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A KOMUNIKAČNÍCH TECHNOLOGIÍ

DEPARTMENT OF RADIO ELECTRONICS

ÚSTAV RADIOELEKTRONIKY

MULTIFUNCTION SPEAKER WITH WIRELESS CONNECTIVITY

MULTIFUNKČNÍ REPRODUKTOR S BEZDRÁTOVOU KONEKTIVITOU

BACHELOR'S THESIS

BAKALÁŘSKÁ PRÁCE

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Bakalářská práce

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Student: Michal Baránek ***ID:*** 219836

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NÁZEV TÉMATU:

Multifunkční reproduktor s bezdrátovou konektivitou

POKYNY PRO VYPRACOVÁNÍ:

V teoretické části práce navrhněte blokovou a obvodovou strukturu multifunkčního reproduktoru, který dokáže přehrávat lokálně uloženou hudbu z paměťové karty nebo USB disku. Zařízení bude mít také schopnost příjmu audiosignálu pomocí bezdrátové technologie jako je Bluetooth nebo WIFI. Zařízení bude napájeno pomocí akumulátoru nebo síťového adaptéru.

V praktické části práce zařízení zkompletujte, oživte, naprogramujte obslužné firmwary, vytvořte vhodný kryt. Experimentálním měřením v laboratoři ověřte jeho činnost. Proveďte detailní měření parametrů multifunkčního reproduktoru. Výsledky měření zpracujte formou protokolu o měření.

DOPORUČENÁ LITERATURA:

[1] KURNIAWAN, A. Internet of Things Projects with ESP32, 2019, ISBN: 1789956870.

[2] WIRSUM, S. Abeceda nf techniky. Praha: BEN – technická literatura, 2003.

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Vedoucí práce: Ing. Jan Kufa, Ph.D.

doc. Ing. Lucie Hudcová, Ph.D.
předseda rady studijního programu

Abstract

The thesis describes a multifunction system with multiple choices to transmit the audio signal to the speaker. The issue of today's highly popular commercial portable speakers is a lack of an autonomous source of music, such as radio. This thesis aims to achieve such a multifunctional speaker that is similar to commercial portable speakers but with the possibility of playing the radio distributed via an internet network. Another of the most important features of those speakers is the wireless connectivity such as Bluetooth, for streaming the audio for example from your phone to the speaker. The next significant aspects of these speakers are a simple user interface mainly in form of control buttons, good quality audio output with reasonable output power for the size of the speaker enclosure, and great battery life for several hours of listening on low or middle volume level. That's why these features are implemented in this work of making multifunction speaker.

Keywords

audio, multifunction, speaker, wireless connectivity, Bluetooth, Wi-Fi, amplifier

Abstrakt

Tato práce se věnuje multifunkčnímu systému s více možnostmi pro přenos zvukového signálu do reproduktoru. Problémem u dnes velmi populárních komerčních přenosných reproduktorů je absence autonomního zdroje hudby, například v podobě rádia. Proto je cílem této práce právě dosáhnout takového multifunkčního reproduktoru podobnému jako jsou komerční přenosné reproduktory, ale s možností přehrávání radia distribuovaného pomocí internetové sítě. Další z nejdůležitějších funkcí těchto reproduktorů je bezdrátové připojení, jako je Bluetooth, pro streamování zvuku, například z telefonu do reproduktoru. Další významné aspekty těchto reproduktorů jsou jednoduché uživatelské rozhraní, převážně ve formě ovládacích tlačítek, kvalitní zvukový výstup s přiměřeným výstupním výkonem k velikosti reproduktoru a velkou výdrží baterie pro několika hodinový poslech na nízké nebo střední úrovni hlasitosti. Proto jsou i tyto všechny funkce implementovány v této práci výroby multifunkčního reproduktoru.

Klíčová slova

zvuk, multifunkční, reproduktor, bezdrátová komunikace, Bluetooth, Wi-Fi, zesilovač

Rozšířený abstrakt

Přenosné multifunkční reproduktory jsou v dnešní době velmi oblíbené, díky jednoduchosti použití, jako zdroj hlasité hudby a cenové dostupnosti. Díky vylepšujícím se technologiím v oblasti reprodukce, zesílení, zpracování audio signálu jsou tyto multifunkční reproduktory čím dál lepší a výkonnější. Problémem ale je, že u většiny těchto vzhledově krásných a výkonných reproduktorů chybí forma autonomního zdroje audio signálu, například v podobě rádia. Nejčastěji jsou totiž jen vybaveny funkcí bezdrátového připojení pomocí Bluetooth.

Tato práce se tedy zaměřuje přesně na tento problém a je zde snaha docílit co nejlepších parametrů jak po multifunkční, tak i po vzhledové stránce a nabídnout uživatelům lehce a intuitivně ovladatelný multifunkční přenosný reproduktor, který nejen dokáže přijímat signál pomocí Bluetooth technologie, ale i přehrávat hudbu z lokální paměti anebo streamovat rádio stanice pomocí sítě internet.

Nejdříve budou v této práci rozebrány naprosté základy, co se týče audia a zpracování těchto audio signálu pro následnou reprodukci. Pro splnění požadavku dnešního trhu budou navrhnuty řešení, které budou splňovat podmínky, které byly určeny na začátku. To zahrnuje výběr vhodného mikrokontroleru pro zpracování příkazů uživatele, zpracování signálu z různých zdrojů, monitorování funkcí atd.., výběr účinného zesilovače, který bude dostatečně výkonný, ale zároveň vhodný pro bateriové napájení, návrhy zlepšujících funkčních bloků, které by mohly zajistit, aby byl systém co neúčinnější, co se týče energie a nejjednodušší co se týče ovládání. Nakonec celou vnitřní část zasadit do moderního vzhledu, který bude pro uživatele atraktivní.

Po dokončení všech definovaných části bude multifunkční reproduktor otestován pro určení výsledných parametrů. Výsledné parametry konečného návrhu se porovnají s nejprodávanějšími přenosnými reproduktory ve stejné kategorii a vzhledem k tomuto srovnání se vyvodí závěr, jestli tato práce a její výsledný produkt dokáže konkurovat dnešnímu komerčnímu trhu.

Posledním tématem bude budoucnost a jak tento projekt ještě vylepšit a zdokonalit na co nejlepší výsledný produkt, protože na vylepšování je prostor vždy.

Bibliographic citation

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Author’s Declaration

Author:	*Michal Baránek*
Author’s ID:	*219836*
Paper type:	*Bachelor’s Thesis*
Academic year:	*2021/22*
Topic:	*Multifunction speaker with wireless connectivity*

I declare that I have written this paper independently, under the guidance of the advisor and using exclusively the technical references and other sources of information cited in the project and listed in the comprehensive bibliography at the end of the project.

As the author, I furthermore declare that, with respect to the creation of this paper, I have not infringed any copyright or violated anyone’s personal and/or ownership rights. In this context, I am fully aware of the consequences of breaking Regulation S 11 of the Copyright Act No. 121/2000 Coll. of the Czech Republic, as amended, and of any breach of rights related to intellectual property or introduced within amendments to relevant Acts such as the Intellectual Property Act or the Criminal Code, Act No. 40/2009 Coll., Section 2, Head VI, Part 4.

Brno, June 1, 2022

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author’s signature

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I would especially like to acknowledge my supervisor, Ing. Jan Kufa, Ph.D. for his valuable advice, great communication, and support with multiple issues I had through the whole way of making this thesis. Thank you very much.

Brno, June 1, 2022 --

Author's signature

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INTRODUCTION

Currently, there are several manufacturers that offer high-quality multifunction portable speaker systems with wireless connectivity. These speakers must meet a range of features, from long battery life to great audio output. With growing competition and new technology, these speakers are constantly being improved and are at the forefront of the best-selling speaker systems.

In this work, an illustrative procedure of processing such a portable speaker should be captured but with the advantage that these speakers do not have, and it is an autonomous source of music in the form of a radio distributed via an internet network. That is the main objective of this work.

The first phase is what exactly such a speaker does. Clarifying the basics of acoustics and sound, how sound is created or propagated at all, and how our human senses react to it. This is followed by the history of a dynamic speaker or just a loudspeaker, which we all know and use every day, and how this type of speaker works. Obviously, cannot forget about low-frequency amplifiers for amplifying and processing a small signal from a source such as a mobile phone, which propagates wirelessly via Bluetooth, etc., and such a signal would not excite any speaker at the output without amplifying.

Then follows the second phase and a substantial part, which deals with this work, and it is the multifunctional speaker itself. The focus is set on audio transmission as well as power options and enclosure for the finished speaker. The audio transmission section describes standards such as the widely used Bluetooth or Wi-Fi, which fall into wireless communication and local playback for example using memory from an SD card. As stated at the beginning the intention is a portable multifunctional speaker, so the main look is on rechargeable batteries, but there is also a brief mention of switching power supplies, which play its role in charging the accumulators.

In the last phase, emphasis is placed on the design of both hardware and software. After considering and defining the properties of a multifunction speaker using a block diagram that the speaker should be able to perform, the appropriate hardware is selected on this basis and the corresponding software is implemented which performs all the necessary functions.

In the end, these decisions are tested and optimized for the best possible result of the multifunction speaker, and subsequently, the results of achieving the set goals are discussed, compared with the competition, and the future of this work.

1. AUDIO BASICS

Whether you are new to sound or just need to freshen up, in this chapter you will learn the basics of what sound is and how it works. So, what exactly is sound? Sound is simply a rapid variation of fluctuating waves of high and low pressure through some medium. In most cases in the air, but this is not always the case. As such, the sound is transmitted not only by a gaseous medium such as mentioned air but also, for example, by a liquid such as water (speakers in swimming pools) or a solid material such as steel, etc. Let's consider just air as the medium since it is the air where the sound propagates in most cases.

1.1 Acoustic fundamentals

Imagine throwing a stone into the water. The water is compressed downwards and then bounces upwards by the force of the falling stone. This causes the circles to radiate from the center where the stone landed so-called waves, as seen in Figure 1.1. [1] Something similar is happening in the air. Imagine a sound source, such as a speaker playing harmonic sound. Just as in the first case speaker acts as the stone and the air as water. Moving the diaphragm from the speaker makes force that compresses and rarefies the air particles around the speaker producing the sound waves in the same way as in Figure 1.2.

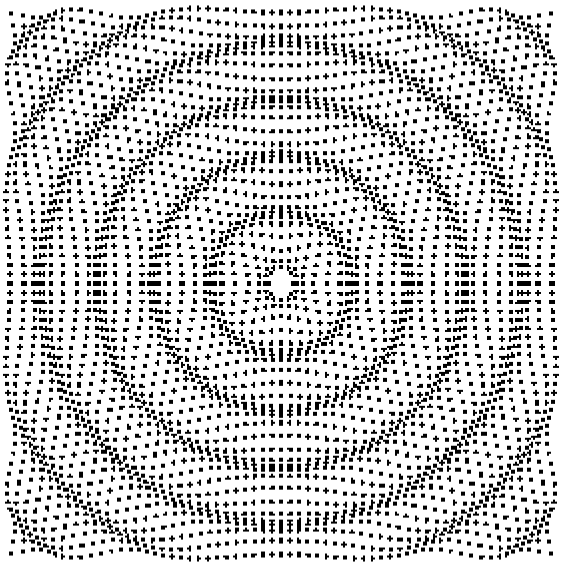

Fig. 1.1 Example of radiation the waves from the center of impact [2]

1.1.1 Sound propagation

Sound needs some medium to propagate. This medium is in most cases air, as indicated at the beginning of the chapter. The movement of the speaker's diaphragm not only affects the air molecules surrounding it, but it causes a chain reaction. Just like the example of the stone in the water, sound waves ripple outward until the air pressure settles back to equilibrium. [1]

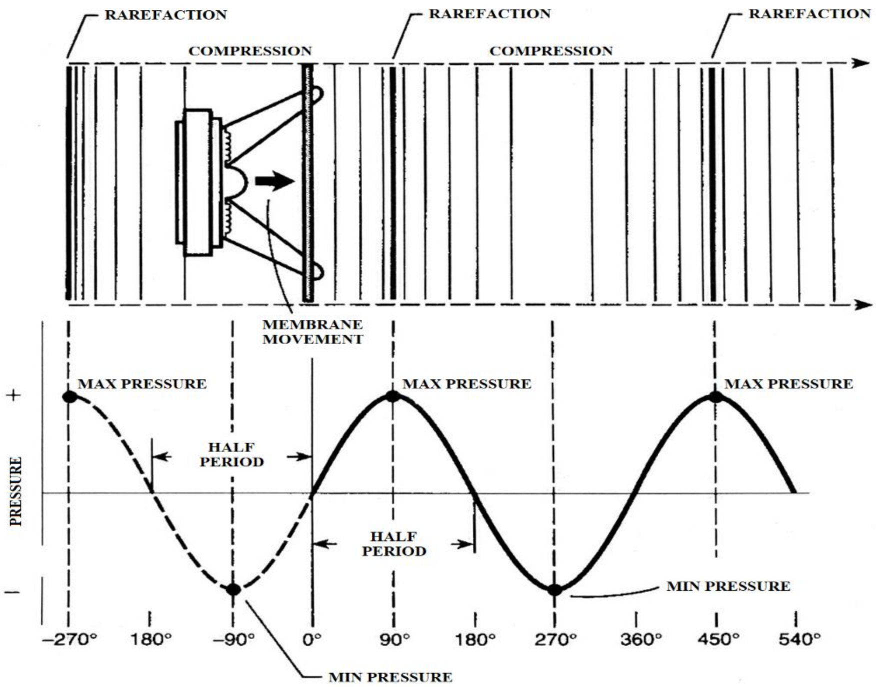

Fig. 1.2 Illustration of sound creation [3]

When you hear the sound from a speaker or any kind of sound source, in fact, you are hearing the vibrations in the air that in this situation the speaker's diaphragm causes. The sound wave that reaches your ear is the result of energy being transferred from one particle to another. As illustrated in Figure 1.3, the particles are barely displaced. [1]

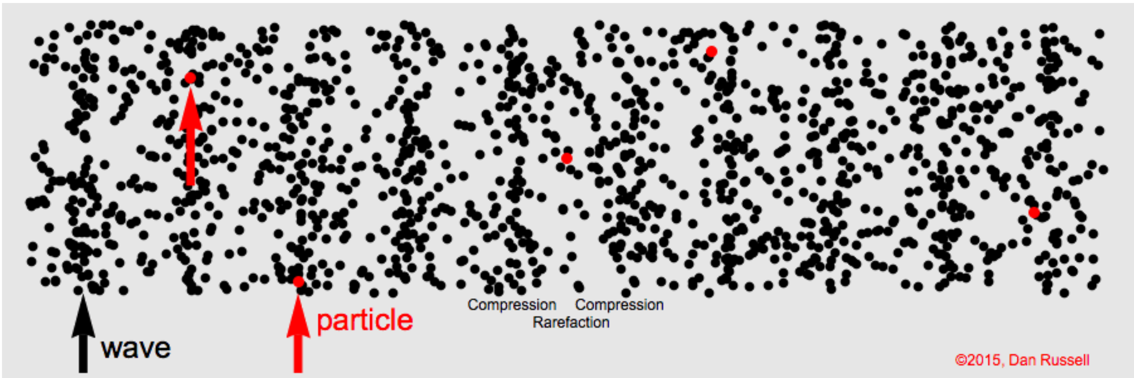

Fig. 1.3 Sound wave propagating through the air particles [2]

Figure 1.1 shows the radiation of waves from the source in two dimensions. Sound waves propagate through space in three dimensions (spherical) and with increasing distance, their acoustic energy decreases due to the loss of energy in the medium as it is partially absorbed and converted into thermal energy. This thermal energy is almost negligible even at very strong sound pressure levels. An idea of how the acoustic energy decreases depending on the distance from the source is given in Table 1.1. [4]

Table 1.1 Dependence of the sound pressure level drop on the distance from the source [4]

Distance [m]	**Sound pressure level [dB]**
1	0
2	-6
4	-12
32	-30
100	-40
400	-52

1.1.2 Sound speed and wavelength

The sound propagates through space according to the rules of wave propagation. Therefore, sound pressure variations are often called sound waves. [5] The speed of these sound waves is determined primarily by the medium in which is the sound carried and physical parameters such as temperature, density, pressure, humidity, etc. Let's stay with the air, which has its chemical composition and physical properties. For simplicity, the speed of sound in the air is $v_0 = 344\ m/s$, which is the standard established on a normal day temperature of 20º Celsius, atmospheric pressure $p = 100\ kPa$, and density of air $\rho = 1.225\ kg/m^3$.

According to wave theory, the relationship between frequency f and wavelength λ (see Figure 1.4) is

$$\lambda = \frac{v_0}{f}\text{, [m]} \tag{1.1}$$

For example, a musical tone A above middle C corresponding to a sound frequency of 440 Hz, which serves as a standard for tuning musical instruments, is a wavelength of 78.2 cm. Sound wavelengths are large. [5]

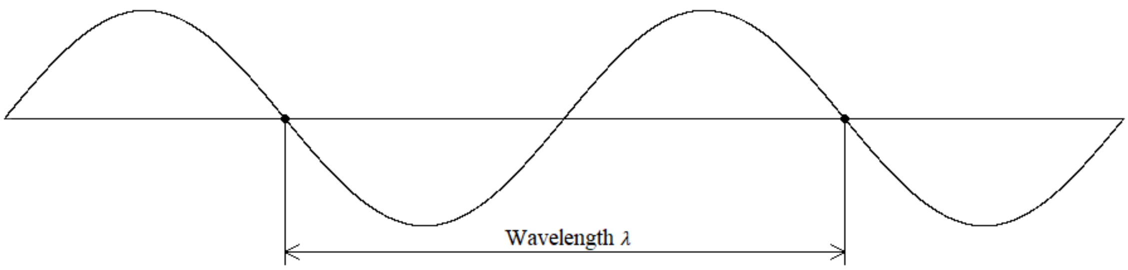

Fig. 1.4 Wavelength on the harmonic signal

1.1.3 The sources of sound

Sounds are created by many natural processes, such as the rustling of tree leaves in the wind, the falling water of a waterfall, the singing of birds, and so on. These are all-natural sounds. Similar pressure variations are produced by many artificial systems, sometimes intentionally and sometimes not. A pianist playing on the piano makes an intentional sound for musical enjoyment, while the squeaking sound produced by a braking train is generally considered as an unpleasant unwanted sound.

