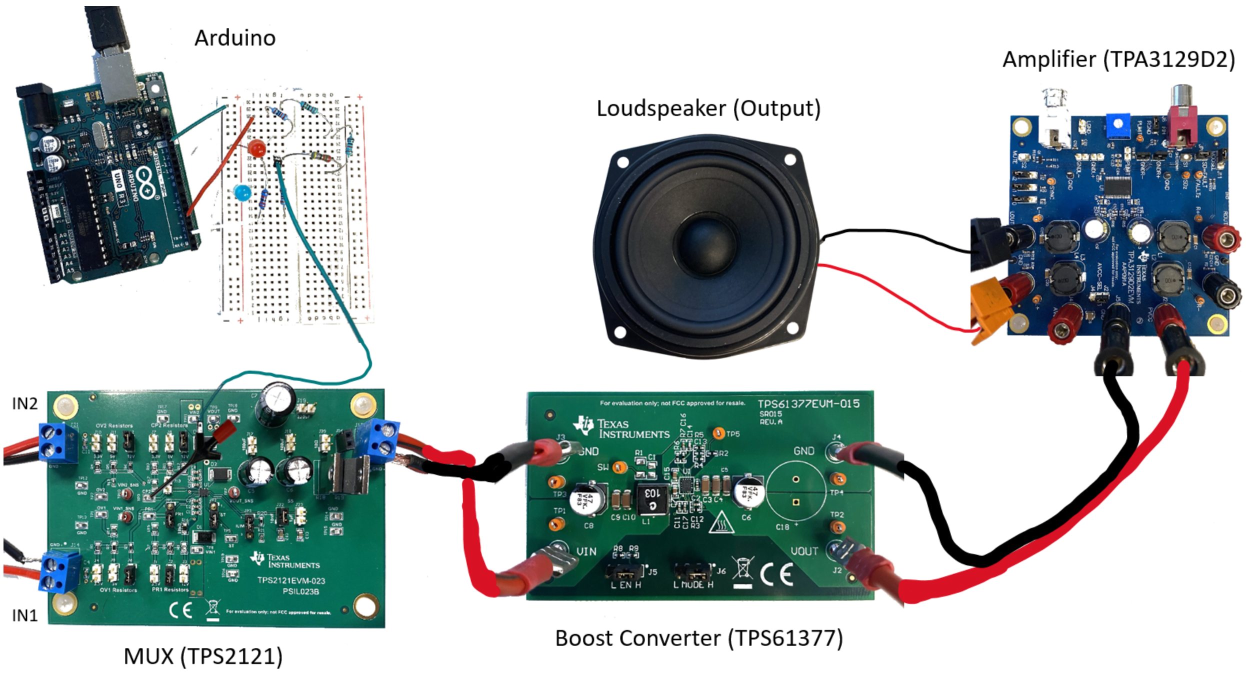


Figure 34: Testbench setup for the measurements. The MUX outputs to a BC which stabilizes and increases the voltage into the amplifier and the speaker. The MUX switching is controlled by and Arduino and Python code.

In order to evaluate the prototype several measurements were conducted. All of these measurements were performed in an anechoic chamber using Audio Precision (AP) hardware and software. They can be divided into two categories; electrical measurements and acoustic measurements. The difference between the two is that the electrical measurements demonstrate the behavior electrically "before" the speaker, and the acoustic measurements exhibits the response "after" the speaker, similarly to how a human would perceive the sound.

The first figure, Figure 35, are two electrical measurements of an audio voice signal, both current limited. One of them (red) is supplied only by the current limited power supply and exhibits heavy clipping. The other (blue), is supplied by both the current limited power supply and the powerbank with its switching algorithm. The upper plot displays both the output voltage from the BC, which sits stable at 24 V with only minimal noise, and the output voltage from the MUX (yellow), which visualizes the algorithm at work. When high, the powerbank is active and when low, the power supply is active.

The following two figures, 36 and 37, are two amplitude sweeps performed at 1 kHz. The first one is an electrical measurement across an 8 Ω load. It shows the output RMS voltage versus input RMS voltage. Figure 37 is the same measurement but with an 8 Ω speaker as the load and the sound pressure is measured instead of the voltage. The yellow curve has the powerbank active while the red is limited to 7 W.

Figure 38 and 39 are electrical measurements with a 4 Ω speaker as load. A 50 mV$_{rms}$ frequency sweep ranging from 20 Hz to 20 kHz was sent through different setup combinations of the prototype, and the output from the amplifier was then measured by the AP to obtain the behavior and response of the system. This was done in order to visualize any effect each component had on the system, if they introduced harmonics or noise in any way. The shape of the curve is not particularly important, it originates from the amplifier, instead, what matters is the difference between the measurements. Since each measurement is made with an additional component added the hope is to see no difference at all between the measurements. And as can be seen, there is only a negligible difference. The only component that adds some distortion is the BC, which slightly alters the appearance for high frequencies. Two measurements were also made while switching between the two input sources, one switching every 10 ms and one switching every 0.5 s. The same outcome can be seen here, while rapidly switching, there is no visible electrical difference between measurements.

Figure 40 is a similar measurement but acoustic, performed in the exact same way, except the input to the AP device comes from a microphone instead of the amplifier output. The left graphs shows the SPL frequency response of the speaker, and the right hand side graphs are spectrograms which are essentially a heat map of the frequency response over time. The same principle applies here, the shape of the curves are not of importance, they are directly related to the speaker and microphone. The interesting part is that there is no difference between Figure 40a and 40b, despite the measurement in Figure 40b is switched between PSU and powerbank every 10 ms.

Figure 41 is the exact same measurement as the one above, except it is done with higher input signal strength, 700 mV. This high amplitude will cause the amplifier to clip when driven by the current limited PSU which emulates PoE. As can be seen in Figure 41a, the signal clips and distorts heavily for almost every frequency in the sweep. Figure 41b, however, shows very little clipping and distortions, and this is because the measurement is done by driving the amplifier with the powerbank (batteries in this case). This is far from a real world situation, but it clearly demonstrates how a powerbank can help reduce clipping when the PoE can not supply the required power.

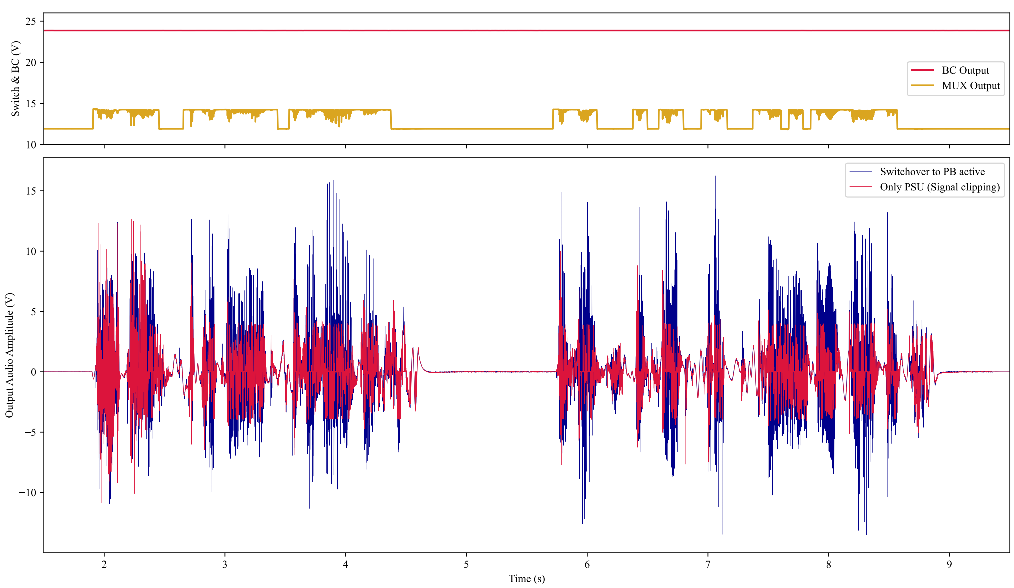


Figure 35: Two measurements of the same audio signal, both current limited. Red shows the waveform without the powerbank activated, heavy clipping. Blue shows the waveform with the powerbank activated, no clipping. The yellow curve in the top graph shows the output from the MUX, i.e. the switching signal. When high, the powerbank is active, when low, the power supply is active. The red flat curve in the top graph is the output from the BC, stable at 24 V.

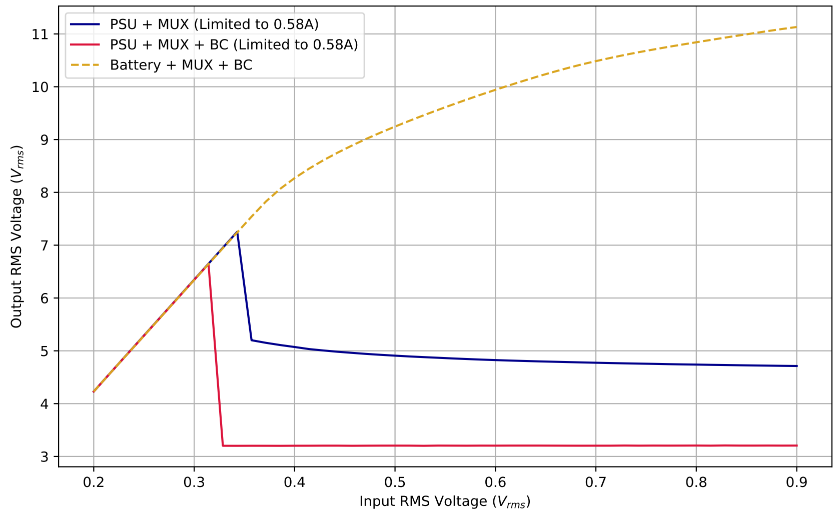


Figure 36: Output voltage across an 8 Ω load for different input signal amplitudes at 1 kHz. Yellow dashed line is with the powerbank solution implemented and red is without. The blue curve is without the boost converter and without the powerbank.

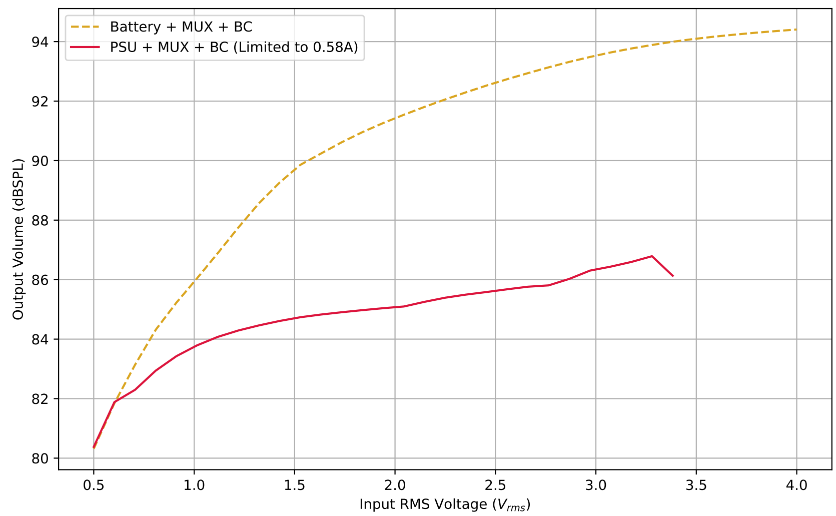


Figure 37: Output volume from an 8 Ω speaker for different input signal amplitudes at 1 kHz. Yellow dashed line is with the powerbank solution implemented and red is without.