The electronic sound is called audio. The idea of electronic sound reproduction is to deliver sound waves in space and at a given time, which the listener will perceive as the same as if he heard the source directly. Other objectives are to enhance the natural sound or create new sounds that do not exist in nature at all. [5]

1.2 Human hearing

The sound quality of an audio system can only be assessed by human hearing. Many items of audio equipment can be well designed only with a good knowledge of the mechanism of human hearing. Human hearing in combination with the brain can receive sound waves in a wide range of amplitudes and frequencies, immediately recognize the direction, characteristics of the source, and information content in the case of speech. [5], [6]

1.2.1 Properties of human ear

The human ear can perceive sound in the range of 20 Hz to 20 kHz. Of course, each person has a different hearing, differently sensitive to a certain frequency and differently frequency-shifted range but approximately every ten years, the upper limit of the frequency range is reduced by 1 kHz. [4]

The ear is not equally sensitive to all frequencies of the sound spectrum. It is most sensitive in the range of 2 to 4 kHz. As for the dynamic range, it is capable of operating in the range of up to 140 dB. Thanks to the perception of the phase, intensity, and frequency response of the signal between the left and right ear, the brain can evaluate and determine relatively accurately the direction from which the sound is coming. [4]

1.2.2 Loudness thresholds

The human response to the amplitude of sound waves is known as loudness. It ranges from the threshold of audibility at low sound levels to the threshold of pain for very loud sounds. A reference level of 0 dB is usually taken as the hearing threshold, as seen in Figure 1.5. 120 dB is approximately the pain threshold. The sound at this level is uncomfortably loud. At higher sound levels, actual hearing damage may occur. [5]

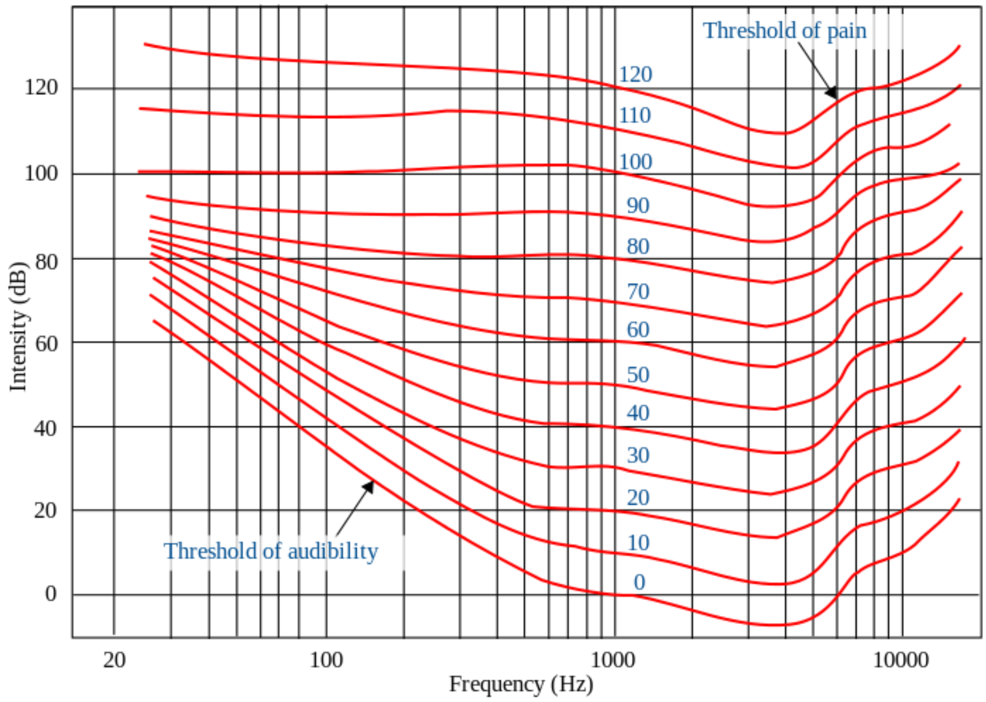

Fig. 1.5 Fletcher-Munson curve of Equal-loudness contour [7]

2. LOUDSPEAKERS

Now it is clarified what sound is and how it propagates but now let's look at the most common method of producing sound in the audio world.

A loudspeaker or just a speaker is a device that converts electrical energy (audio signal) into acoustic energy (sound), so it is sometimes called an electroacoustic transducer. Let's define exactly what we want from a speaker.

- accurately reproduce frequencies over the entire audible range
- nondirectional radiation
- frequency-independent impedance
- zero distortion
- arbitrary diaphragm deflection
- maximum efficiency and sensitivity

Of course, these assumptions cannot be encountered in the real world. Each speaker has parasitic properties that cannot be removed. [3]

2.1 History

In the past, there is a lot of people and inventions that stand behind the history of loudspeakers. The focus is placed on the dynamic loudspeaker or in other words moving-coil transducer. The first who describes a dynamic speaker or moving-coil transducer, with a circular coil of wire in a magnetic field and supported so that it could move axially was in 1874 Ernst W. Siemens. However, he did not use it for audible transmission. [8]

In 1898, Oliver Lodge invented an improvement of loudspeaker with non-magnetic spacers that maintained an air gap between the inner and outer poles of a moving-coil transducer. This improvement was later claimed by Pridham and Jensen in the Magnavox application in 1911 (speakers for radios and public address systems). All these used horns to amplify the sound produced by a small diaphragm. [8]

2.1.1 Today's speaker

The most common type of today's speaker is commonly called a dynamic speaker. This speaker uses in most cases paper diaphragm or cone connected to a rigid gasket or frame by a flexible suspension called a spider that restricts the voice coil to move axially through a cylindrical magnetic gap, as detailed in Figure 2.1. The protective dust cap glued in the center of the diaphragm provides protection against the dust rushing into the gap.

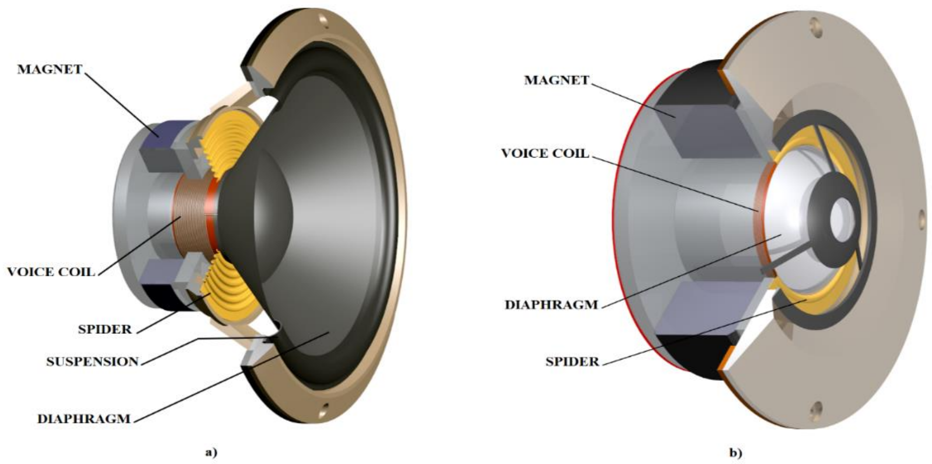

Fig. 2.1 Cutaway view of a dynamic loudspeaker a) subwoofer b) tweeter [9]

When an alternating current is applied to the speaker voice coil, a magnetic field begins to induce around the coil, which creates a variable electromagnet from the coil. The magnetic field of the permanent magnet begins to attract or repel the coil, depending on the polarity of the alternating electrical signal. This movement of the coil also results in the movement of the speaker's diaphragm, which reproduces the sound waves.

2.2 Frequency response

This characteristic indicates the dependence of the sound pressure level at a certain distance from the loudspeaker (usually 1 meter) on the frequency at a constant level of the voltage signal at the loudspeaker terminals. The low-frequency reproduction system should faithfully transmit signals in the minimum range of 40 to 16000 Hz (preferably 20 Hz to 20 kHz) and the deviation of the characteristic in this band should not exceed the tolerance field shown in Figure 2.2. [10]

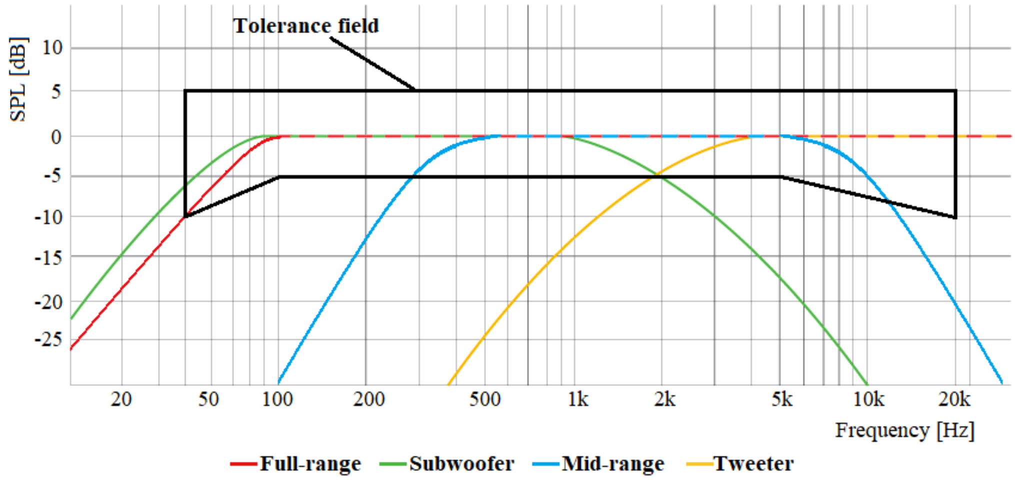

Fig. 2.2 Ideal frequency response of speakers and tolerance field

2.3 Types of loudspeakers by frequency range

The right speaker must be selected for each purpose. Without prior consideration, sound reproduction could occur inaccurately, and the quality of the entire sound system could deteriorate. Speakers can be generally divided into several groups either according to whether it is direct or indirect radiating, further according to the principle of operation, construction, method of use, etc. The focus here is only on the divisions by the frequency range.

2.3.1 Full-range

These speakers are designed for playback from bass (i.e., low frequencies) to sopranos (i.e., high frequencies). They are widely used, for example, in studio monitors, sound systems in cars, or mainly for ceiling speakers. [3] Full-range speakers have different designs:

a) **Without any additional accessories.** Principally a loudspeaker of classic design, but their re-dimensions and parameters allow a wide frequency range. They are the most affordable. [3]

b) **With diffuser (so-called dual cone, wizzer cone).** A cup-like diffuser is connected directly to the same voice coil, which creates a tweeter effect in the speaker without an additional driver, thus extending the frequency response. The diffuser can also function as a cooler for the loudspeaker's magnetic circuit, so some manufacturers make it from metal to dissipate heat better. [3] An example of this type of speaker is shown in Figure 2.3.

c) **Coaxial, triaxial.** These speakers have a coaxially mounted treble or midrange unit that is under the main diaphragm dust cap. It is the most expensive type and achieves excellent directional characteristics because it is closest to the essence of a point source. [3]

Fig. 2.3 Full-range speaker with diffuser - Visaton BG 20 [11]

2.3.2 Subwoofer

This type of speaker is among the most common. The principle is to reproduce low frequencies. When choosing these speakers, it is good to pay attention to how big the speaker cabinet is (the bigger, the better transmission of low frequencies) then also the diameter of the voice coil, at higher powers the coil heats up, and its resistance increases and as a result, the sound pressure decreases. This phenomenon is called thermal compression and affects fewer speakers with a larger voice coil. [3] An example of this type of speaker is shown in Figure 2.4.

Fig. 2.4 Subwoofer speaker - Eminence LAB 15 [11]

2.3.3 Mid-range

These speakers aim to reproduce frequencies from approximately 500 to 4000 Hz. Structurally, their gasket is both closed or open and the diaphragm is either conical or like a dome. Dome types can reproduce higher frequencies with lower distortion and without ripple the characteristic while conical diaphragm systems can operate at lower frequencies but at the cost of greater directionality at higher frequencies. Statistically, the frequency spectrum of music and speech is in the midrange, so the mid-range speaker is one of the most stressed speakers in the speaker system. [3] An example of this type of speaker is shown in Figure 2.5.

Fig. 2.5 Mid-range speaker - Eminence Beta-8A [11]

2.3.4 Tweeter

These speakers are designed to reproduce typically from 3 kHz up to the audible limit of human hearing (typically 20 kHz). The most used constructions are a conical diaphragm, dome, compression types, electrostatic and piezoelectric. In professional sound systems is mostly used the pressure type. Hi-Fi uses a dome that has lower distortion, and a more balanced frequency response but also less sensitivity. [3] An example of this type of speaker is shown in Figure 2.6.

Fig. 2.6 Tweeter speaker - Yamaha NS 40M [11]

3. LOW-FREQUENCY AMPLIFIER

From Chapter 2., is clear what a speaker is and what it does but how a speaker can reproduce a sound signal, at least as loudly as needed in certain situations. This effect requires some energy and power. For example, the audio signal from the microphone does not have enough power to excite the speaker's diaphragm to move. Therefore, it is needed to amplify this signal to a higher power level to be able to drive the speaker by this signal. That's why an amplifier is used. In the case of audio production, the low-frequency audio amplifier.

3.1 Definition

An ideal LF (Low-Frequency) or audio amplifier is an electrical two-port network that works in a frequency range from 20 Hz to 20 kHz that reproduces and processes the input signal at the output with increased magnitude and with no distortion or harm. Audio amplifiers are divided into voltage and power amplifiers.

3.1.1 Voltage amplifier

The task of these types of amplifiers (preamplifiers, intermediate amplifiers, excitation amplifiers, correctors, etc..) is to amplify the signal to the required level for further processing. [10] An important parameter of preamplifiers is, for example, their distortion or noise. Signals that would be noisy or distorted after amplification by the preamplifier would affect the output of the entire system, where it is difficult to suppress this phenomenon. So, the ideal preamplifiers should be linear, not incorrectly affect the signal source (even when it comes to mixing multiple sources), have high input impedance and a low output impedance, have as minimal noise as possible, be resistant to overexcitation, and reduce the effect of interference. [10]

3.1.2 Power amplifier

This type of amplifier must amplify both current and voltage to excite the speaker at the output with the best possible efficiency. The power amplifier can be seen as a voltage-to-current converter. As stated at the beginning of the chapter, some energy is needed to move the speaker's diaphragm. In this case, in the form of electrical power, which is then partially converted into acoustic energy (see. First chapter). This power can range from a few mW (driving headphones) to tens, hundreds of W (home audio system) to a few kW (concert sound). Power components (transistors, vacuum tubes) must withstand difficult conditions at high powers (high currents, thermal loads) and must be sufficiently well cooled. Nowadays, power integrated circuits and bipolar or unipolar power transistors are most adapted to design power amplifiers. The major parameters are frequency response (linear over the entire frequency range), gain, distortion, and noise. [10]

3.2 Parameters

The parameters of audio amplifiers vary according to the requirements and purpose of use. In general, the main parameters are the following:

3.2.1 Frequency response

The audio device should faithfully transmit signals in the range of 40 Hz to 16 kHz (preferably 20 Hz to 20 kHz) which means flat frequency response across the whole audible range. In the Hi-Fi category, the deviation of the magnitude characteristic in this band should not exceed a tolerance range of ±1.5 dB as illustrated in Figure 3.1. Phase characteristic expresses the dependence of the phase differences of the input and output signal of the amplifier on the frequency and in the multi-channel transmission, the phase of the channels should be identical. [10]

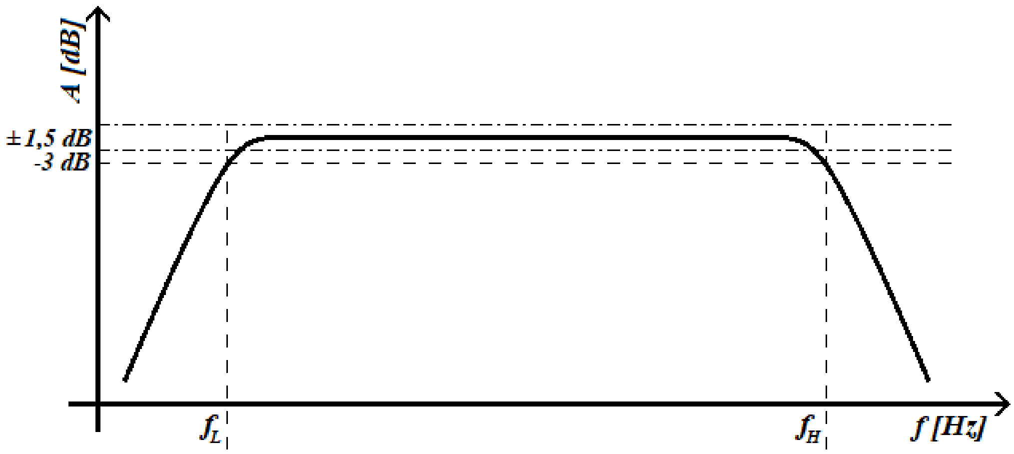

Fig. 3.1 Frequency response of amplifier with maximum deviation for Hi-Fi category

Amplifier bandwidth is often deliberately limited to ensure greater amplifier stability. The bandwidth of the amplifier can be calculated as:

$$B = f_H - f_L, \text{[Hz]} \tag{3.1}$$

3.2.2 Total harmonic distortion (THD)

With non-linear distortion, the spectrum of the useful signal is affected. This distortion is produced by, for example, a non-linear characteristic of power components such as BJT (Bipolar Junction Transistor) or a poorly set quiescent point, etc. This creates new harmonics that did not occur in the spectrum of the original signal and consequently, the signal is distorted in the time domain, as shown in Figure 3.2. The degree of this distortion is given by THD (Total Harmonic Distortion) and represents the percentage of the useful harmonic signal represented by the mixture of higher harmonic components generated by the amplifier.