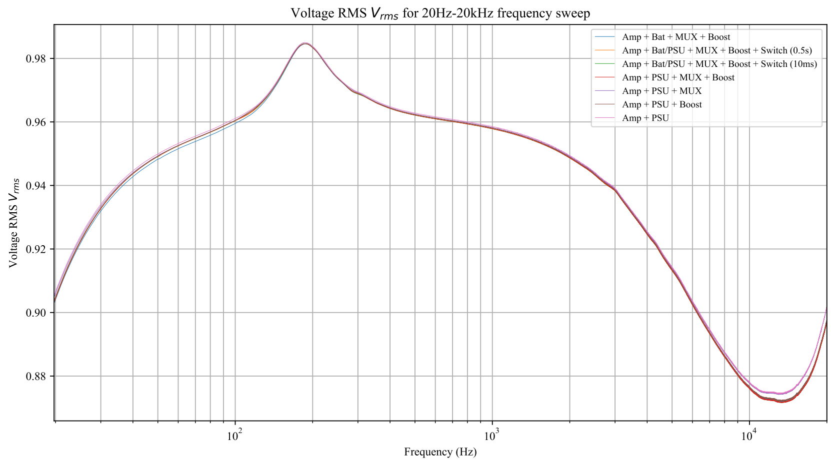


Figure 38: Electrical Voltage RMS versus a 50 mV$_{rms}$ sinusoidal frequency sweep from 20 Hz to 20 kHz. The measurement was made several times with added components to find if components affected the response. Two measurements were also made while switching between powerbank and PSU.

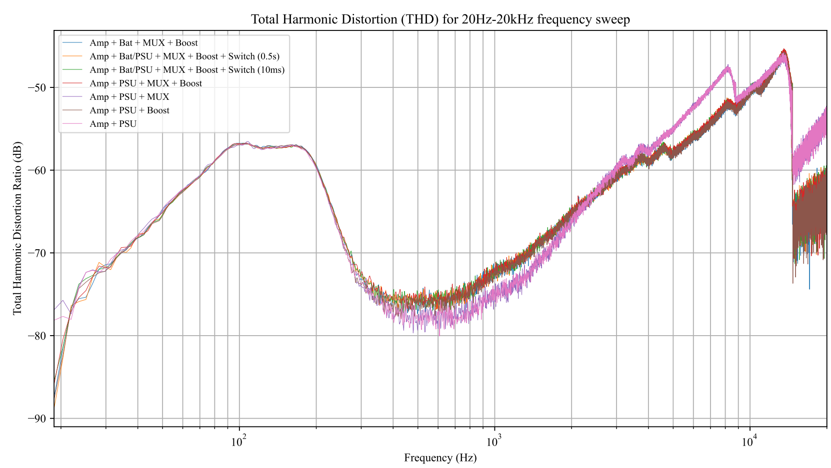
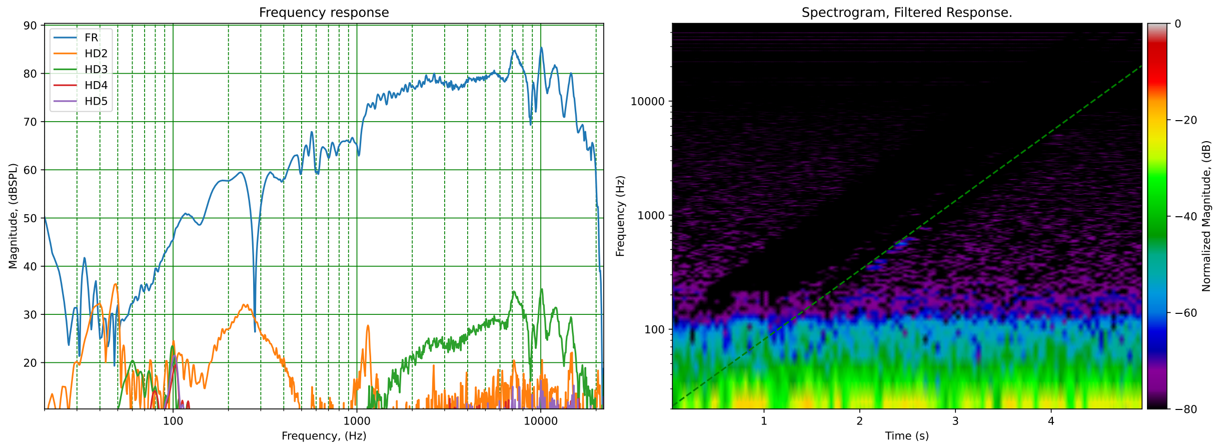
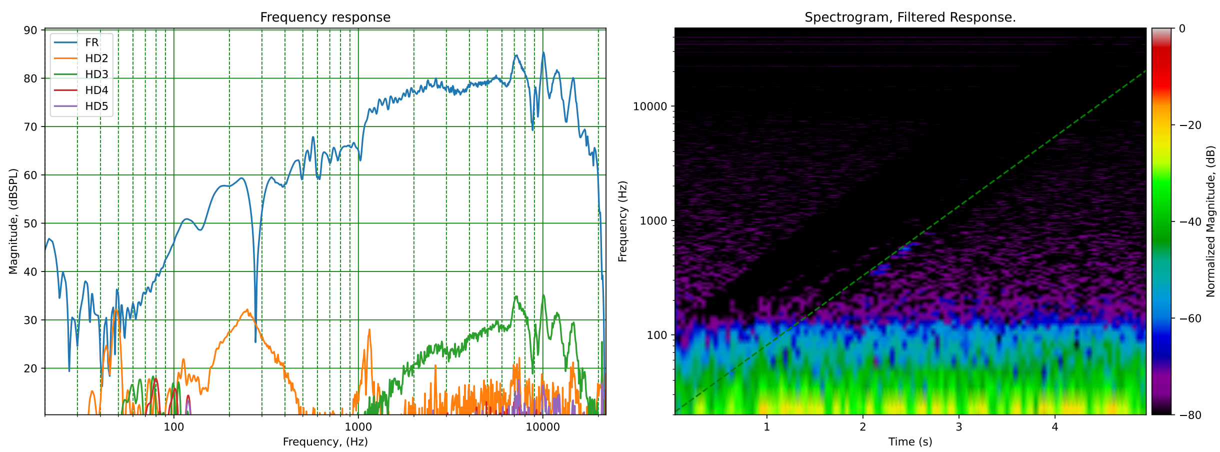


Figure 39: Electrical THD vs mV$_{rms}$ sinusoidal frequency sweep from 20 Hz to 20 kHz. The measurement was made several times with added components to find if components affected the response. Two measurements were also made while switching between powerbank and PSU.

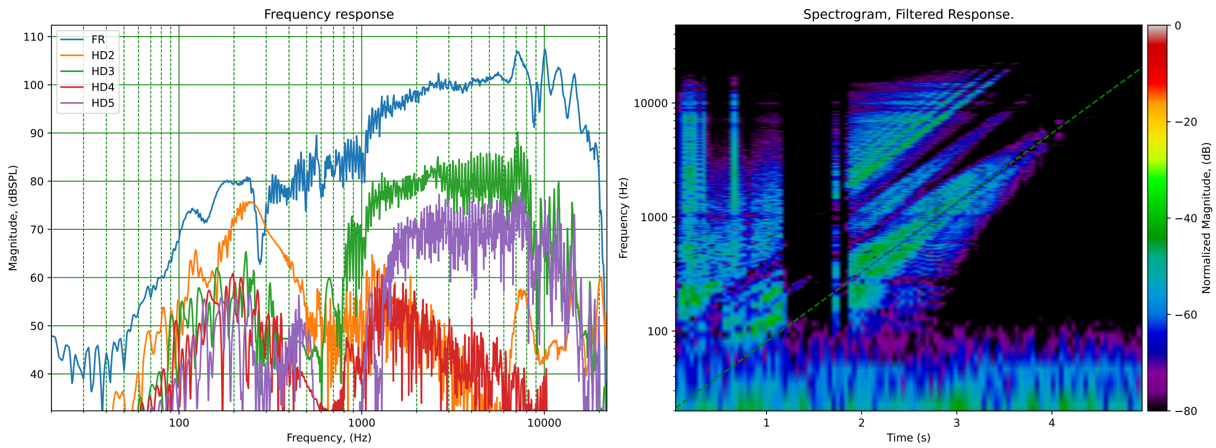


(a) Measurement while powering the amplifier with a PSU.

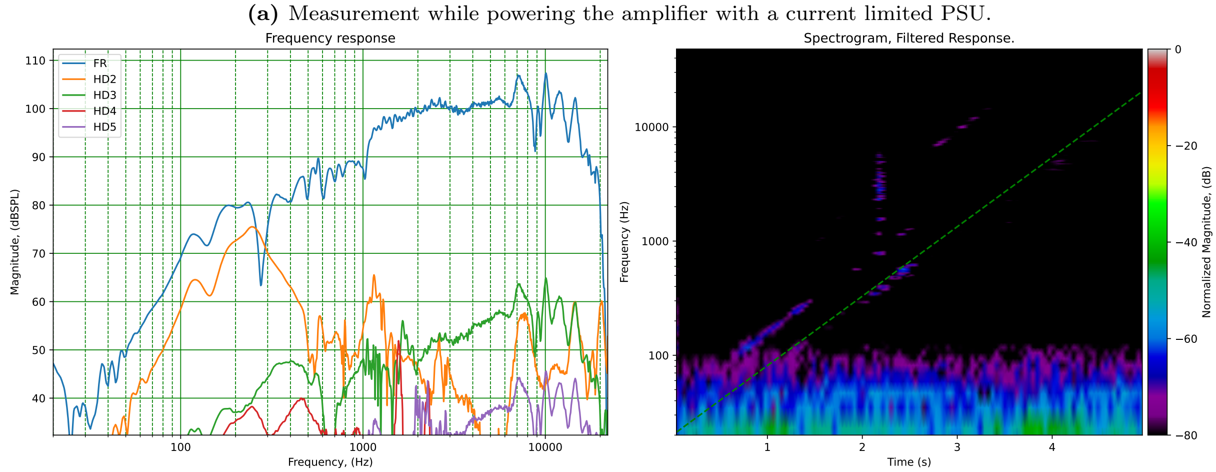


(b) Measurement where switching between PSU and battery occurs every 10 ms.

Figure 40: Acoustic SPL measurement of a 50 mV$_{rms}$ sinusoidal frequency sweep from 20 Hz to 20 kHz.



(a) Measurement while powering the amplifier with a current limited PSU.



(b) Measurement while powering the amplifier with a battery.

Figure 41: Acoustic SPL measurement of a 700 mV$_{rms}$ sinusoidal frequency sweep from 20 Hz to 20 kHz.

8 Outlook

The developed prototype mostly serves as a proof of concept - that the available power can be temporarily increased with the use of a powerbank. However, for a complete system there is a lot that remain unexplored or needs improving. Below are a few points that a finished product has to consider. Many of these remaining tasks could not be done during this project due to time constraints, and some of them could not be done due to not directing the project towards a certain application or product.

- Custom circuit board for all components.
- Charging part of the system.
- Define thresholds for each speaker system.
- Explore the most suitable powerbank for each application.
- Integration to SoC/DSP.
- Optimize the software.

In the prototype built, all components are mounted on evaluation modules, see Figure 34. These modules are meant to test their respective component in a certain setting and they are therefore very limited. All the components used can without a doubt be placed on a single circuit board, since much of the space taken on EVMs are test points or extra resistances or capacitors for testing. Designing and printing a circuit board is time consuming, and not something that fit the time frame of the project, since designing it would be done after the prototype was complete and evaluated, which is at the end of the project.

Defining thresholds for when the algorithm decides to toggle the MUX is easily done. However, since there haven't been a clear application or use case for the system since the start, it is simply unnecessary. All speakers, even though using the same PA, will load the system differently and draw different amount of current with regards to input amplitude. If a speaker is to use a system like this, then one would simply have to evaluate which input level corresponds to what output power and define the threshold thereafter.

Same principle applies for determining which powerbank is most suitable. It highly depends on the speaker system and what it is used for. A smaller system with less demanding speakers and fewer burst of high amplitude would require a much smaller powerbank, while more demanding system (for example outdoors Public Announcement speakers) would require powerbank with higher capacity.

Integrating the software is something that both requires a specific product to work with as well as being very time consuming. The prototype built runs, as previously explained, on Python code on a PC which controls an Arduino and uses an external DAC, or runs Python code directly on a Raspberry Pi and utilizes its internal DAC. The Raspberry Pi setup is as close a prototype can get to a real application. The change would not be drastic but the audio path would slightly change. Instead of the digital audio going directly to the DAC after DSP it would have to pass through the algorithm. Usually DSP is done in the System on Chip (SoC) and the algorithm could in all probability even be integrated into the DSP.