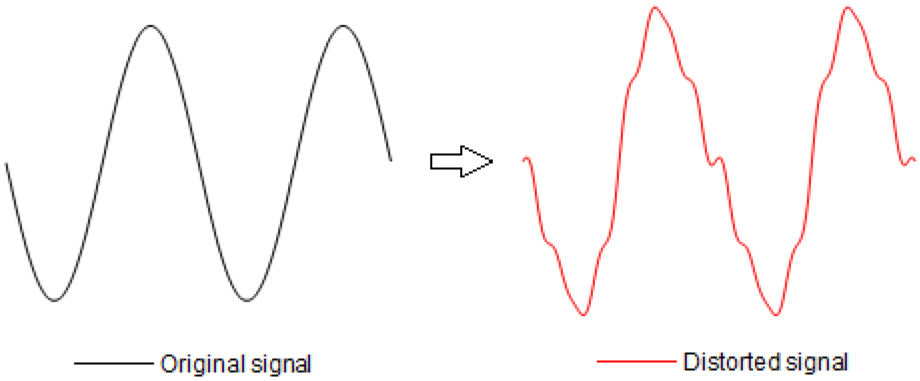

Fig. 3.2 Comparison between original and distorted signal

The harmonic distortion factor depends on the magnitude of the output voltage, thus on power and frequency. Power amplifiers should have a total harmonic distortion below 1 %. [10] THD can be calculated from the relation:

$$THD = \sqrt{\frac{{V_2}^2+{V_3}^2+\cdots+{V_n}^2}{{V_1}^2+{V_2}^2+{V_3}^2+\cdots+{V_n}^2}} \cdot 100, [\%] \tag{3.2}$$

where V_1 is the effective value of the amplified harmonic signal and V_n are effective values of higher harmonics produced by non-linear distortion.

3.2.3 THD+N

As said before the THD indicates the percentage of the useful harmonic signal represented by the mixture of higher harmonic components generated by the amplifier or other source on non-linearities. The Noise of a device is all the energy coming out of the device that is not related to the input signal. Noise sources can include power supply hum, radio frequency interference (RFI so-called EMI), switching noise of the transistors, or the thermal noise of the circuit components. THD+N is the sum of all the energy of a residual signal which consists of the harmonics and noise at the output excluding the fundamental harmonic signal of the tested tone and is expressed in Volts RMS (Root Mean Square) or a related absolute unit. THD+N ratio is the more common measurement because what a person usually wants to look at it's not the absolute distortion level, but the relative level of the distortion products compared to the total signal. As a ratio, it's usually expressed in either percent or decibels and is calculated by the formula 3.3.

$$THD + N = \frac{\sqrt{({V_2}^2+{V_3}^2+\cdots+{V_n}^2)+{V_{NOISE}}^2}}{V_1} \cdot 100, [\%] \tag{3.3}$$

Where V_1 is the effective value of the fundamental harmonic signal, V_n are the effective values of higher harmonics produced by non-linear distortion, and V_{NOISE} is the effective value of noise. An idea of a comparison between the percentage value and the logarithmic value in decibels is given in Table 3.1.

Table 3.1 Comparison between the percentage value and the logarithmic value in decibels of THD+N [4]

Percentage value [%]	Logarithmic value [dB]
100	0
89	-1
32	-10
10	-20
1	-40
0.1	-60
0.01	-80
0.001	-100

3.2.4 Input sensitivity

Indicates the amount of input voltage that is necessary to reach the rated output voltage or rated output power. If the signal would be less amplified than the specified input sensitivity, the signal-to-noise ratio (SNR) deteriorates. An important value is the magnitude of the maximum input voltage that the amplifier can process without much distortion. [10]

3.2.5 Input impedance

The input impedance of the amplifiers should be 5 to 10 times larger than the output impedance of the source, so it is important to know the input impedance to optimally match the signal source to the amplifier input. The standardized input impedance is usually 47 kΩ/250 pF or an absolute impedance at 1 kHz. The following relationship applies to the absolute value of the impedance: [10]

$$|Z| = \sqrt{R^2 + X^2}, [\Omega] \tag{3.4}$$

where $|Z|$ is absolute impedance, R is ohmic resistance and X is reactance. Overloading the output leads to a reduction in voltage and thus to a reduction of the SNR, a distortion, and a change in the frequency response. [10]

3.2.6 Crosstalk

It is the introduction of noise from other channels produced by ground loops, parasitic inductance, capacitance, etc. The risk of crosstalk rises with increasing frequency, increasing signal amplitude, length of parallel wires, and a small distance between them. [10]

3.2.7 Output impedance

Optimal power matching of the audio amplifier and speaker occurs when the output impedance of the amplifier is as large as the load (speaker) impedance. Typical load impedance values are 4, 8, 16, or 100 Ω. [10]

3.2.8 Dynamic range

The ratio between the maximum and minimum value of the signal, either sources or amplifiers. The dynamic range of the amplifier is limited by noise or hum (low frequencies) for small signals, and by the maximum output voltage or power for large signals. [10]

3.2.9 Internal resistance (impedance)

Depends on the design of the amplifier (for example, the influence of negative feedback). The smaller the internal impedance, the more the connected speakers are attenuated and the better the reproduction. The inverse of the internal resistance is the **Damping factor**. It is defined as the ratio of the rated impedance of the loudspeaker to the internal impedance of the amplifier. Its large value means strong speaker damping, which greatly reduces unwanted transients. [10]

3.2.10 Other

Not all parameters of audio amplifiers have been listed, there are several texts devoted only to amplifiers where the reader can find out all the other details. For example, other parameters include output power, supply voltage, slew rate, gain, and more.

3.3 Classes

This subchapter deals with the division of amplifiers according to their setting of the operation point (quiescent point, Q-point, bias point), and thus how many degrees of each input signal cycle is conducted. According to this operation point, amplifiers are classified into classes. Only the classes that are used for audio processing are described because other classes are either based on those or are not used in audio amplification.

3.3.1 Class A

The simplest type of amplifier.[12] The setting of the quiescent point is in the middle of the linear range of the output characteristic, as shown in Figure 3.3, i.e., the quiescent current is constantly flowing through the transistor (conduction angle is 360º), even if there is no input signal at the input of the amplifier. Due to this property of linearity, this class has a low efficiency of a maximum of about 25 % (in practice even less) and at higher quiescent currents there are demands for great cooling. Class A amplifiers are used in preamplifiers, due to their excellent linearity and minimal distortion or in demanding applications in the Hi-Fi category.

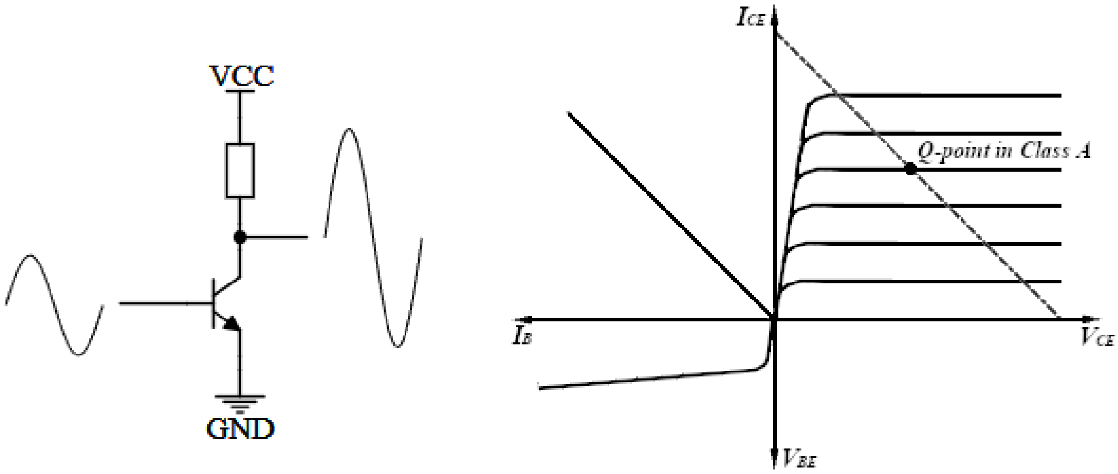

Fig. 3.3 Simplified scheme and the setting of the quiescent point of a Class A amplifier

3.3.2 Class B

The quiescent point of this class is on the line of zero collector current, as seen in Figure 3.4, which means there is no quiescent current flowing. Thus, the efficiency reaches the theoretical 78.5 % (in practice around 70 %).

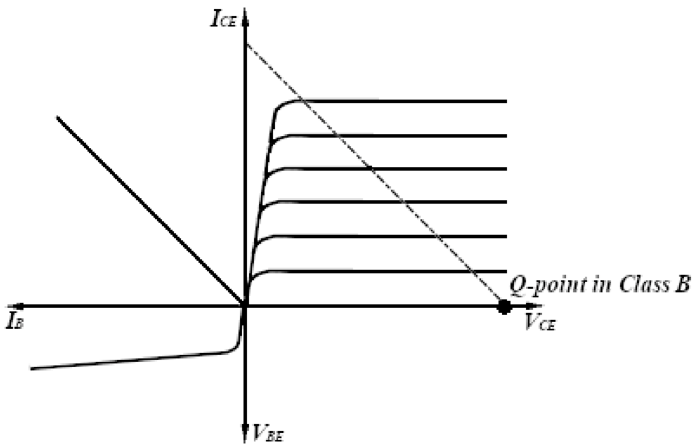

Fig. 3.4 Quiescent point of a Class B amplifier

This type of amplifier is also known as push-pull because it uses two complementary transistors that conduct only each positive or negative cycle of the signal (conduction angle is 180º). When the input signal is positive the NPN transistor conducts and the

PNP is closed. When the input signal is negative the PNP transistor conducts and the NPN is closed. Hence, the transistors have the threshold voltage from base to emitter around ±0.7 V, so it takes 0.7 V for the NPN transistor to start to conduct and -0.7 V for the PNP. This means that the part of the waveform which falls within this ±0.7 V range will not be reproduced and causes the Crossover distortion, as reflected in Figure 3.5.

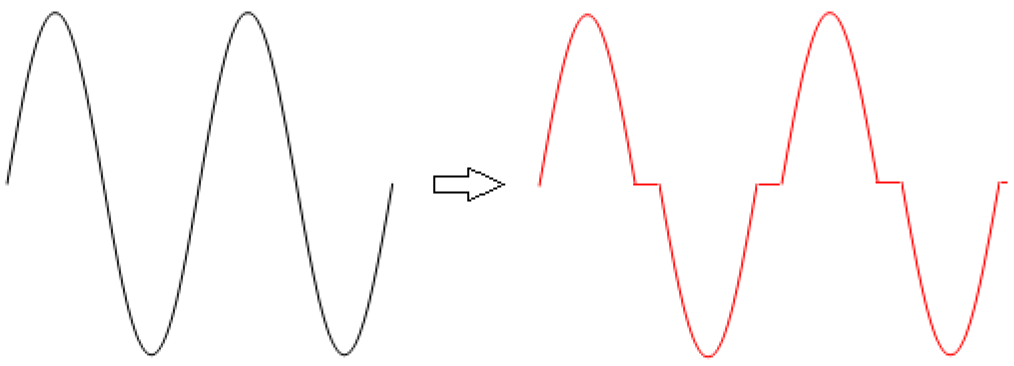

Fig. 3.5 Crossover distortion

The configuration of this amplifier can be done with quasi-complementary transistors, as seen in Figure 3.6 b), where the two same signals shifted from each other by 180º are driving the transistors. The supply voltage can be symmetrical (as shown in Figure 3.6) or unsymmetrical, but the unsymmetrical option required decoupling capacitors. [10]

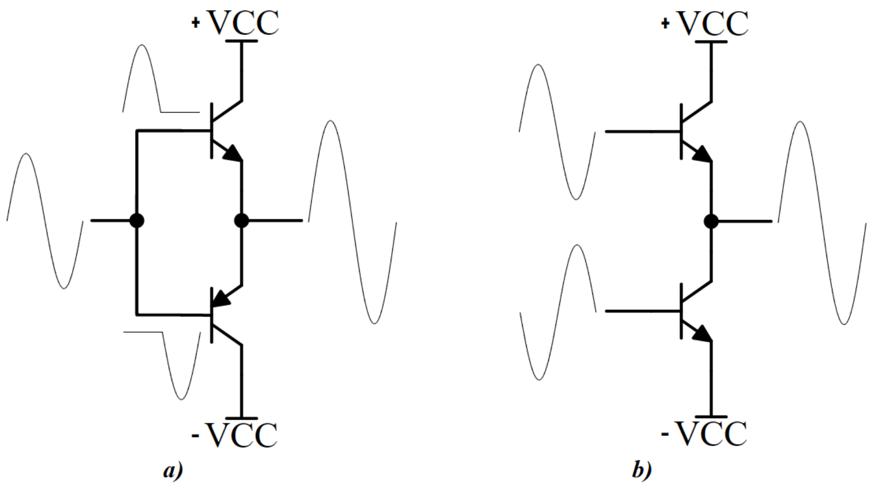

Fig. 3.6 Simplified scheme with a) Complementary b) quasi-complementary transistors

3.3.3 Class AB

Due to the high distortion, Class B is not usable for audio amplification. Therefore, there is a compromise based on Class A and B topologies. Class AB sets a quiescent point above the zero-collector current like in Class A but many times smaller (conduction angle is 270º), as seen in Figure 3.7, and thus ensures linearity of operating characteristics and avoids crossover distortion. This performance is achieved by a small bias voltage (provided by diodes, resistors, etc..) added to the transistor's base-to-emitter voltage to conduct even at a near-zero signal output, i.e., the point where Class B amplifiers introduce nonlinearities. Thanks to this improvement, the Class AB is characterized by high SNR and low THD + N. That's why it is ideal and widely used for Hi-Fi speaker drivers. The efficiency of this class is slightly smaller than Class B and is around 60%. [12]

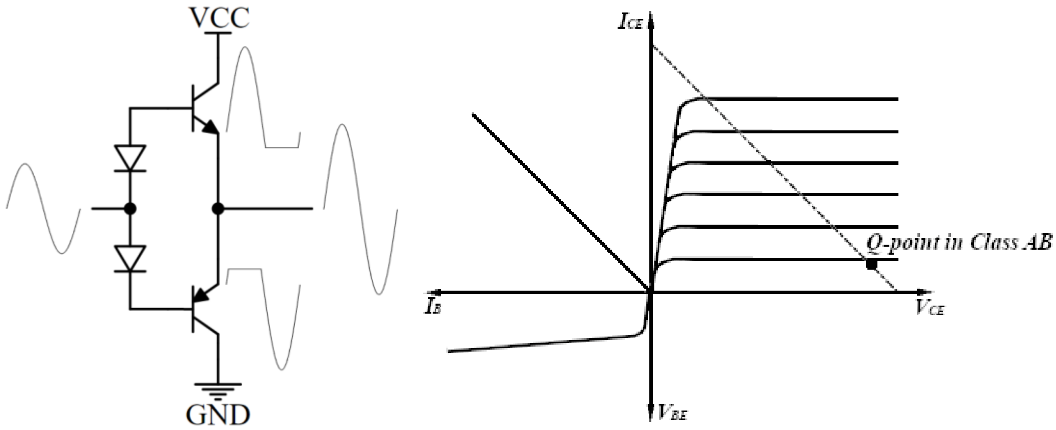

Fig. 3.7 Simplified scheme and the setting of the quiescent point of a Class AB amplifier

3.3.4 Class D

In today's portable devices, great emphasis is placed on efficiency because it is necessary to reduce power consumption to maintain the life of batteries.[12] In class D, the amplifiers use pulse width modulation (PWM) to switch the transistors, thus avoiding the linear region of the transistors, which strongly affects the final power efficiency. PWM is created by comparing the input signal with a sawtooth or triangular wave signal of a much higher frequency. In this way, the input signal is converted into a pulse stream that represents the input signal using pulse width modulation. This PWM signal then drives the switching of the transistors, which switch between the power rails to produce an amplified PWM output signal with a variable duty cycle that is filtered by a low pass filter and converted to an analog signal, as shown in Figure 3.8. This type of amplifier achieves high efficiencies theoretically up to 100 % (assuming zero transistor resistance) but in practice, they reach a maximum of about 90-95 %. Unipolar transistors (MOSFETs) are often used for switching due to their low transient resistance when fully open, thus the lowest power dissipation and low cooling requirements. In the beginning, they were characterized by greater distortion, but today they are much better off and comparable to Class AB. Class D amplifiers are therefore widely used in portable devices or high-power devices.

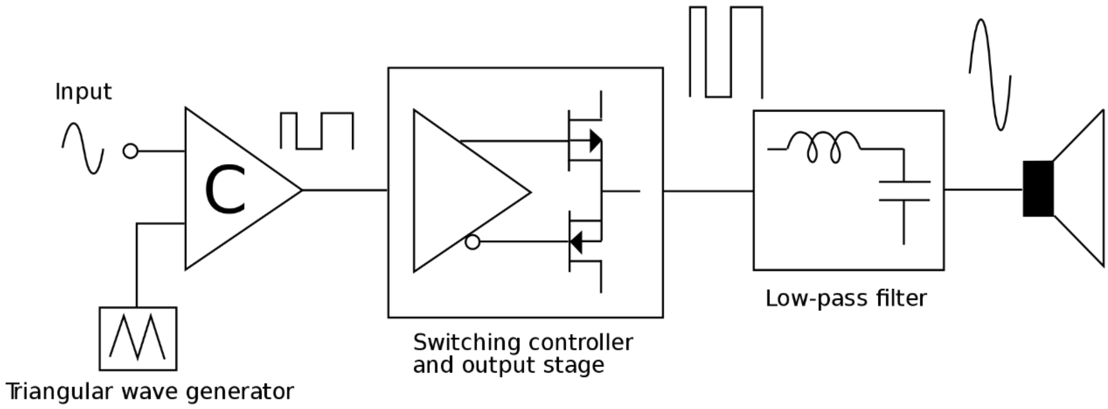

Fig. 3.8 Block diagram of Class D amplifier [13]

4. MULTIFUNCTION SPEAKER

This chapter discusses the main content of this work, the multifunction speaker itself. Today's market offers many variants of such multifunctional speakers, which have various parameters, designs, features, etc. The multifunctional speaker in this work has features such as wireless communication via Bluetooth, a slot for microSD cards, which can play music locally on the speaker, also the radio function via Wi-Fi with multiple accessible radio stations, and its unique design. The main thing that today's multifunctional speaker's lack is the already mentioned radio, either in the form of a classic FM (Frequency Modulation) radio receiver or as mentioned streamed over the internet. Therefore, the radio feature is in this work and allows the user to listen to the speaker without an external source such as a mobile phone, laptop in the case of Bluetooth, or the necessary microSD card for data storage and subsequent playback. The power supply in the form of rechargeable batteries (18650) will be integrated directly into the speaker, making it easily portable.