The charging part of the system has been a part of the plan for the project since the beginning. The idea was to include a charger and BMS to allow the powerbank to recharge. However, it became clear almost from the start that this would be outside the scope of the project due to it being very complex and time consuming. Despite not having time to properly explore the charging part of the system, it has been included in the plan and big picture of the system and considered throughout the project. Figure 42 below shows a hypothetical diagram of how the solution for the power flow could end up. An algorithm toggles the MUX to switch between the standard power source, in this case the PoE, and the powerbank. The algorithm would in this implementation have to depend both on the SOC of the powerbank in addition to the audio input signal. The output of the MUX is then connected to a Boost Converter which stabilizes and increases the voltage delivered to the PA, which then drives the speaker, just as it did previously.

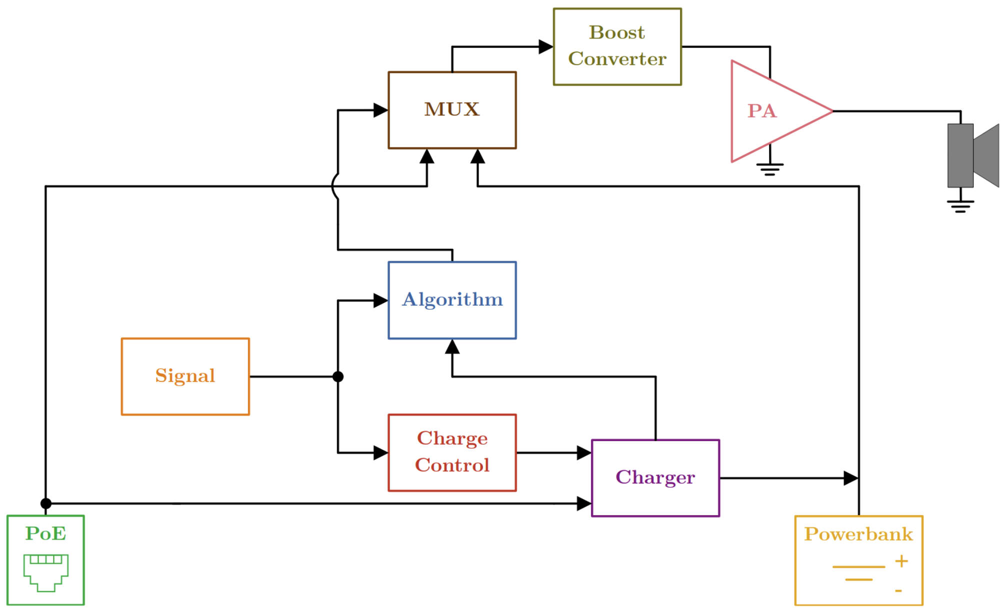


Figure 42: Block circuit of the power flow including the different components.

An idea of how a this system might function using a MUX can be seen in the Sequential Function Chart (SFC) in Figure 43. The amplitude of the input audio signal is measured and if the signal exceeds a predetermined threshold, the SOC of the powerbank is checked to see if the powerbank has enough charge to deliver power. Then, the charging is turned off and the system switches from the PoE to the powerbank. After this, the system should wait for some time (a few ms) and restart the processes, that is, measuring the input signal again.

In the opposite case, if the signal's amplitude is below the threshold, the system should keep supplying from the PoE and not switch to the powerbank. Instead the SOC of the powerbank should be checked. If the powerbank is fully charged, the system should simply wait and loop back. If it is not fully charged, the current going into the PA from the PoE should be measured to get the available left-over power not used by the PA. The charger should then be limited to charge the powerbank with this left-over power only.

It could be problematic if the threshold is exceeded and the powerbank is not sufficiently charged. In this case, the system will not be able to switch, and the signal will either be clipped, the PoE system will shutdown because the power limit is exceeded, or the signal has to be played at a lower volume. Therefore, it is important that this case is avoided. By implementing a good charging system and a reasonable threshold limit that ensures the powerbank is always adequately charged, this scenario can hopefully be circumvented. However, this entire sequence might also need to be handled differently.