The next parts of this chapter show more details on how the audio signal is transmitted into the speaker, what the power supply options are, and the enclosure of the whole multifunction speaker system.

4.1 Audio transmission

The way how and by what medium the audio signal gets directly into the speaker is significant. Let's discuss ways of audio transmission, whether wireless or local, that are used in the speaker and are the worldwide standards among portable multifunction speakers.

4.1.1 Bluetooth

Bluetooth is an open standard specification for radio frequency (RF)-based short-range connectivity technology connecting two or more devices and is indicated by the logo, as seen in Figure 4.1.

Fig. 4.1 Bluetooth logo [14]

The name Bluetooth comes from the Danish King Harald Blatand, who was called Bluetooth. Harald unites the countries and expands his kingdom after his sister asked him for help in securing control in Norway. Bluetooth as a name for wireless connectivity was chosen because its developers hope it will unite the mobile world, just as King Harald united his kingdom. [15]

Any complete Bluetooth system will require the following four basic components: an RF portion for receiving and transmitting data, a module with a baseband microprocessor to process the data transmissions, link management software, and supporting application software that enables the host device to do its job. As shown in Figure 4.2 these functional blocks work together to establish the connections in the Bluetooth system. [15]

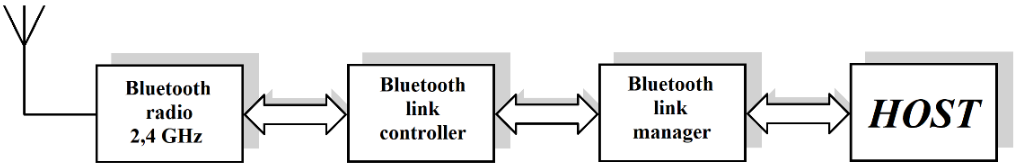

Fig. 4.2 Bluetooth function diagram

The RF portion can be implemented as a module or as a single chip operating in the globally available 2.4 GHz ISM band, the unlicensed portion of the radio frequency spectrum. To avoid interference from other RF transmissions, Bluetooth technology uses the FHSS method, which is a pseudo-random hop sequence of 79 (or reduced 23) hop frequencies spaced 1 MHz apart. The hop rate is 1600 hops per second. The available frequency width is 83.5 MHz, so the frequency band is 2.4 GHz to 2.485 GHz. The Bluetooth radio transmits data at a maximum gross rate of up to 1 Mbps, but protocol overhead limits the data rate to a little over 721 Kbps. [15]

The baseband module is the hardware that converts the received radio signals into a digital format that a host application can process. Conversely, the baseband also converts digital or voice data into a format that a radio signal can transmit. When the master unit sends data to a slave unit, it must transfer the digital data into a radio signal. The baseband compresses the data, puts the data into packets, and adds administrative information about the data packet. [15]

The link management software runs on a microprocessor to handle connections between Bluetooth devices. Each Bluetooth device has a link manager (LM) that discovers other remote link managers and communicates with them. To perform its services, the LM relies on the functions provided by the hardware component, the link controller. The link controller functions include sending and receiving data, authenticating links, and setting up link types. [15]

The application software, which is embedded in the host device, operates an application over the Bluetooth protocol stack. Typically, in an embedded Bluetooth solution, the upper-layer protocols reside on a single chip in the host device. The Bluetooth unit can then connect to the host through a serial interface such as a universal serial bus (USB). [15]

4.1.2 Wi-Fi

Everybody can often come across the term Wi-Fi in your area. Surely you already know that Wi-Fi is a technology that allows wireless access to an internet network. Wi-Fi is a trademarked logo (shown in Figure 4.3) owned by a nonprofit organization known as the Wireless Ethernet Compatibility Alliance (WECA). [16] WECA's primary purpose is to certify the interoperability of wireless LAN (Local Area Network) equipment, specifically equipment that complies with IEEE 802.11 industry standards operating in the 2400-2483.5 MHz band and the 5 GHz band, the operation is possible in the 5.15-5.35 GHz band (indoor only), in the 5.470-5.725 GHz band (IEEE 802.11a standard) on the territory of the Czech Republic. [17]

Fig. 4.3 Wi-Fi logo [18]

The idea of Wi-Fi was to replace cable systems in buildings. But it turned out that the low efficiency and low information transfer speeds of Wi-Fi could not replace the cable. However, it has become a perspective for mobile users who are not limited by cable length.

Wireless networking involves at least two radio devices. The device that you use to connect to the wireless LAN is called a station or client. The device that your station talks to is an access point (AP) that connects your station to the larger LAN.

Each Wi-Fi network has its name, which is called SSID (Service Set IDentifier). Each AP regularly broadcasts its SSID in a frame called Beacon. If the client wants to connect to the network, it must set its SSID to the same value. Then user authentication, as in Figure 4.4 is performed using algorithms such as Open System Authentication and Shared Key Authentication. After successful authentication, communication takes place between the AP and the station, which is usually encrypted. The Wired Equivalent Privacy (WEP) standard is usually used for this encryption. [16], [19]

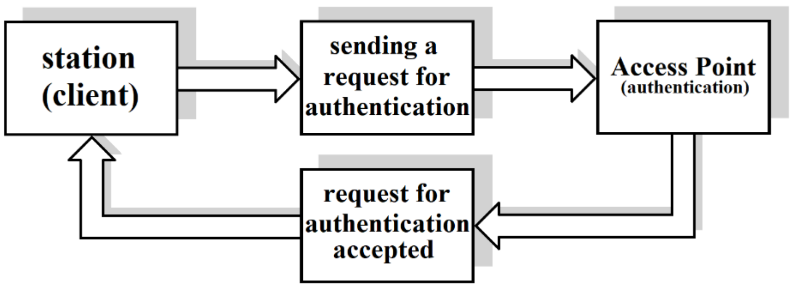

Fig. 4.4 Open system authentication

4.1.3 SD card

A secure digital card (SD card) is a small flash memory device designed to provide high-capacity memory storage in a portable device. SD cards are usually marked with a logo, as shown in Figure 4.5. The SD technology is managed by the SD Association (SDA). SD cards are manufactured in three different forms standard SD, miniSD, and microSD cards.

Fig. 4.5 Secure Digital logo [20]

Capacities are reaching from several gigabytes up to 2TB depending on four standards for capacity, which are listed in Table 4.1.

Table 4.1 Capacities of four capacity standards for SD cards [20]

	SD Standard	**SDHC Standard**	**SDXC Standard**	**SDUC Standard**
Capacity	up to 2 GB	more than 2 GB up to 32 GB	more than 32 GB up to 2 TB	more than 2 TB up to 128 TB
File System	FAT 12, 16	FAT 32	exFAT	exFAT

The initial SD bus speed of 12.5 MB/s is the Default Mode and was defined by SD1.0. Then a 25 MB/s High-Speed Mode was defined by SD1.1 to support digital cameras. As higher performance levels were needed to support new and faster devices, the SD Association introduced faster speed bus interfaces: UHS-I (50 MB/s), UHS-II (156 MB/s Full Duplex, 312 MB/s Half Duplex), UHS-III (312 MB/s Full Duplex, 624 MB/s Full Duplex), and SD Express (from 985 MB/s to 3940 MB/s). All these faster interfaces are available for SDHC, SDXC, and SDUC memory cards. [20]

4.2 Power supply

The power supply is an integral part of every electronic device. This chapter does not go into detail on the various topologies of linear, switching, or rechargeable power supplies, but rather a brief introduction to the power options of the multifunction speaker and which of them are advantageous for implementation directly into the multifunction speaker.

4.2.1 Rechargeable battery power supply

A rechargeable battery power supply is very popular because it allows the device to be portable, which is essential for a multifunction speaker. Lithium-ion or so-called Li-ion batteries are mostly used in today's consumer electronics due to their very good properties such as very high energy density, or low self-discharging (for example 18650). The nominal voltage of the Li-ion cell is 3.6 or 3.7 V. It then varies according to the series connection, where the voltages of the individual accumulators add up or according to the degree of charge or discharge. Therefore, the battery has a voltage in the range of 3.2 to 4.2 V, below or above these values there is a risk of either undercharging, where the battery may fail to recharge to its nominal value or at all, or overcharging, which may cause the explosion or ignition and injury to the user. To ensure that this does not happen, it is necessary to control and monitor the entire charging and discharging process, therefore the Battery Management System (BMS) is used.

The BMS is an integral part of any battery-powered device. The BMS is intended to keep the batteries within an acceptable voltage range (so in the charging process when the batteries reach the upper threshold of 4.2 V, turn off charging, and in discharging process when the batteries reach the bottom threshold of 3.2 V, cut them off from the load), when using multiple accumulators, balancing the voltage between them, and thus increasing their life and preventing the dangerous situations mentioned above.

4.2.2 Switching-mode power supply

Switching power supplies are used in all types of consumer electronics that use DC (Direct Current) voltage. The main purpose of such sources is to convert AC (Alternating Current) voltage from the mains outlet (230 V in Europe) to DC voltage, most often 3.3, 5, or 12 V (or any other). Thanks to high-frequency switching and low energy loss in the internal components, the transmitted power is higher compared to the classic variant with a transformer and rectifier at the same device size, so it is lighter, smaller, and has high efficiency. However, due to this switching, there is considerable interference in the source, which must be well suppressed, and thus the structure of the entire switched source is much more complex than the transformer and rectifier solution.

In the solution of this work, the switching power supply is used mainly as a charger for Li-ion 18650 batteries.

4.3 Multifunction speaker enclosure

There are several types of speaker enclosures and multifunction speaker enclosures. In this work, the enclosure is made of chipboard and adapted to be convenient for both carrying and handling, as well as for controlling various speaker functions.

5. DESIGN OF SUITABLE HARDWARE

This part of the work gets to how to assemble a portable multifunctional speaker defined by us by summarizing everything this project may need and finding a solution to build suitable hardware that is powerful enough for data processing, but also ideal for portable use to get the speaker in active playback mode as long as possible. Furthermore, a suitable amplification method for high-quality audio output is again concerning battery operation, i.e., an amplifier with considerable power but very high efficiency. The project is therefore looking for a solution that saves energy for a longer listening time but at the same time a quality music listening experience.

5.1 Properties

We require 3 functions, namely Bluetooth, reading from an SD card, and radio via Wi-Fi connection. This also comes with suitable control peripherals such as adjusting the volume, switching tracks, retuning radio stations, etc. All this needs to be controlled somehow.

When designing a project with the ability to process audio signals or audio data as the multifunction speaker we typically consider a subset of the following components, such as inputs like storage media, e.g. microSD card with audio files to read, a Wi-Fi interface to obtain an audio data stream from the internet, Bluetooth interface to obtain an audio data stream from a Bluetooth headset, user interface e.g. buttons or some other means to provide user input, and the outputs, such as analog signal output for power amplifier input or speakers, Bluetooth interface to stream audio data to e.g. a Bluetooth headset (for example, playing or pausing music, etc..), I2S interface to stream some data to a codec chip, and the main processing unit, that is a microcontroller with processing power to read the data from the input, process (e.g. encode/decode) and send to the output.

On this basis, the ESP32, which is a low-power system on a chip microcontroller with integrated Wi-Fi and dual-mode Bluetooth, was chosen for this work, which ideally meets our requirements for Bluetooth, Wi-Fi, and processing power.

An alternative to ESP32 would be, for example, an 8-bit AVR microcontroller unit such as ATmega328p or ATmega2560, but these microcontrollers would need to be supplied with external modules for Wi-Fi and Bluetooth, which are already directly implemented in ESP32, and they have also much smaller processing power. But that's not all, the ESP32 chip also needs other external components for music processing, so an audio codec chip, SD card input, and more.

A class D amplifier was selected for the appropriate amplification method, namely an amplifier from Texas Instruments the TPA3106D1. This amplifier has a performance exceeding 90 % and a sufficiently high power, which makes it an ideal choice for amplifying the audio output from the codec chip. A suitable alternative to this amplifier would be other similar types with the highest possible efficiency and quality of the output signal.

The main power supply is the battery pack consisting of a 18650 Li-ion battery, which supplies all the individual hardware components. The pack consists of five 18650 Li-ion batteries with a capacity of approx. 2000 to 3500 mAh and are connected in series to create a supply voltage ranging from 16 V (lowest battery pack voltage) to 21 V (highest battery pack voltage), hence the 5S battery pack. The power output directly from this pack is connected to the power pins of the TPA3106D1 power amplifier and via the BD9E104FJ which is a synchronous buck DC/DC converter with built-in low on-resistance power MOSFETs that is supply the power branch for the ESP32 microcontroller.

5.2 Block diagram

Let's define these parameters that we stated before with the block diagram that is in Figure 5.1 for a better idea of the design of the portable multifunction speaker itself.

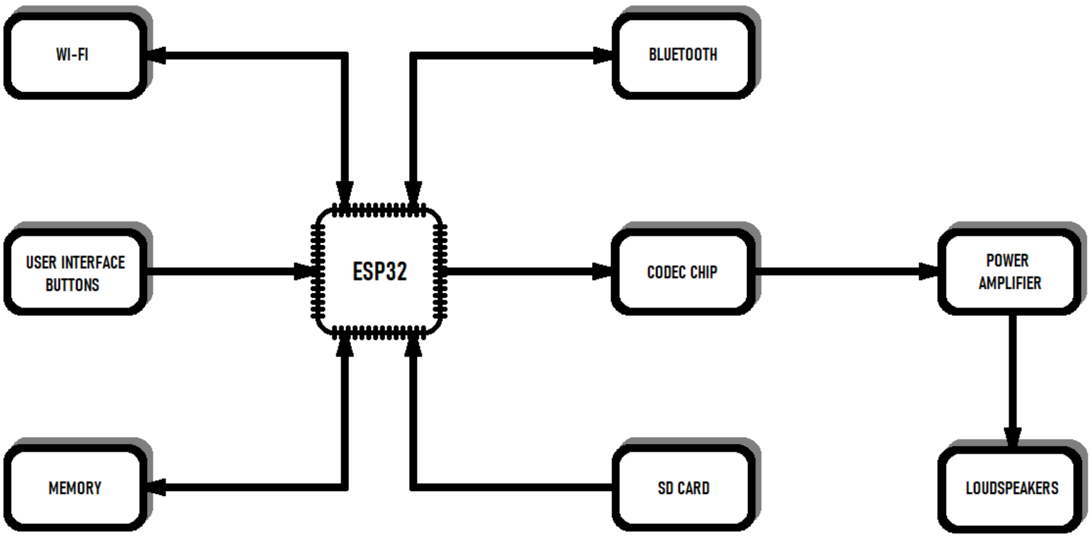

Fig. 5.1 Block diagram of a multifunction speaker

5.3 ESP32-LyraT-Mini

Based on the parameters determined in the first part of this chapter, a development board from Espressif was selected, namely ESP32-LyraT-Mini, which perfectly meets all requirements. It has all the peripherals like control buttons, Wi-Fi, Bluetooth, SD card slot, and other features.

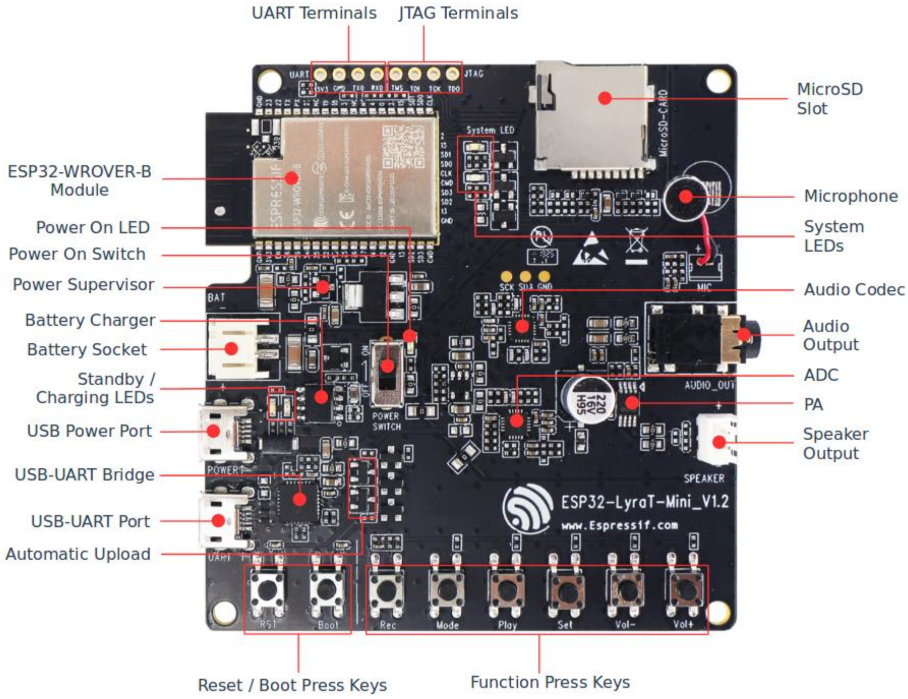

Fig. 5.2 ESP32-LyraT-Mini [21]

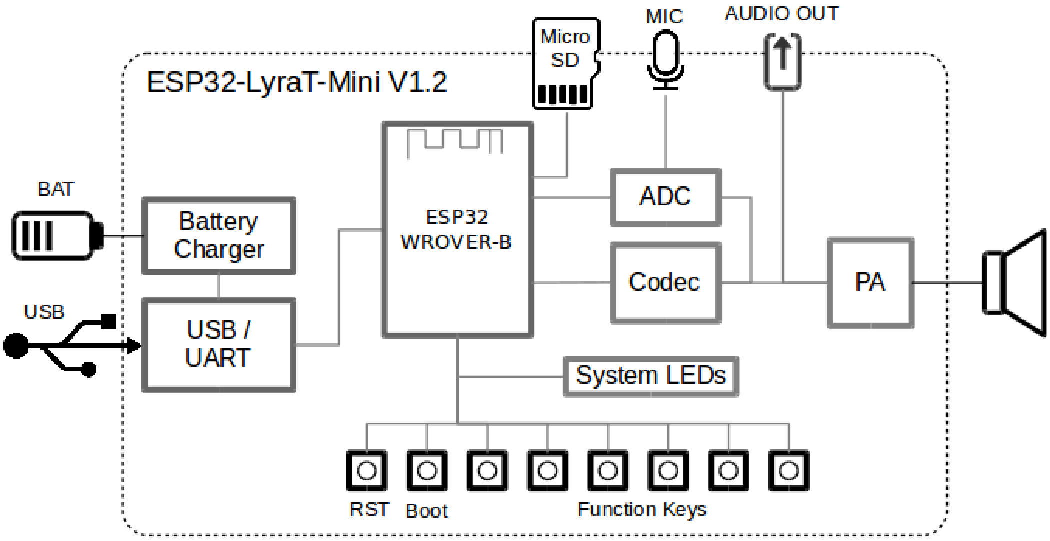

Fig. 5.3 Block diagram of ESP32-LyraT-Mini [21]

As can be seen from Figure 5.3, the Espressif ESP32-LyraT-Mini development board has all the parameters and components a project needs. The microcontroller ESP32, in this case, ESP32-WROVER-B module which contains an ESP32 chip to provide Wi-Fi and Bluetooth connectivity and data processing power as well as integrates 32 Mbit SPI flash and 64 Mbit PSRAM for flexible data storage [21] is implemented on the development board and together with the UART interface it is very easy to program and communicate with the chip.