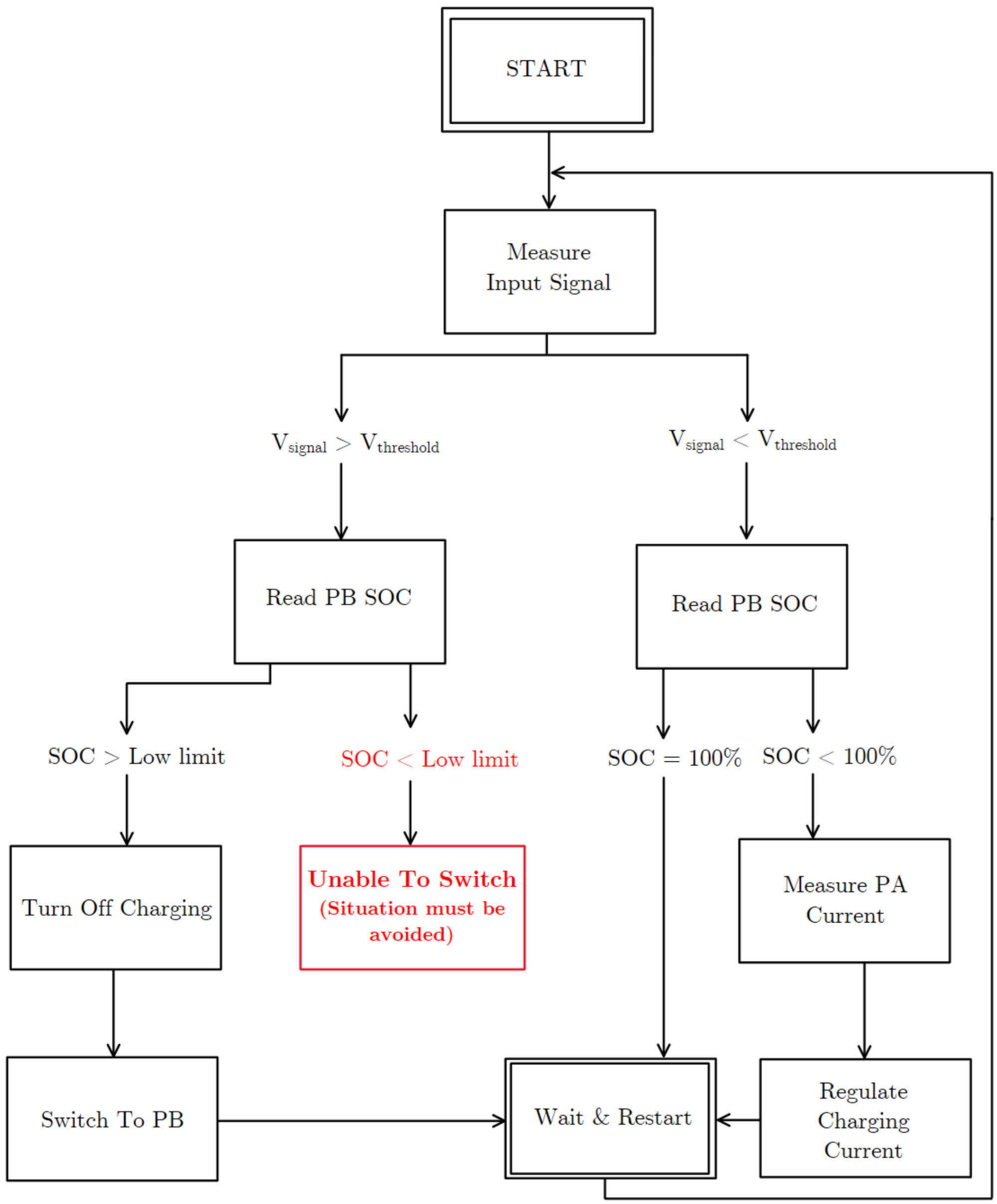


Figure 43: Hypothetical Sequential Function Chart (SFC) of the system.

9 Discussion

It would be gratifying to be able to say that the results speak for them selves, but unfortunately they are not self explanatory, and they might not be easily interpreted as a clear improvement to speakers. If it was possible to provide a snippet of audio in the report it would be easier to show a direct improvement. Measurements have been done in a anechoic chamber to achieve measurements and graphs that provide a visual interpretations of the audible improvement while running the algorithm and powerbank. This proved however to be a wasted effort, as the clippings which are very audible does not appear as clearly in the recorded waveforms. The best guess as to why, is that the microphone and measurement equipment reconstructs and polishes the signal.

The measurements that have been obtained shows a clear improvement. The first measurement, 35 clearly displays how the powerbank solution removes clipping, and simultaneously that the algorithm tracks the peaks really well. The ripple effect on the MUX output can be attributed to the inherent chemistry in lithium ion batteries when current is drawn. This is fortunately not an issue since the boost converter stabilizes the output.

The following two figures, 36 and 37, paints a promising picture. The two measurements without the powerbank starts to clip early due to power constraints and can therefore never increase any further. The sharp drop off in voltage is due to a safety feature in the power supply. As can be seen, the measurement with the powerbank active does continue to increase with increasing input, however, it slowly tapers off. This is because the power supplied from the powerbank is theoretically unlimited, but the amplifier has a power limit. Figure 37 is the same measurement but acoustic. It plainly demonstrates the potential in a powerbank solution. The measurement with the powerbank active allows for an increased output volume of 7 dB and above. However, this should not be the case, since theoretically the sound pressure should only increase by 6.3 dB, see Equation 18. The improved result is probably due to the current limiting in the power supply not properly limiting the current and decreasing the power below 7 W.

$$10 \log\left(\frac{30 \text{ W}}{7 \text{ W}}\right) = 6.3 \text{ dB} \tag{18}$$

Figure 39, shows that, even without proper filtering, neither the components nor switching adds any substantial distortions to the signal, which was one of the goals set for the project. The acoustical measurements paints a similar picture. The first one, Figure 40, is as explained essentially an acoustical recreation of Figure 39, which provides the same result. It does not however test each individual components effect on THD, mostly because it was deemed unnecessary after seeing the minimal change on electrical THD, and partly because acoustical THD is not very closely related to actual audible distortion. The last measurement, Figure 41, is probably the most interesting measurement. It clearly demonstrates how the signal distorts when driven by a limited power source, such as PoE, and the amplifier clips the signal, as in Figure 18. It also shows how the signal is "supposed" to appear when driven by an unlimited source, such as a battery powerbank in this case. This is what would happen in a real situation with the exception that in this measurement the entire sweep clips, while in a more realistic situation only the peaks would be clipped. Even so, the effect of power limiting is clear.

An observation that was made relatively late in the project was when measuring the speaker's output in the setup without using the BC. No noises or "click" sounds were audible, despite the PSU and powerbank having different voltages. This was the main concern and the main reason for including a BC and suggests that the BC may be redundant for this application. However, more accurate and dependable measurements need to be conducted, including testing with different speakers, to validate this. The primary reason this has not been done is that the EVMs used for the MUX does not properly support testing with a large voltage difference between the inputs, which makes tests unreliable.

The prototype that has been built serves mostly as a proof of concept. It demonstrates that it is possible to use a powerbank to supply higher power whenever necessary. Despite this, the way it is built is most certainly not the best way. There are a lot of improvements that could potentially be made to the system. Many naturally comes when developing the product further, as explained in Section 8. Developing the charging part for example would require changes to the algorithm to include the SOC and buffer times to allow the charger to switch modes. Another very easy potential improvement is simply to change components. The components used in the prototype were selected after weeks over research and searching, but after evaluating them and reconsidering the requirements of the components there are in all probability better suited components on the market. The algorithm could potentially be improved as well, especially the triggering signal. The algorithm was developed during a week to ensure the MUX switched as coveted for testing. As a short reminder, it works by dividing the digital signal into chunks, analyzing each chunk and determining whether the powerbank is needed or not. The output is then controlled by a logical function creates buffer zones around chunks that does or does not require the powerbank. This was done to avoid certain circumstances, but depending on the application, those circumstances may never arise, rendering the need for the logical function nonexistent. Alternatively, there could be other circumstances that require a different type of logical function.