Another necessary part is the codec chip ES8311 which is a low-power mono audio codec. It consists of 1-channel ADC, 1-channel DAC, low noise pre-amplifier, headphone driver, digital sound effects, analog mixing, and gain functions. It is interfaced with ESP32-WROVER-B over I2S and I2C buses to provide audio processing in hardware independently of the audio application. [21]

And finally, the desired peripherals, such as the user interface in the form of buttons (play, pause, mode, next, preview, set) that can be easily desoldered from the board and moved to a convenient location and a microSD card slot for reading data from the microSD card memory.

The ESP32-LyraT-Mini development board also contains other features and functions (battery charger, microphone, etc.) that are not used in this project or might be implemented in the future if necessary.

5.4 Class D power amplifier TPA3106D1

An integral part of this project is a power amplifier that is strong enough to sound large room. Since the speaker is portable and thus powered by batteries, it is necessary to use an amplifier with maximum possible efficiency. The low-frequency amplifier which has efficiency theoretically up to 100 % is a Class D amplifier, see Chapter 3. Therefore, the Class D amplifier was chosen for this project.

5.4.1 Description

The TPA3106D1 is a 40 W efficient, Class-D audio power amplifier for driving bridged-tied stereo. The TPA3106D1 can drive stereo speakers as low as 4 Ω. The high efficiency, of about 92 % the TPA3106D1 eliminates the need for an external heat sink when playing music. The gain of the amplifier is controlled by two gain select pins (GAIN0, GAIN1). The gain selections are 20, 26, 32, and 36 dB. The outputs are fully protected against shorts to GND, VCC, and output-to-output shorts with an auto-recovery feature and monitor output. [22]

5.4.2 Functional block diagram

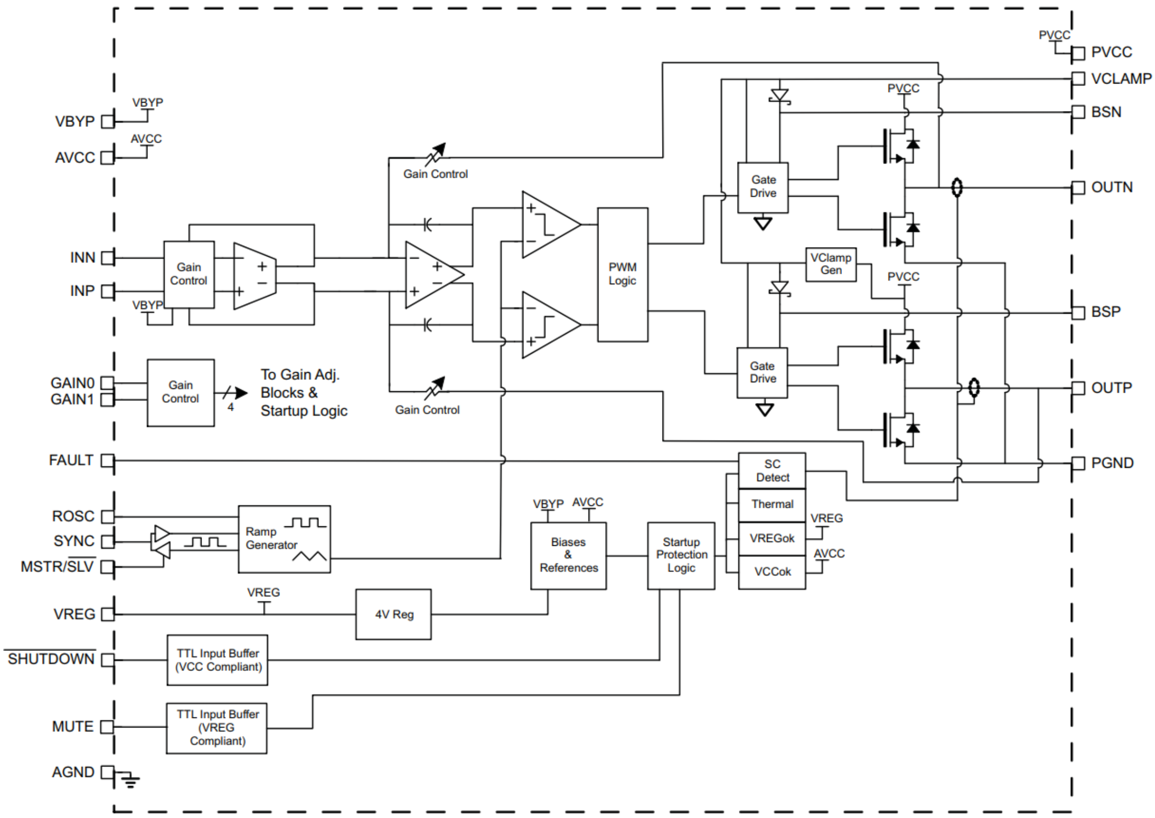

Fig. 5.4 Block diagram of TPA3106D1 [22]

On the left side of Figure 5.4 are the input pins of the TPA3106D1 power amplifier, which are then connected to the gain control. The gain of this amplifier is selected using the TTL logic levels with compliance to VREG on pins GAIN0 and GAIN1 according to the table in the datasheet of the TPA3106D1. In this project, the gain is fixed at 26 dB and thus with an input impedance of 16 kΩ ±20 % (see Gain Setting via GAIN0 and GAIN1 Inputs in [22]). The audio output from the ESP32-LyraT-Mini development board is connected via the input filter to the INN and INP pins of the amplifier. The values of the passive components of this filter were calculated according to equation 5.1 with a cut-off frequency of 10 Hz.

$$C_i = \frac{1}{2\pi Z_i f_c} = \frac{1}{2\pi \cdot 16 \cdot 10^3 \cdot 10} \cong 1\ \mu F, \tag{5.1}$$

Another interesting input is also the SHUTDOWN pin which is active low. Pulling SHUTDOWN low causes, the outputs to mute and the amplifier to enter a low-current state, thus reducing supply current to the absolute minimum level during periods of nonuse for power conservation. This operation is very advantageous when the amplifier is not used (i.e., when there is nothing to play) since the entire multifunction speaker is supplied by a battery pack, and the playtime is improved many times over.

The detecting function of the audio signal at the amplifier input does the NJU7181 which is a signal level sensor system IC. It sends a high logic level to the amplifier whenever it detects the existence of the audio signal and by that deactivates the SHUTDOWN mode of the amplifier. The NJU7181 includes a delay circuit that allows the amplifier to continue to hold the logic level after the absence of the audio signal. The 15 seconds delay time was selected.

The value of the resistor connected to the ROSC terminal that controls the class-D output switching frequency was calculated from equation 5.2 with a selected switching frequency of 250 kHz with consideration of a recommendation from the datasheet of TPA3106D1.

$$R_{OSC} = \frac{1}{2 f_{OSC} C_{OSC}} = \frac{1}{2 \cdot 250 \cdot 10^{3} \cdot 20 \cdot 10^{-12}} = 100\ k\Omega, \qquad (5.2)$$

All other passive components, including their parameters, were selected considering the recommendations from the TPA3106D1 datasheet.

5.4.3 Graphs with characteristics

This part shows some of the interesting graph characteristics that are expected from the amplifier.

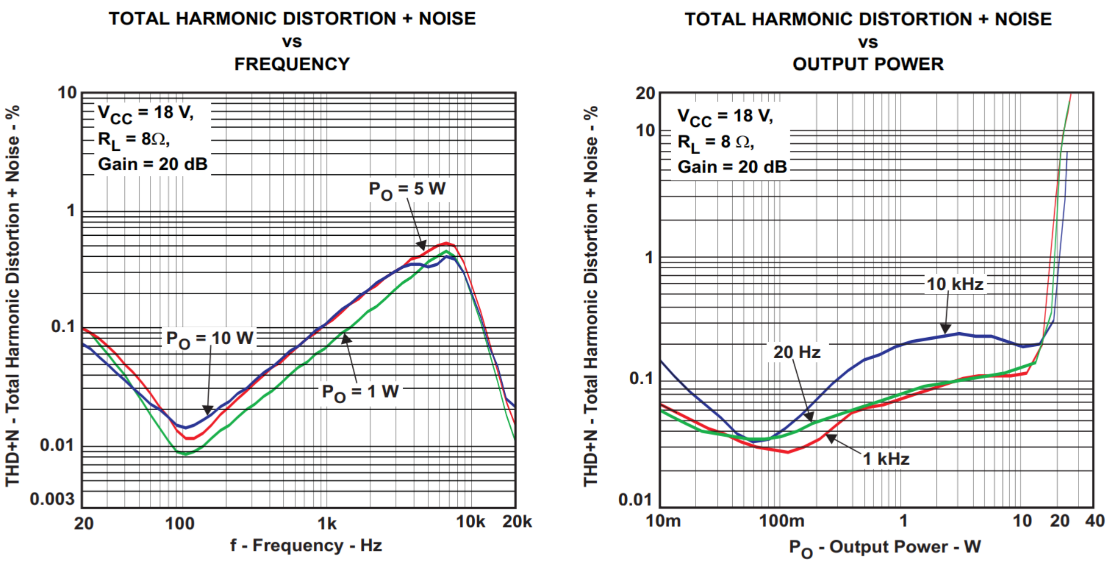

Fig. 5.5 Dependence of THD + N on frequency and output power [22]

From Figure 5.5 can be seen in the first graph that at low frequencies THD+N increases because a larger noise component predominates, at about 100 Hz it reaches a minimum then the THD+N increases, in this case, due to THD.

From the second graph in Figure 5.5, a THD+N varies in the range of about 0.1 % at output powers from 0.1 to approx. 15 W which is very good output signal quality. However, from 15 W up, THD+N increases exponentially, which is not horrible considering that the speaker is usually at a medium power level and not at maximum.

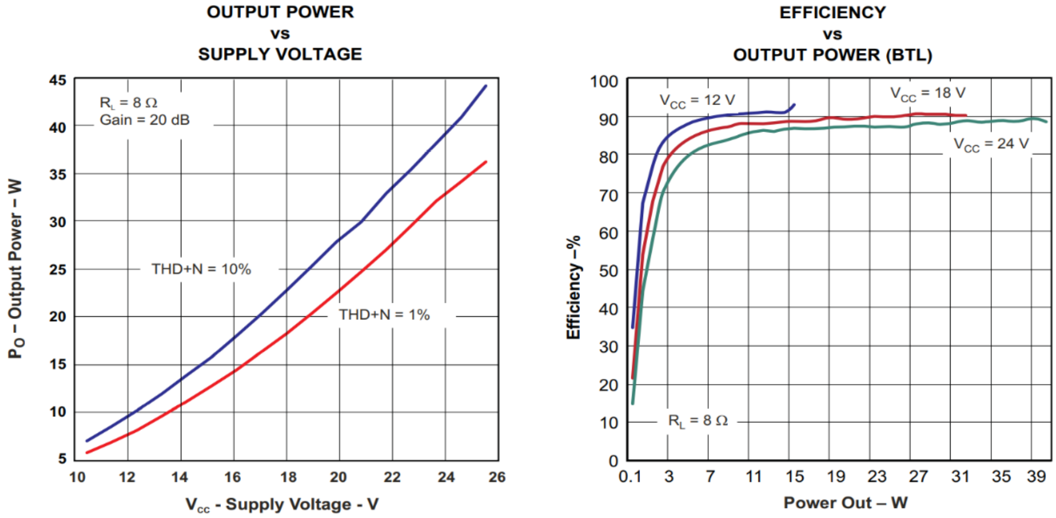

Fig. 5.6 Dependence of output power on supply voltage and efficiency on output power [22]

The voltage of the battery pack changes depending on the charge level from 16 to 21 V. As can be seen in Figure 5.6, the first graph shows that this change in voltage will also affect the output power, so with gradually discharging batteries, the output power will gradually decrease. The second graph shows a course at a supply voltage of 18 V, which is about the average value of the battery pack voltage. The efficiency is around 90 % at higher output powers from 3 W, where the multifunction speaker operates most of the time.

6. SOFTWARE DESIGN

Finally, the last part of this work analyzes the structure of the operating software that controls the entire operation of the multifunction speaker. Earlier in this work, a board from Espressif was chosen, namely ESP32-LyraT-Mini with an ESP32 microcontroller module with sufficient data processing power to work with data from Bluetooth, Wi-Fi, SD card, and more. This chapter aims to design an approximate form of the whole software and its function using brief diagrams.

6.1 State and flowchart diagrams

To better understand the whole system that makes up the software, it is better to illustrate these functions in diagrams. The multifunction speaker has three main playback modes that communicate with different peripherals.

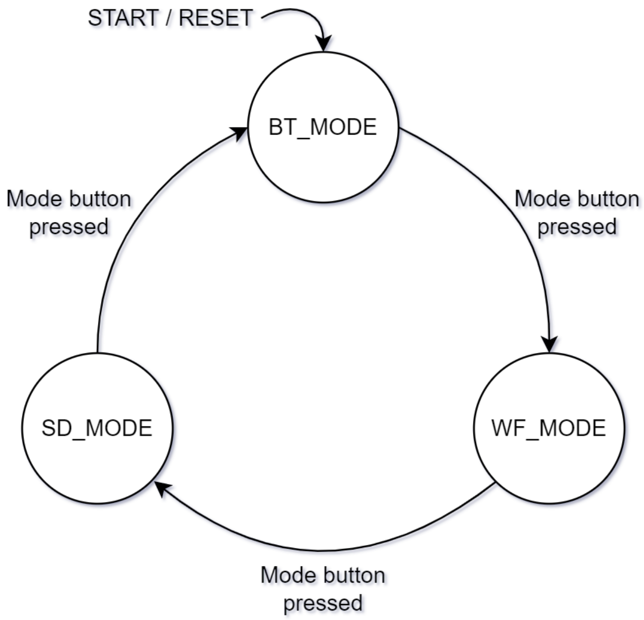

Fig. 6.1 State diagram of the interaction between individual modes

As can be seen in Figure 6.1, one of these modes is Bluetooth reception of audio signals (BT_MODE) from a phone or computer to a multifunction speaker, the other is Wi-Fi mode (WF_MODE) which plays MP3 downloaded from HTTP, where MP3 is an online internet radio station, and the last SD card mode (SD_MODE), where MP3 files stored on the SD card are played.

Of course, each of these modes must be user-controlled in some way. The buttons that are directly located on the ESP32-LyraT-Mini development board are used to control interaction in these modes, namely the "Play" button for playing and pausing the currently playing media, the "Set" button usually has the function of advancing to the next song or preview song (in Wi-Fi mode to switch online radio stations), the "Vol+"

button to increase and "Vol-" button to decrease the volume, and the "Mode" button to switch between the modes, as seen in flowchart diagram in Figure 6.2.

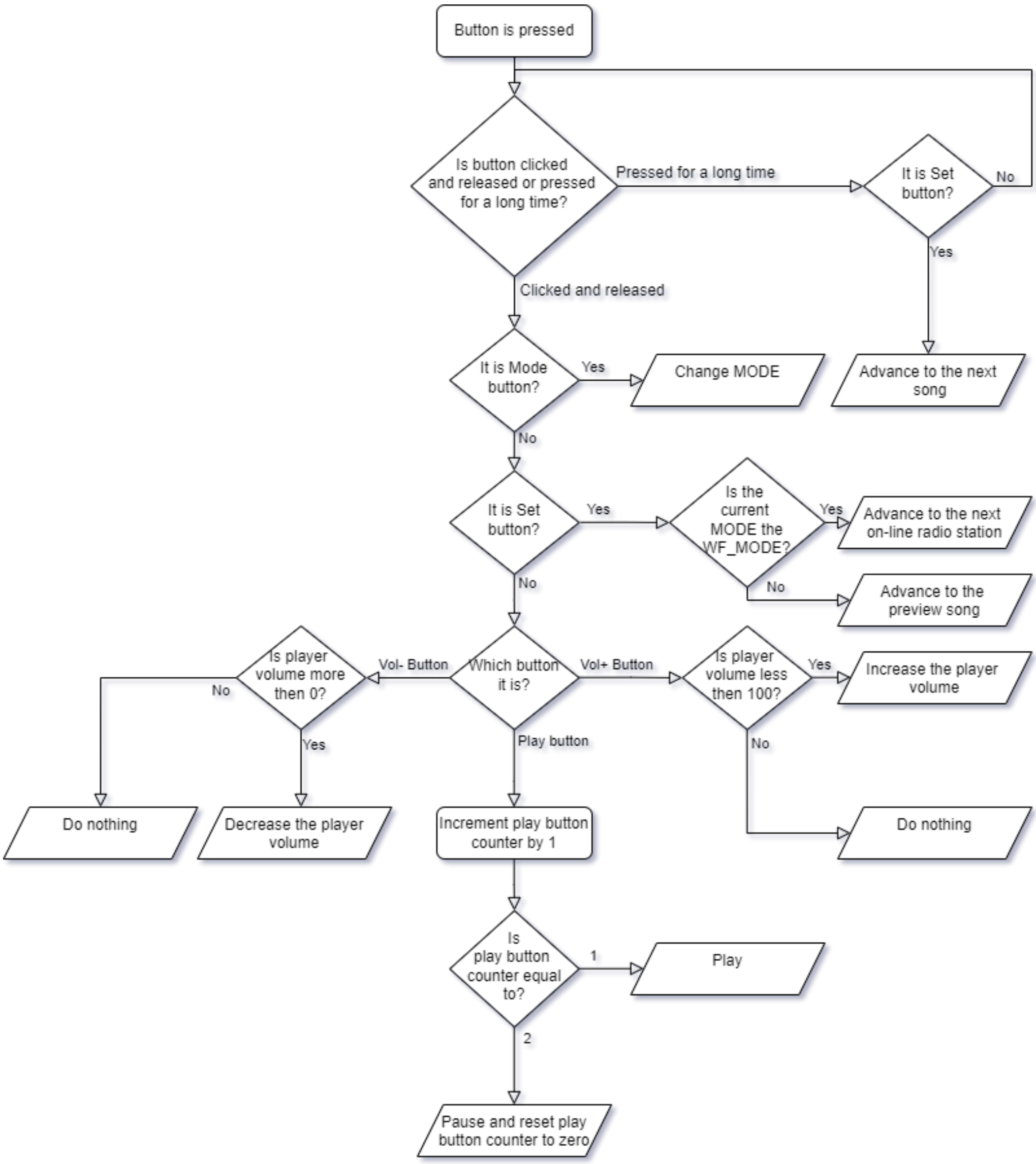

Fig. 6.2 Flowchart diagram of buttons interaction

6.2 Espressif Audio Development Framework

Inventing and writing the whole software from scratch would be very demanding and difficult to accomplish in the scope of this work. Therefore, already written verified libraries are used directly from the manufacturer of the development board, namely Espressif Audio Development Framework. This API provides a way to develop audio applications using Elements like Codecs, Streams, or Audio Processing functions.

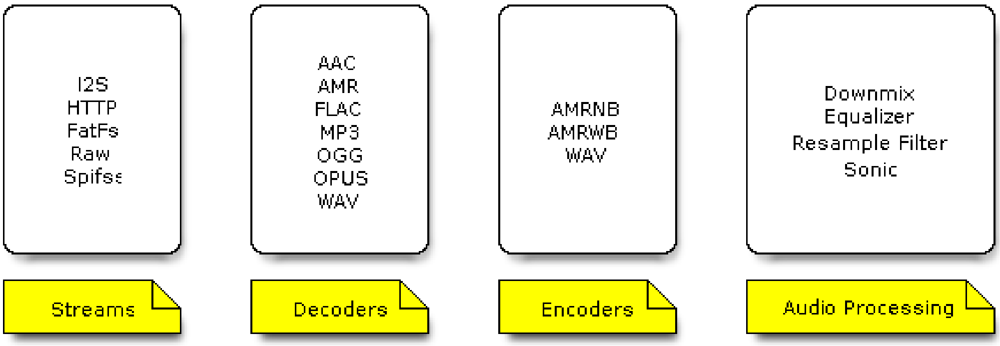

Fig. 6.3 Elements of the Audio Development Framework [23]

As can be seen from Figure 6.3, many elements are perfect for controlling and processing the audio signal in this project. Streams are an Audio Element responsible for acquiring audio data and then sending the data out after processing, from this audio element project uses I2S, HTTP, and FatFs streams. From Codecs, for example, MP3 Decoder to decode an audio data stream provided in MP3 format, and from Audio Processing, Equalizer to set the gains of individual bands for balanced sound at the output.

6.3 Audio Pipeline

The resulting application is developed by combining the elements into an Audio Pipeline. Figure 6.4 presents the example of the organization of three elements in the Audio Pipeline, that is used in BT_MODE.

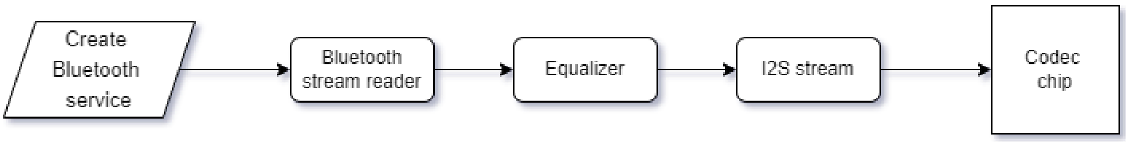

Fig. 6.4 Organization of Elements in Audio Pipeline

7. OPTIMALIZATION AND TESTING

In the previous chapters, the design was considered both in terms of hardware and software. In this chapter, as the name suggests, these design solutions are implemented, tested, and optimized as well as the design of the appearance solution for the best possible result of the multifunction speaker.

7.1 Software

So first comes the software. Thanks to the selection of the development board from Chapter 5., it was very easy to start directly with the implementation of the software part of the project according to Chapter 6.

The goal, according to Chapter 6. and Figure 6.1, is to create an FSM (Final State Machine), which is based on the user's decisions to move between the various modes such as SD card mode, Bluetooth mode, and Wi-Fi radio mode. Each mode is composed of the so-called Audio Pipeline which consists of elements from the Audio Development Framework (see Figure 6.3). For the control of the basic playback parameters (play/pause, volume settings, switching between tracks, or directly switching between modes) was designed the procedure shown in Figure 6.2 as a user interface in the form of buttons.

However, the problems were appearing from the beginning and the whole system of this part of the work had to be repaired and optimized. The structure of the software given by the idea from Figure 6.1 is below in a simplified source code.

```
/* Main function */
void app_main(void)
{
// Initialize peripherals management …
// Start audio codec chip …
// Setup the volume …

// Forever loop
     while (1) {
          switch (mode) {
                case SD_MODE:   { …
                case BT_MODE:   { …
                case WIFI_MODE: { …
                default:        { …
          }
     }
}
```

7.1.1 Bluetooth/Wi-Fi coexistence (IRAM overflow)

The first major problem was with the IRAM overflow of the ESP32 microcontroller. The app build was failing with the linker errors such as **section `.iram0.text' will not fit in region `iram0_0_seg'**, **IRAM0 segment data does not fit** and **region `iram0_0_seg' overflowed by 84 bytes**. There are several ways to troubleshoot this issue in the microcontroller manufacturer's documentation (where this issue is reported) [24]. However, none of these steps helped and the resulting solution was ultimately the reinstallation of the entire software development environment for the hardware based on the ESP32 chip by Espressif. After this step and the correct settings of the ESP32 menu config, the problem no longer occurred.

7.1.2 Crashing without the mounted SD card

Another major problem was the inability to run the application without a mounted SD card, which is a non-accessible and inadmissible issue for the end-user. The un-operation and subsequent crash of the application were caused by the initialization and start of the SD card peripheral and the setting up of the SD card playlist and the scanning of the music, which must be done to ensure the reading of data from the SD card. If this step is performed without a mounted SD card, the application will crash and restart immediately, ending in an endless loop. Fortunately, the solution was simply using the ESP32 pin IO34 which is connected to the microSD Card module, as shown in Figure 7.1.

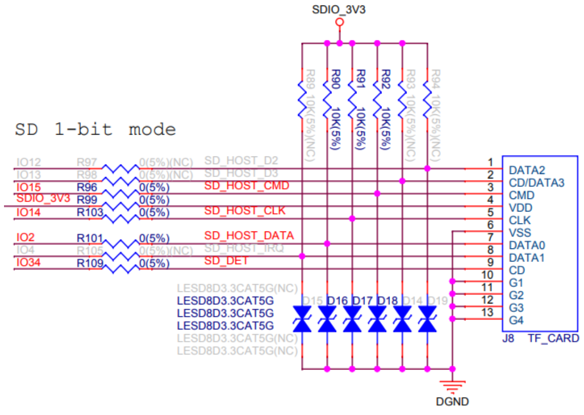

Fig. 7.1 Connections of the SD card mounter [25]

Via this pin, it was possible to detect whether is or is not an SD card mounted. Therefore, every time the application is started, the detection of the SD card is first ensured, and then, based on the result, it is evaluated whether the peripherals of the SD card are initialized or not. If so, it will jump to SD card mode and if not, it will skip this mode and jump to Bluetooth mode right away. Due to this solution, the designed state diagram from Figure 6.1, had to be adapted to the new one, which is shown in Figure 7.2.

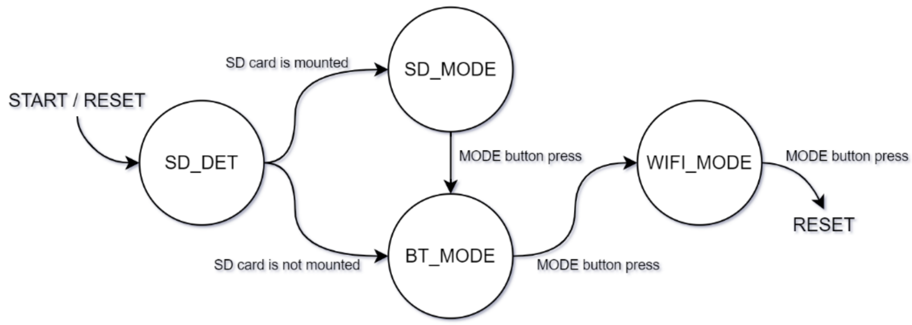

Fig. 7.2 Updated state diagram

7.1.3 Bluetooth remote control

The last bigger problem is the music volume control via the AVRC profile of Bluetooth technology. With most consumer devices similar to a multifunction speaker, it is always possible to control basic functions such as play/pause, volume up and down, advancing to the next or previous song, etc. on both devices. Of course, this is also the goal here. When it comes to functions like play/pause and advancing to the next or previous song, there is no problem. However, the problem is with adjusting the volume. The main disadvantage is that both devices have separate audio volume settings. Due to multiple modes in the multifunction speaker, volume control is created separately here, and in the case of Bluetooth mode, this control is not paired with the connected device, so the user has two separate volume controls instead of one. The problem is that if these volumes are paired, there is a high-volume overdrive on the LyraT-Mini development board itself from approx. 80 % of the total volume of the audio signal, which after subsequent amplification by the power amplifier causes unbearable listening. Therefore, the volume in the application is limited from 44 % to 80 %, where the volume is ideal for listening. Unfortunately, this is quite a bug in the Espressif Framework, which has this suboptimally solved volume level where at less than 40 % the music is almost inaudible and at more than 80 % is completely distorted. In short, it is best for user to set one of the devices fixed and control the functions on only one. Unfortunately, the solution to this problem did not fit into this work, but it will not stay that way, and this work will continue to look for a solution to the best result on this problem.

7.1.4 Other problems

In retrospect, there were many minor problems in this part of the design that were not interesting enough to be described in more detail as the mentioned ones above. Therefore, they are not discussed furthermore. These problems included mainly issues with memory such as flash, PSRAM, and IRAM, setting the microcontroller menu configuration for the correct operation, and, of course, general issues with the completion of the entire application.

7.1.5 Final version

The issues have been described and now it is time for a final proposal of the software part. The overall design had to be modified a bit to suit all requirements and perform all functions correctly. Thus, the application follows the state diagram in Figure 7.2. For a better idea, a simplified version of the Flowchart diagram from Figure 6.2 is shown in Figure 7.3, according to which the operation of the control buttons is controlled.

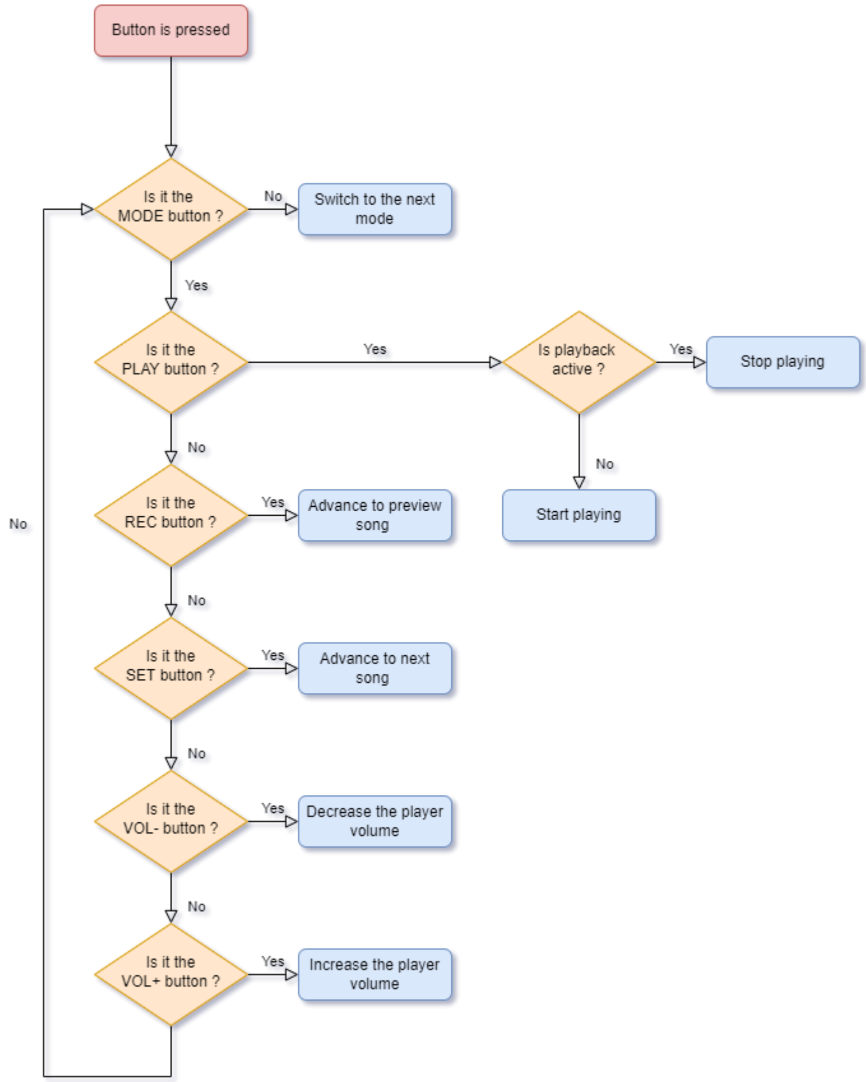

Fig. 7.3 Simplified flowchart diagram

As discussed in Subchapter 6.3., each mode consists of an audio pipeline. This audio pipeline is constructed of the elements shown in Figure 6.3. To better understand the operation of the modes, the block structure of the audio pipelines is shown in Figure 7.4.

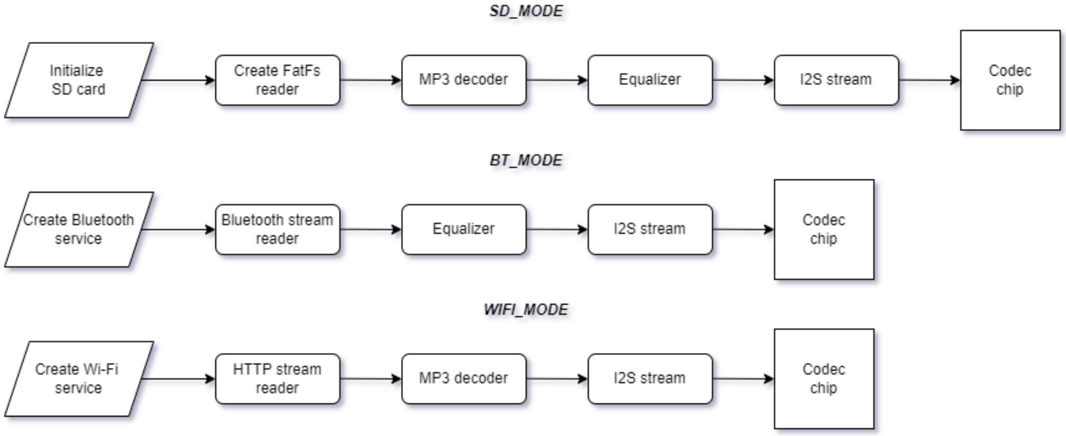

Fig. 7.4 Block diagrams of the Audio Pipelines

Each of the modes is announced by a system prompt tone files, which are stored in the ESP32 flash memory. The audio announcements were created via a text-to-speech API from Google Cloud, and it was necessary to convert these announcements from MP3 to binary using a python script provided by Espressif and then upload them to flash memory. The modes are thus easily distinguishable, and the user is informed in advance of which mode he is currently located. For example, before entering SD mode, "SD card mode" is reported, for BT mode "Bluetooth mode" and finally for WIFI mode, "Wi-Fi radio mode". Again, it is needed to create the Audio Pipeline shown in Figure 7.5, for these audio announcements, which takes care of this function.

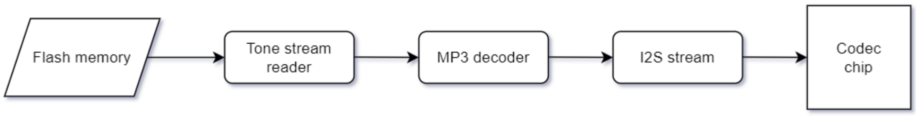

Fig. 7.5 Block diagrams of the Audio Pipelines

The application also includes SHUTDOWN control of the power amplifier as a replacement for the analog variant from Chapter 5., using pin IO22, which normally serves as a control for the system GREEN LED, as shown in Figure 7.6.