There is definitely room for improvement in the triggering signal, even though no direct improvement could be found in this work. Most of the alternatives have already been discussed in Section 6.2.5, but there are other signals that can trigger the MUX to switch. For example the current drawn could be measured or the analog signal before the PA could be directly used. However, the reason none of these were explored were because potential problems with synchronization between MUX switches and the audio signal. The purely digitally based system seemed suitable since the synchronization can be manually controlled via delays.

The two other power combining methods presented in Section 6.1 were obviously never tested, which is unfortunate, since it would have been interesting to compare the different methods. Apart from the time constraint impeding its exploration, there is also the fact that the power MUX method functioned really well and it was deemed more reasonable to continue work on the MUX instead of testing another solution. This has repeatedly been the case during this project where several solutions were explored, one was tested and performed above expectation, and therefore the others were deemed unnecessary, for example with triggering signals or with the algorithm.

Implementing solutions that involve energy storage devices, such as rechargeable batteries, has environmental impacts for future products. Li-ion batteries contains harmful Lithium salts and flammable organic solvents. Some types contain hazardous heavy metal such as cobalt (Co). Concerns about cobalt miner's health are particularly significant due to their vulnerability and exposure and the mining of lithium has a negative impact on local biodiversity due to contamination of soil and air [39]. If these batteries are not handled and recycled properly these dangerous substances can impact the environment negatively. [40]. Even though methods for recycling can reduce the impact and the wasting of resources, these recycling processes are complicated, expensive and consumes energy.

If the prototype was instead implemented with supercapacitors, which also requires a specific recycling procedure, it would have comparatively much less complicated composition of materials and does not contain any expensive metals [41]. The result is that supercapacitors are easier to recycle than batteries [42]. Therefore, supercapacitors could definitely be a strong candidate to replace batteries.

Including a powerbank to increase the total power output of loudspeakers allows users to play sounds at higher SPL. This can be problematic if mishandled, as excessively loud sounds can cause hearing damage. Additionally, loud sounds can interfere with nearby wildlife and ecosystems.

10 Conclusion

In conclusion, the prototype developed serves as a proof of concept to whether or not a powerbank can be utilized to improve PoE speaker system with regards to power output. Testing has shown that the prototype functions as expected. It switches sufficiently fast on command between the two power supplies, does not introduce interference or harmonics, the powerbank can supply enough power to avoid clipping or simply allow a higher SPL. As such, all of the goals that were set for the project were achieved. However, there are many other questions that needs answering before this concept could actually be implemented in a product, many of them raised in the previous two sections. But the main questions that have to be answered are: *Is this the correct solution for the application and product?* and *Can it be implemented without increasing the cost of the product too much?*. Quite obvious questions but they do need answering before any dared ventures. However, with the obtained results from measurements, the main questions asked at the beginning have been answered. It is indeed possible to increase the total sound pressure by including a powerbank without deteriorating the audible quality and experience. The powerbank implementation can also be used to avoid power limited clipping of signals and can, with an advanced algorithm, be performed in a regulated, dynamic and automated fashion.

References

[1] M. Kochanowska, W. R. Gagliardi, and with reference to Jonathan Ball, "The double diamond model: In pursuit of simplicity and flexibility," in *Perspectives on Design II: Research, Education and Practice*, D. Raposo, J. Neves, and J. Silva, Eds. Cham: Springer International Publishing, 2022, pp. 19–32, ISBN: 978-3-030-79879-6. DOI: 10.1007/978-3-030-79879-6_2. [Online]. Available: https://doi.org/10.1007/978-3-030-79879-6_2.

[2] G. Mendelson, "All you need to know about power over ethernet (poe) and the ieee 802.3 af standard," *Internet Citation,[Online] Jun*, p. 13, 2004.

[3] https://www.burglaryalarmsystem.com/technology-news/poe-ieee-802-3af-and-ieee-802-3at.html, "Poe ieee 802.3af and ieee 802.3at," Unifore, Tech. Rep., 2014.

[4] "Ieee standard for ethernet," *IEEE Std 802.3-2022 (Revision of IEEE Std 802.3-2018)*, pp. 1311–1403, 5643–5813, 2022. DOI: 10.1109/IEEESTD.2022.9844436.

[5] L. Shen, "Power over ethernet (poe) technical overview," Tech. Rep., Jan. 2019. DOI: 10.13140/RG.2.2.11680.53769.

[6] N. Coruh, S. Urgun, and T. Erfidan, "Design and implementation of flyback converters," pp. 1189–1193, 2010. DOI: 10.1109/ICIEA.2010.5515894.

[7] J. Kadis. "Dynamic range processing and digital effects." (2018), [Online]. Available: https://ccrma.stanford.edu/courses/192b/192b_Lecture_1_18.pdf.

[8] R. Hansen. "Dynamic range compression: Techniques, applications, and tips." (2023), [Online]. Available: https://soundscapehq.com/dynamic-range-compression/.

[9] "Crest factor." (2023), [Online]. Available: https://en.wikipedia.org/wiki/Crest_factor.

[10] X. Jiang, "Fundamentals of audio class d amplifier design: A review of schemes and architectures," *IEEE Solid-State Circuits Magazine*, vol. 9, no. 3, pp. 14–25, 2017. DOI: 10.1109/MSSC.2017.2712368.

[11] Y. Kang, T. Ge, H. He, and J. S. Chang, "A review of audio class d amplifiers," pp. 1–4, 2016. DOI: 10.1109/ISICIR.2016.7829693.

[12] "Using pwm to generate an analog output," Tech. Rep. TB3250, 2020. [Online]. Available: https://ww1.microchip.com/downloads/en/Appnotes/90003250A.pdf.

[13] J. Mitchell, *Section 2 - Loudspeakers*, G. Ballou, Ed. Boston: Focal Press, 2009, pp. 195–278, ISBN: 978-0-240-81267-0. DOI: https://doi.org/10.1016/B978-0-240-81267-0.00004-1. [Online]. Available: https://www.sciencedirect.com/science/article/pii/B9780240812670000041.

[14] "Cutaway view of a dynamic loudspeaker for the bass register." (2014), [Online]. Available: https://en.wikipedia.org/wiki/Loudspeaker#/media/File:Loudspeaker-bass.png.

[15] "Electrical characteristics of dynamic loudspeakers." (2023), [Online]. Available: https://en.wikipedia.org/wiki/Electrical_characteristics_of_dynamic_loudspeakers#cite_ref-Davis&Jones,_p.205_1-0.