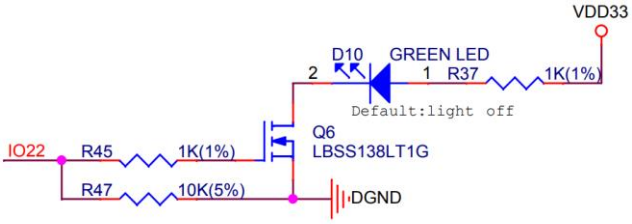

Fig. 7.6 Connection of the system LED [25]

The ESP32 microcontroller on the LyraT-Mini development board already has the vast majority of pins used. Due to the non-use of this system LED in this project, the IO22 pin is ideal for SHUTDOWN control of the power amplifier. However, more about this topic is in Subchapter 7.2.

7.2 Hardware

This section discusses and optimizes the proposal in Chapter 5. and everything related to hardware. According to the block diagram in Figure 5.1., essentially most of it is mounted directly on the ESP32-LyraT-Mini development board. The other external parts are just the power amplifier and the speakers. Individual solutions, both circuit and functional, of these external parts were proposed, together with an audio detector for controlling the SHUTDOWN pin of the power amplifier and buck converter for powering the development board.

During the creation and testing of the design mentioned in Chapter 5., again, as was the case with the software implementation, there were problems with the function of individual blocks and non-compliance with the requirements that were given for this project. Therefore, these solutions had to undergo adjustments, which this chapter deals with.

This chapter not only modifies the mentioned blocks of the multifunction speaker but also adds a completely new circuitry that improves the operation of the entire multifunction speaker and the user's enjoyment of use.

7.2.1 First PCB design, problems, and measurements

The first hardware design included a power amplifier from Texas Instruments TPA3106, a synchronous buck DC/DC converter BD9E104FJ as a power supply for the ESP32 microcontroller, and an audio detector IC NJU7181. The final design of the PCB (Printed Circuit Board) and the actual implementation of this configuration are shown in Figure 7.7.

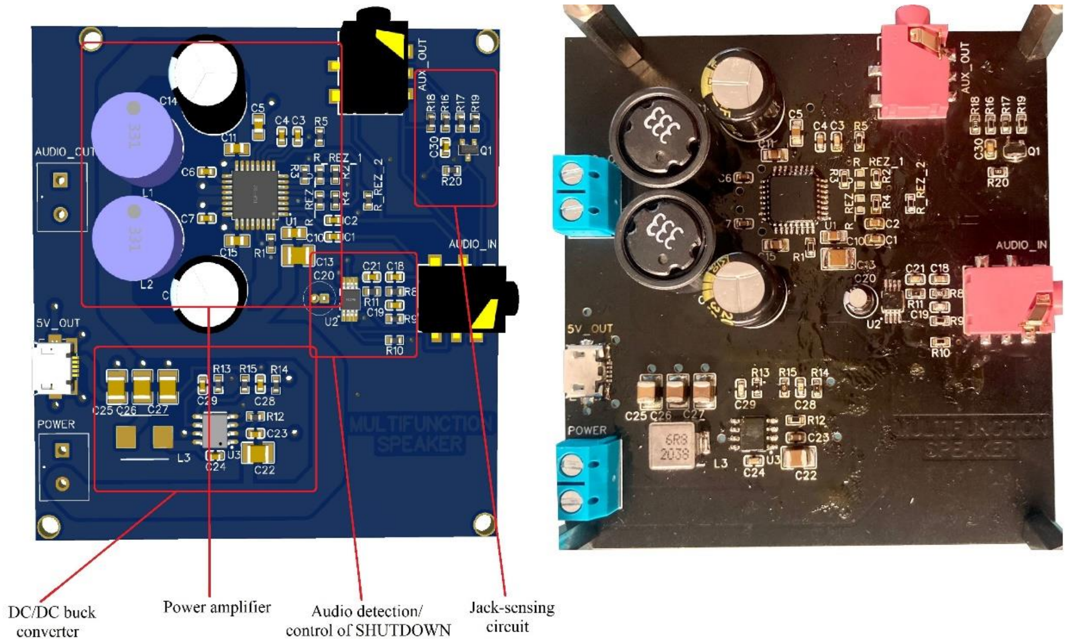

Fig. 7.7 First PCB design

A Jack-Sensing circuit has been added to detect the insertion of a 3.5 mm jack. This output serves as a headphone output from the speaker and thanks to this detection it can send a signal to the external trigger pin of the audio detector about whether the SHUTDOWN of the power amplifier should be activated. Therefore, when the headphones are plugged into the multifunction speaker, the amplifier is put into shutdown mode after a certain time delay set by the resistor and capacitor at the audio detector.

The first tests of this PCB design turned out very badly, mainly due to an unknown failure and destruction of the PA (Power Amplifier) during the first measurement (most likely due to the shorting of the chip legs during the measurement). Unfortunately, this was the only piece of this amplifier and at that time a completely scarce commodity (sold out everywhere). Therefore, it was very difficult to find this chip, but fortunately, there was still stock in one online store, which is not an authorized seller of these chips, and it may happen that the resulting chips are not of the original quality. However, at a time when it is not possible to assemble the hardware part of the solution without the main PA, there was no choice but to order it from there. Subsequent mounting of the same board and measurement was successful.

The measurement was performed without other functional blocks such as a buck converter, audio detector, and jack-sensing circuit connected as is shown in Figure 7.8.

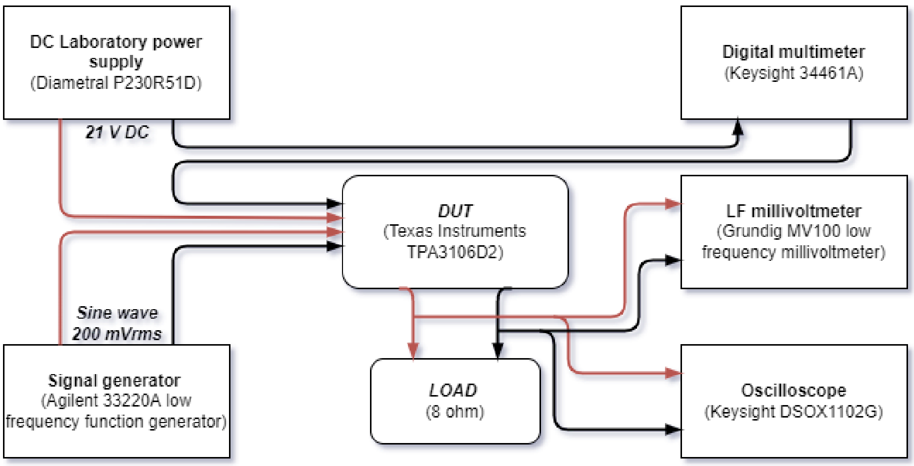

Fig. 7.8 Block diagram and connection of measurement conditions

With the set-up that was connected as in Figure 7.8, the basic parameters of an amplifier were measured.

The first measurement was to measure the maximum output power for limitation (all parameters are measured in RMS values). The maximum power, where the waveform is still without visible distortion (at $THD + N = 2.25\ \%$, $V_{IN} = 21\ V$, and amplifier gain set at 20 dB) was around 14 W and with the value of $THD + N = 10\ \%$ that manufacturers like to quote was power 16,6 W (in the datasheet of the PA is stated 35 W). This distortion value is already quite high and does not agree with the graph shown by the manufacturer in Figure 5.6, where under these conditions and THD+N 1 % the amplifier should achieve powers around 25 W. Or according to Figure 5.5, where according to the output power 14 W THD + N should move around 0.1 %. These facts and especially THD+N are very dependent on the output LC filter and potentially this filter was not ideally designed (although the design included the recommended components from the amplifier manufacturer). Therefore, the measurement results are so inaccurate. The second option may be the copy of the original chip with worse parameters than the original (purchased from an unauthorized store). Subsequently, efficiency measurements were performed, which is very important since the amplifier is powered via a battery pack. The efficiency of this amplifier is 86 % (at $V_{IN} = 21\,V$ and $P_O = 14\,W$). This efficiency according to the second graph in Figure 5.6, corresponds to the assumptions.

The last measurement was the frequency response of the TPA3106D2 amplifier, which is illustrated in Figure 7.9. Here the amplifier behaves as it should. At set constant 20 dB gain over the entire audible range and a decrease of 3 dB at a frequency of 20 kHz.

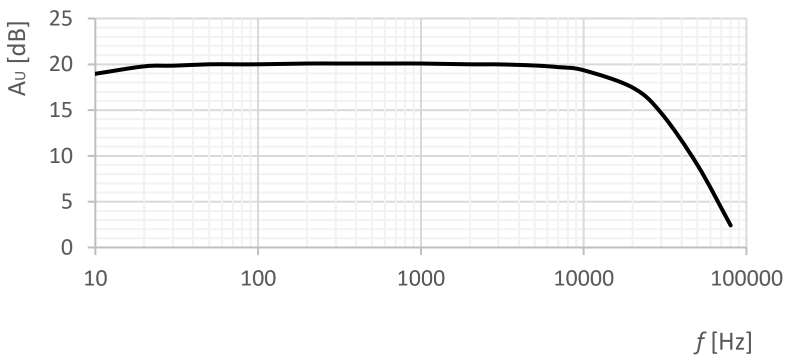

Fig. 7.9 Frequency response of the TPA3106D2

However, this solution had several disadvantages, such as high noise and interference with the connected buck converter (ground loop), large voltage ripple of the buck converter output, and the audio detector IC only detected signals higher than 500 mV_{RMS}, which is not a voltage level that would occur (up to exceptions) at the input of the amplifier where the detection is performed. Therefore, various solutions were proposed to correct these errors.

7.2.2 Second PCB design

The Second PCB design dealt mainly with solving the problem with the audio detector. The problem was that the audio detector did not have enough sensitivity, which was needed, mainly due to the small signals that occurred at the input of the amplifier. One of the solutions was to connect an output of the NA4150 PA located on the LyraT-Mini development board to the input of the audio detector. The signal there was already amplified to some level but in form of a PWM signal, so just a low pass filter was needed to filter it and pass it to the audio detector, as shown in Figure 7.10.

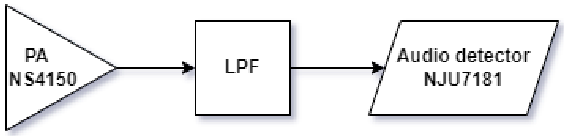

Fig. 7.10 First solution for the audio detection

Although this solution was the simplest, it had disadvantages. Again, small signals were below the threshold of the audio detector and the volume had to be adjusted directly in the program from a minimum of 44 % to 50 % to work it properly. This was already a significantly higher volume and not one that would sit listeners at the lowest volumes. Therefore, this solution has not been used.

Another solution was to add a preamplifier for the audio detector to be able to detect even the smallest audio signals. This step created a new PCB design, shown in Figure 7.11.

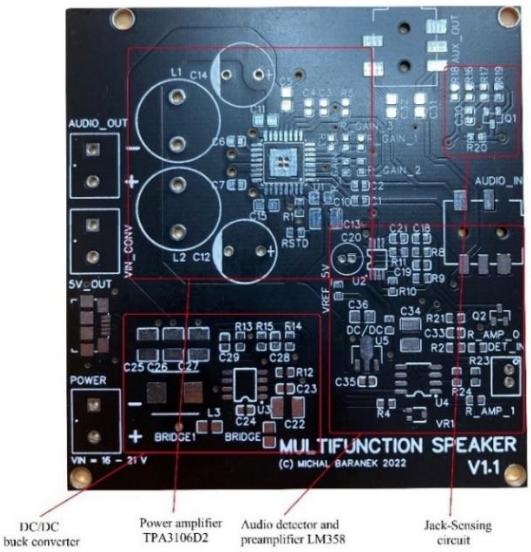

Fig. 7.11 Second PCB design

The LM358 preamplifier has two operational amplifiers in the SOIC-8 package. These two operational amplifiers have been connected into a cascade. Thus, the first amplifier fulfilled the function of amplifying small signals, and the second always excited the signal to a rectangular shape via a huge amplification. The limitation here was not a problem since the audio detector does not depend on the shape of the signal, but only on whether it is present at the input with a sufficient voltage level.

This solution was already able to work and perform its function. However, there was still the problem of high amplifier interference and noise in the combination of using a power amplifier and other function blocks such as a buck converter or audio detector.

In the end, this solution was not even used and assembled, and before this second form of PCB design was ready from the PCB manufacturer, a new design that was much more efficient and cheaper, was designed.

7.2.3 Final PCB design, advantages, and measurements

This layout was a big leap from the two previous designs, in such a way that almost everything changed and was designed differently. The first big change was the replacement of the TPA3106D2 power amplifier with the TPA3110D2. This power amplifier TPA3110D2 has the advantage over the amplifier in previous designs that it does not need filtering by LC filter and thus a reduction in price and comparable performance and parameters as TPA3106D2. The only filter at the amplifier output is a ferrite chip bead filter to suppress EMC interferences. The main motivation why the PA was replaced, was based on a sound test of the already installed first PCB design, where the TPA3106D2 was located, and subsequently a test with the help of an amplifier module, which was equipped with the TPA3110D2 chip. These tests were performed with all connected blocks such as buck converter, etc. and the difference was drastic. The first design with the TPA3106D2 had a lot of noise and interference, which was unbearable with the idle speaker mode, and unlike the TPA3110D2, the noise and interference were minimal. Unfortunately, in both cases, there was ground loop interference, which is discussed later in this chapter. For some references, here are a few graphs, shown in Figure 7.12 and Figure 7.13, with the parameters that can be expected from this amplifier.

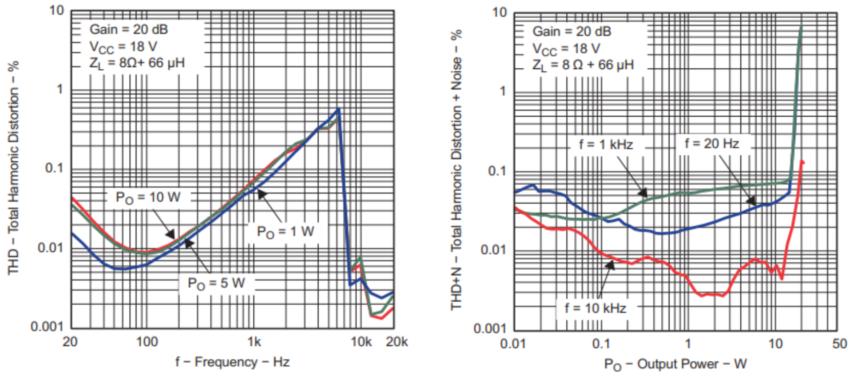

Fig. 7.12 Dependence of THD + N on frequency and output power [26]

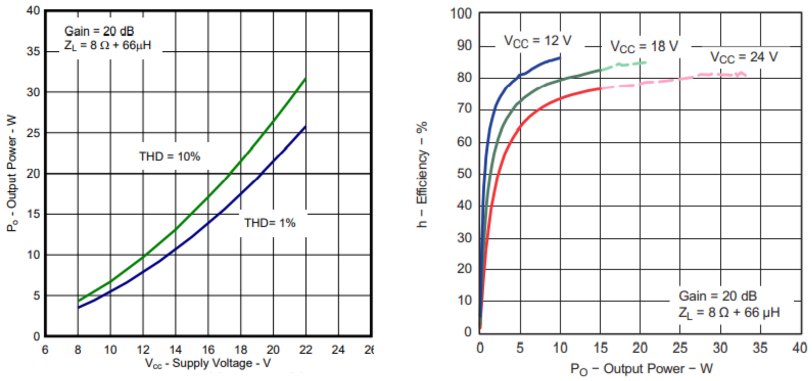

Fig. 7.13 Dependence of output power on supply voltage and efficiency on output power [26]

Thus, the amplifier was designed using a procedure similar to that described in Chapter 5, and the final design of the final PCB is shown in Figure 7.14.

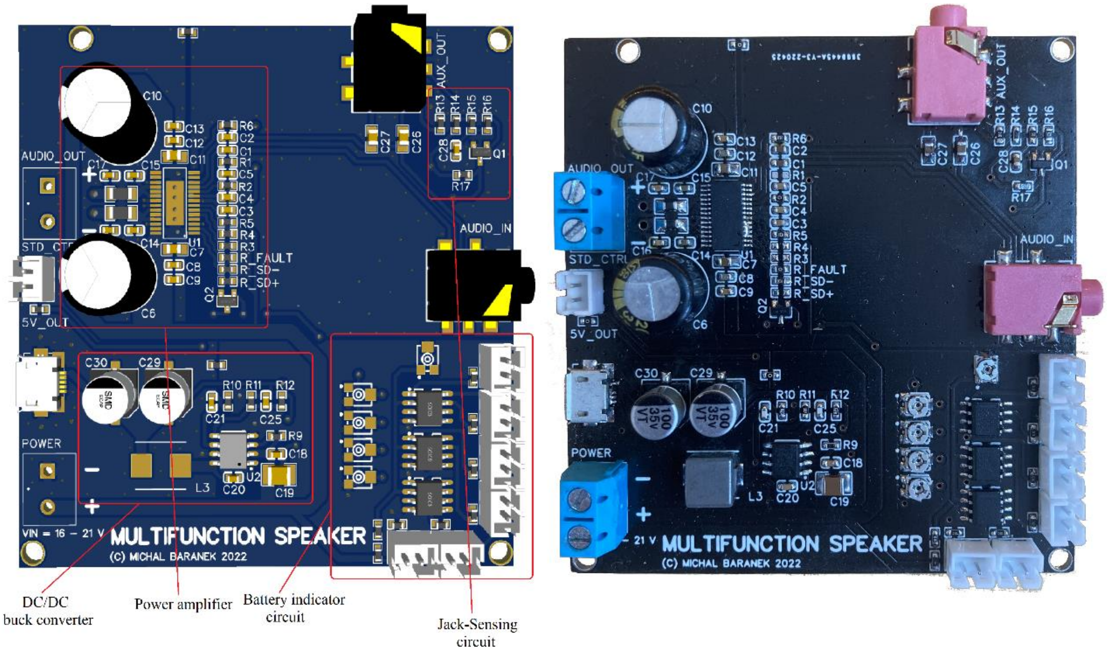

Fig. 7.14 Final version of PCB design

As can be seen from Figure 7.14, it is different from the two previous designs. The NJU7181 audio detector was removed, and the battery pack indication circuit was added. The whole structure of the PCB is clean and simple. Compared to previous models, it is much cheaper and more efficient.