[16] "Thiele / small parameters explained with real world cases." (2016), [Online]. Available: https://audiojudgement.com/thiele-small-parameters-explained/.

[17] J. Borwick, *Loudspeaker and Headphone Handbook*. Focal Press, 2001, pp. 2–8, ISBN: 978-0-240-51578-6.

[18] "Hearing range." (2024), [Online]. Available: https://en.wikipedia.org/wiki/Hearing_range.

[19] J. Borwick, *Loudspeaker and Headphone Handbook*. Focal Press, 2001, p. 555, ISBN: 978-0-240-51578-6.

[20] L. Z. William Shepherd, *Power Converter Circuits*. CRC Press, 2004.

[21] D. M. Robert W. Erickson, *Fundamentals of Power Electronics - Third Edition*. Springer Nature Switzerland AG, 2020.

[22] "Boost converter — step up chopper." (2021), [Online]. Available: https://www.electrical4u.com/boost-converter-step-up-chopper/?utm_content=cmp-true.

[23] B. Subedi. "Boost converter: Basics, working, design & application." (2023), [Online]. Available: https://how2electronics.com/boost-converter-basics-working-design-application/.

[24] R. M. M. Karl Johan Åström, *Feedback Systems - An Introduction for Scientists and Engineers - SECOND EDITION*. Princeton University Press, 2020.

[25] C.-Y. Hsu, Y. Ajaj, G. K. Ghadir, *et al.*, "Rechargeable batteries for energy storage: A review," *e-Prime - Advances in Electrical Engineering, Electronics and Energy*, vol. 8, p. 100 510, 2024, ISSN: 2772-6711. DOI: https://doi.org/10.1016/j.prime.2024.100510. [Online]. Available: https://www.sciencedirect.com/science/article/pii/S2772671124000925.

[26] B. Hariprakash, A. Shukla, and S. Venugoplan, “Secondary batteries – nickel systems — nickel–metal hydride: Overview,” in *Encyclopedia of Electrochemical Power Sources*, J. Garche, Ed., Amsterdam: Elsevier, 2009, pp. 494–501, ISBN: 978-0-444-52745-5. DOI: https://doi.org/10.1016/B978-044452745-5.00158-1. [Online]. Available: https://www.sciencedirect.com/science/article/pii/B9780444527455001581.
[27] S. Sharma and P. Chand, “Supercapacitor and electrochemical techniques: A brief review,” *Results in Chemistry*, vol. 5, p. 100 885, 2023, ISSN: 2211-7156. DOI: https://doi.org/10.1016/j.rechem.2023.100885. [Online]. Available: https://www.sciencedirect.com/science/article/pii/S2211715623001248.
[28] B. Arabsalmanabadi, N. Tashakor, A. Javadi, and K. Al-Haddad, “Charging techniques in lithium-ion battery charger: Review and new solution,” in *IECON 2018 - 44th Annual Conference of the IEEE Industrial Electronics Society*, 2018, pp. 5731–5738. DOI: 10.1109/IECON.2018.8591173.
[29] “Bu-408: Charging nickel-metal-hydride.” (2021), [Online]. Available: https://batteryuniversity.com/article/bu-408-charging-nickel-metal-hydride.
[30] M. U. Roberto Scibilia, “Power solutions for class-d audio amplifiers,” 2018.
[31] D. D. Babak Abedpour, “Power-sourced charging over poe to enable high power peaks,” 2022.
[32] “Enhance power supply protection with power oring diodes.” (2021), [Online]. Available: https://resources.system-analysis.cadence.com/blog/msa2021-enhance-power-supply-protection-with-power-oring-diodes.
[33] R. White, “Using redundant dc power in high availability systems,” in *Twenty-First Annual IEEE Applied Power Electronics Conference and Exposition, 2006. APEC ’06.*, 2006, 6 pp. DOI: 10.1109/APEC.2006.1620638.
[34] D. Ma and J. Xing, “The design of ideal diode,” in *Anti-counterfeiting, Security, and Identification*, 2012, pp. 1–4. DOI: 10.1109/ICASID.2012.6325307.
[35] “Why we use diode protection modules.” (2016), [Online]. Available: https://instrumentationtools.com/why-we-use-diode-protection-modules/.
[36] A. Triano, “Application report - basics of power mux,” Tech. Rep., 2020. DOI: 10.13140/RG.2.2.11680.53769.
[37] *Tps212x 2.8-v to 22-v priority power mux with seamless switchover*, SLVSEA3F, Texas Instruments Inc., 2020. [Online]. Available: https://www.ti.com/lit/ds/symlink/tps2121.pdf?ts=1714644593360&ref_url=https%253A%252F%252Fwww.ti.com%252Fproduct%252FTPS2121.
[38] *Tps61377 23-vin, 25-vout, 6-a, synchronous boost converter with programmable peak current limit*, SLVSH38, Texas Instruments Inc., 2023.
[39] J. B. Dunn, L. Gaines, J. C. Kelly, C. James, and K. G. Gallagher, “The significance of li-ion batteries in electric vehicle life-cycle energy and emissions and recycling’s role in its reduction,” *Energy Environ. Sci.*, vol. 8, pp. 158–168, 1 2015. DOI: 10.1039/C4EE03029J. [Online]. Available: http://dx.doi.org/10.1039/C4EE03029J.
[40] B. Huang, Z. Pan, X. Su, and L. An, “Recycling of lithium-ion batteries: Recent advances and perspectives,” *Journal of Power Sources*, vol. 399, pp. 274–286, 2018, ISSN: 0378-7753. DOI: https://doi.org/10.1016/j.jpowsour.2018.07.116. [Online]. Available: https://www.sciencedirect.com/science/article/pii/S0378775318308498.
[41] G. Jiang and S. Pickering, “Recycling supercapacitors based on shredding and mild thermal treatment,” *Waste management (New York, N.Y.)*, vol. 48, Nov. 2015. DOI: 10.1016/j.wasman.2015.10.027.
[42] O. Ali. “What is the difference between supercapacitors and batteries?” (2023), [Online]. Available: https://www.azonano.com/article.aspx?ArticleID=6519.