Before explaining the new blocks in detail or why the old ones have been modified or completely removed, the measurement results of this TPA3110D2 amplifier are given for comparison with the graphs from the manufacturer in Figures 7.12 and 7.13 and the old TPA3106D2 amplifier parameters. The measurement was done by the following setup shown in Figure 7.15.

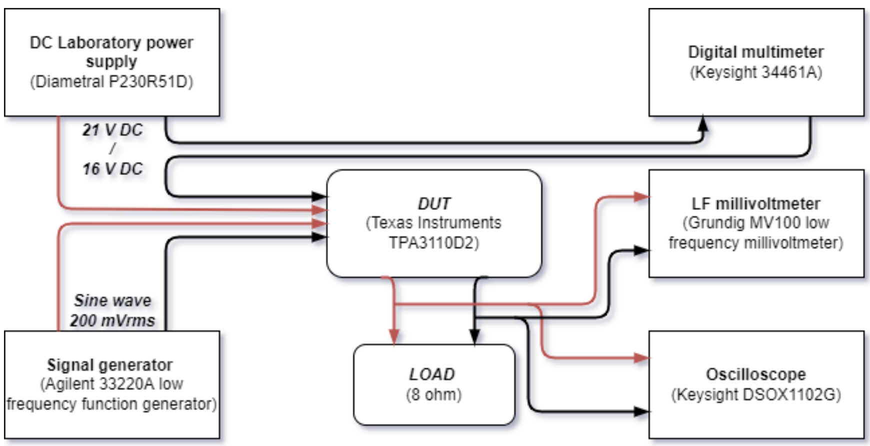

Fig. 7.15 Block diagram and connection of measurement conditions

The first measurement was to measure the maximum output power for limitation (all parameters are measured in RMS values). The maximum power, where the waveform is still without visible distortion (at $THD + N = 1$ %, $V_{IN} = 21\ V$, and amplifier gain set at 20 dB) was 11 W and with the value of $THD + N = 10$ % that manufacturers like to quote was power 16 W (in the datasheet of the PA is stated 30 W from $V_{IN} = 16\ V$). This distortion value is quite high and does not agree with the graph shown by the manufacturer in Figure 7.13, where under these conditions and THD+N 1 % the amplifier should achieve powers around 23 W. Or according to Figure 7.12, where for the output power 11 W THD + N should move lower than 0.1 %. These facts and especially THD+N are again as it was with the measurement of TPA3106 very dependent on the output LC filter, which this amplifier doesn't have, and measurement was done with the same connected LC filter as it was in the first measurement with the TPA3106. Potentially this filter was not ideally designed. Therefore, the measurement results are so inaccurate. The second option may be the copy of the original chip with worse parameters than the original (purchased from an unauthorized store, again as it was with TPA3106).

These measurements were all for a fully charged battery pack of 21 V. For comparison, the values of the maximum output power for limitation were measured at full discharge, or the lowest voltage when the battery management system turns off the power path to the amplifier. The maximum power, where the waveform is still without visible distortion (at $THD + N = 1.67$ %, $V_{IN} = 16\ V$, and amplifier gain set at 20 dB) was 5.6 W and with the value of $THD + N = 10$ % was power 8 W.

Subsequently, efficiency measurements were performed. The efficiency of this amplifier is 81 % (at $V_{IN} = 21\ V$ and $P_O = 11\ W$). This efficiency corresponds to the second graph in Figure 7.13.

The last measurement was the frequency response of the TPA3110D2 amplifier, which is illustrated in Figure 7.16.

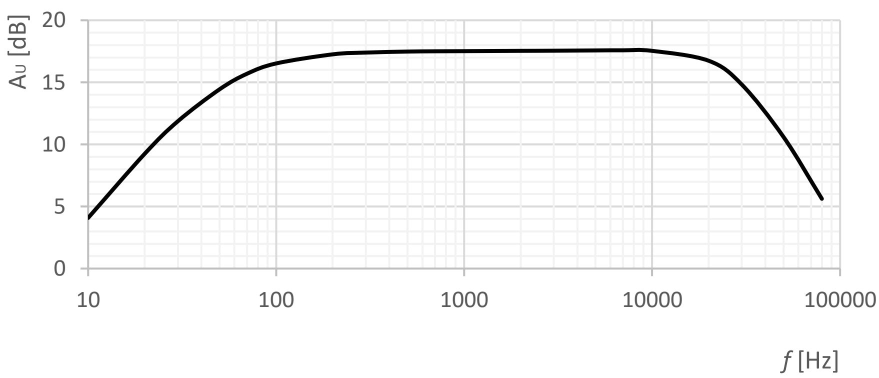

Fig. 7.16 Frequency response of TPA3110D2

The frequency response should be constantly at 20 dB, but in this case, it is at 17.5 dB. This decrease may be due to the chip not being original. The lower limit of the 3 dB drop is at 70 Hz and the upper is about 27 kHz where the output LC low pass filter is applied. The frequency response of this amplifier is not ideal, especially in low frequencies, where deeper bass is just slightly amplified. This chip will be replaced as soon as the original amplifiers from Texas Instruments will be available. Unfortunately, in today's lack of chips, this cannot be done otherwise.

Now more to the already mentioned new blocks, removed/modified old blocks, and the problems that were solved thanks to this final solution.

7.2.4 SHUTDOWN control of power amplifier

As can be seen from Figure 7.14, the audio detector block is missing on the final PCB. This step was caused by the fact that the audio detection was too complicated, sometimes did not work at all, jumping into shutdown mode was delayed, due to the need to set a delay circuit for proper operation of the IC, and a total cost (about 100 CZK for all the components).

Due to all these disadvantages of this analog system, a much more efficient and reliable digital system has been proposed, which has already been mentioned in Subchapter 7.1. Figure 7.6 shows the pin of the ESP32 microcontroller used for this activity. The idea is that it is already clear in the program when the music is playing or is stopped. In other words, when the amplifier should be active and when not. Thus, it was possible to remove all these semi-functional analog systems and only connect the output from pin IO22 on ESP32 to the SHUTDOWN pin on PA and drive the shutdown mode directly from the program part.

In this way, the shutdown mode occurs immediately without any delay and saves energy for the maximum possible time. Another advantage is the elimination of switching interference in the form of "pops" between the transition of individual modes, which occurs directly from the development board, and the elimination of noise in the form of complete silence because the amplifier is essentially switched off.

An integral part is the function of connecting headphones to the multifunction speaker. Without shutdown mode, this would not be possible because the amplifier needs to be turned off depending on whether the headphones are plugged into the speaker. This function is performed by a jack-sensing circuit that simply switches the path between pin IO22 ESP32 and SHUTDOWN pin PA, as shown in Figure 7.17, without any delay and switching noise.

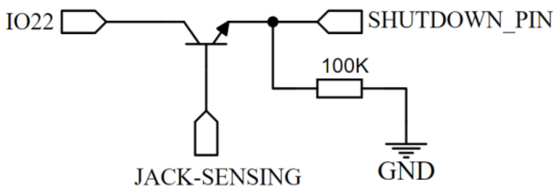

Fig. 7.17 Activation of shutdown mode via Jack-Sensing circuit

7.2.5 Buck converter

From Figure 7.14 can be seen that part of the DC/DC buck converter has not changed much. The only change is the replacement of three 10 μF ceramic capacitors with two larger 100 μF electrolytic capacitors to reduce the ripple effect at the converter output.

There were essentially no major problems with this function block from the beginning, so no major modifications were needed. The only problem was the interference that got from the development board to the amplifier and buck converter. Thus, repeating in the loop. This interference was then analyzed as a ground loop.

7.2.6 Ground loop problem

One of the biggest problems was interference in the form of cyclically repeated knocking in speakers. This interference was most intense, for example, with Bluetooth and Wi-Fi technologies. An example of this interference (captured by the RIGOL DS1054 oscilloscope) is shown in Figure 7.18, where the multifunction speaker is in Bluetooth mode.

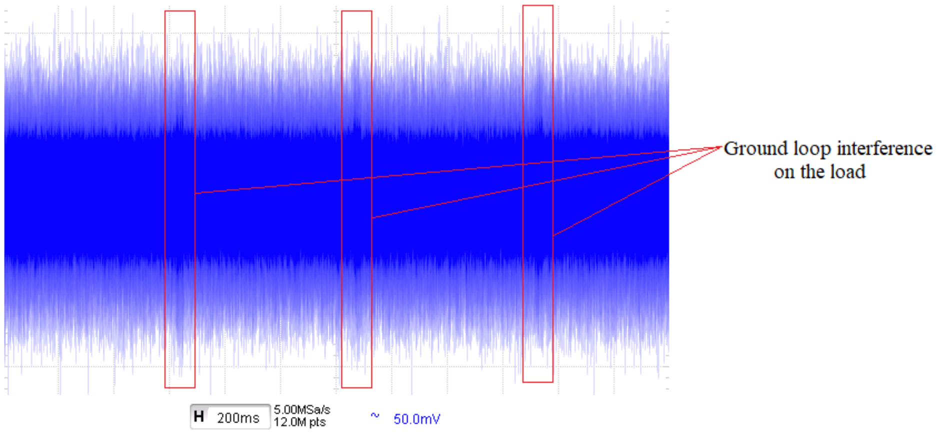

Fig. 7.18 Interference on the load side

After analyzing and measuring this interference, was concluded, that it was a ground loop interference. A ground loop occurs when a circuit with two or more points, that are nominally at ground potential, are connected by a conducting path such that either or both points are not at the same ground potential. [27] To overcome this issue, several solutions have been provided, each with its advantages and disadvantages.

Ground loop isolator - This method is used in audio engineering to eliminate ground loop noise with the possibility of a connection between two audio devices, such as a signal source and an amplifier. It uses two transformers with a 1: 1 ratio to filter out any voltage fluctuations. This solution seemed like a good choice, but it is spacious and expensive. Unfortunately, after trying this method, nothing happened, and the interference continued.

Isolated DC-DC converter - These converters convert the DC voltage back to usually the same DC voltage, with the input being electrically isolated from the output.

This method was tested by connecting this isolation DC/DC converter between the output of the buck converter and the power input of the development board. This solution worked well, and no interference occurred at the output of the amplifier. However, this solution also has its disadvantages in the form of relatively large quiescent current consumption. In this case, the consumption was 35 mA from the output of the 5 V buck converter, so with 175 mW constant power consumption from the power supply. This is not ideal since the multifunction speaker is powered by a battery pack. Another advantage is the unregulated voltage at the output and the price (approx. 200 CZK).

In the end, none of the above methods had to be used and the interferences were removed using the final PCB design. Figure 7.14 shows that the ground is divided between PGND and AGND and connected by interconnecting resistors, vias are placed along the signal paths for the best possible interference suppression, and the balanced input of the amplifier prevents noise. Thus, the amplifier is almost inaudible in no-play time, and no interferences are present at the output.

7.2.7 Battery indicator

The last part that was added to the final PCB design is the battery indicator. This circuit was added because it is a very convenient feature for the user to keep track of the battery status and possibly charge the battery.

The circuit is very simple and consists of only three LM358 operational amplifiers used as comparators, voltage divider, and trimers for setting the reference voltages. The LM358 was chosen because it has a wide power supply range of up to 32 V, which was needed because it uses a voltage from the battery pack of up to 21 V and a sufficient output sink current of up to 20 mA. Furthermore, only the voltage divider is used to reduce the voltage from the battery pack and subsequently the trimmers, which reduce the reference 5 V voltage from the buck converter to set the reference value for comparison. Therefore, whenever the divided voltage of the battery pack falls below the level of one of the reference values set by the trimmer, the operational amplifier applies a supply voltage to its output and thus the LED goes off due to the same voltage on the anode and cathode.

To limit the LED current, 10k ohm resistors have been added and the used voltage divider consists of megaohm resistors as well as a trimmer value which is 1M ohm. In this way, the entire circuit, at full load, takes about 10 mA from the source, which is still an acceptable value of quiescent consumption in exchange for a function of indicating the status of the battery that is very pleasant for the end-user.

7.2.8 Changes on ESP32-LyraT-Mini

Changes had to be made to the ESP32-LyraT-Mini development board to meet all the requirements and functions required. The main problem with this board was that nothing happened after turning on the power, thus the program did not start executing by itself and it was always necessary to press the RESET button located in the lower-left corner of the board to start executing the program. This problem was reported to Espressif technicians, that are unable to answer this problem to this day. However, after seeing the documentation on Boot Mode Selection from Espressif [28], the error was analyzed as a hardware problem right on the board.

Therefore, after connecting the power supply, the GND signal time on the EN and BOOT (IO0) pins is very short due to the charging of the capacitors. In this configuration, the microcontroller is in the download state and waiting for the download. Therefore, nothing starts to work. So, the solution was to use different capacitors for one of the pins, remove the capacitor from GPIO0 or add a 10k ohm resistor between GPIO0 and the power supply. The simplest solution, in this case, was to use a resistor between the power supply and GPIO0 because the pin was close to the power socket, as shown in Figure 7.19.

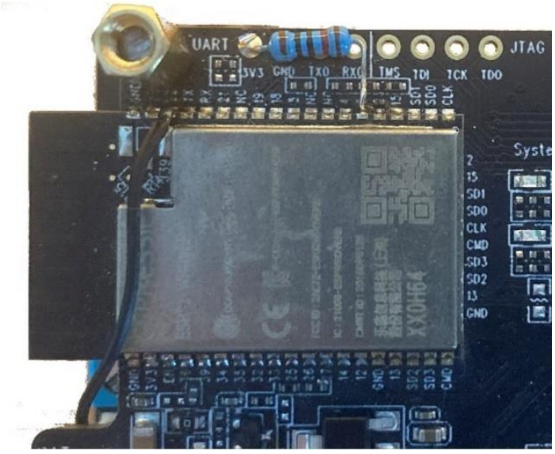

Fig. 7.19 Resistor connected between a supply voltage and GPIO0

Another problem was the need for a balanced audio signal for the amplifier. According to the development board scheme shown in Figure 7.20, only one signal from the codec chip is connected to the jack output, from where the signal from the development board is taken to the amplifier.

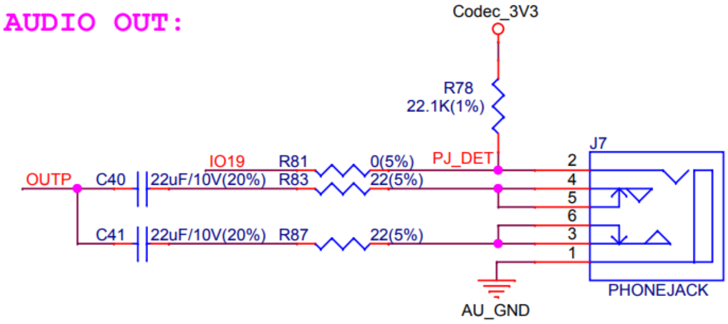

Fig. 7.20 Audio output on ESP32-LyraT-Mini

Therefore, it was necessary to interrupt this path from OUTP to C41 and replace it with a line from the OUTN codec chip output to the C41 capacitor. This jumper is shown in Figure 7.21.

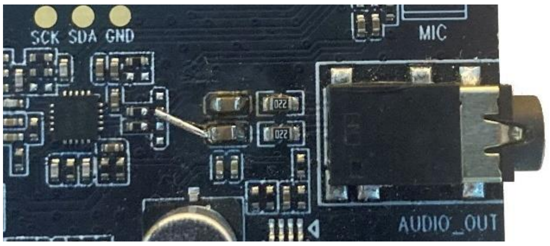

Fig. 7.21 Jumper for connecting the OUTN to C41

7.2.9 Battery pack

Until the hardware was fully assembled, it was not sure exactly how much current the multifunction speaker would draw. Therefore, current consumption tests were performed after assembly. The entire hardware had very similar current consumption across the entire 16 to 21 V supply range of the battery pack. 50 mA in an idle mode without the amplifier shutdown mode enabled (otherwise 30 mA), 60 mA at low volume 100 mA at medium volume, and 250 mA at full volume with up to 500 mA peaks. These values are very dependent on the mode used. In SD mode is the lowest and in BT or WIFI mode, the consumption is higher by about 20 mA due to the use of the radio.

On this basis, Li-Ion 18650 3.7 V/2000 mAh 3 C batteries from MOTOMA were selected. Five of these batteries should keep the multifunction speaker running at the highest volume of approx. 7 hours, at medium volume than 20 hours, and at the lowest volumes up to 30 hours.

These five batteries are then connected in series to a final voltage of 16 to 21 V and protected by a BMS for safe use. The final battery pack can be seen in Figure 7.22.

Charging from a fully discharged battery pack to a fully charged takes about 4 and a half hours (tested).

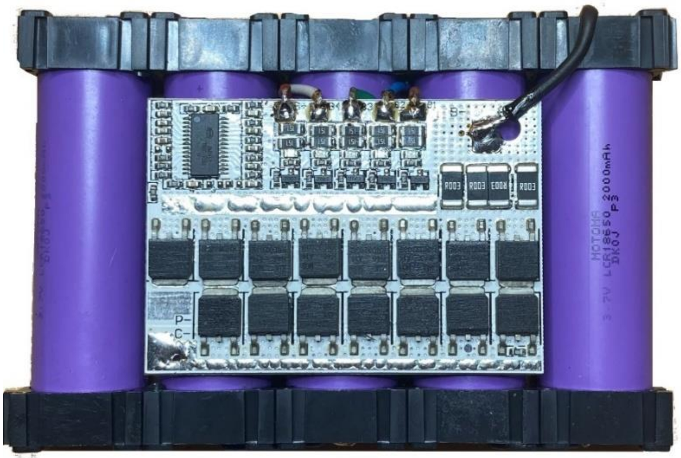

Fig. 7.22 Battery pack consisting of five 18650 Li-Ion 2000 mAh batteries

7.2.10 Battery charging

For safe and smooth charging, a Buck/Boost converter with a function to limit the output current was added to the multifunction speaker, which is a highly sought-after function for charging Li-Ion batteries. This converter module is shown in Figure 7.23.

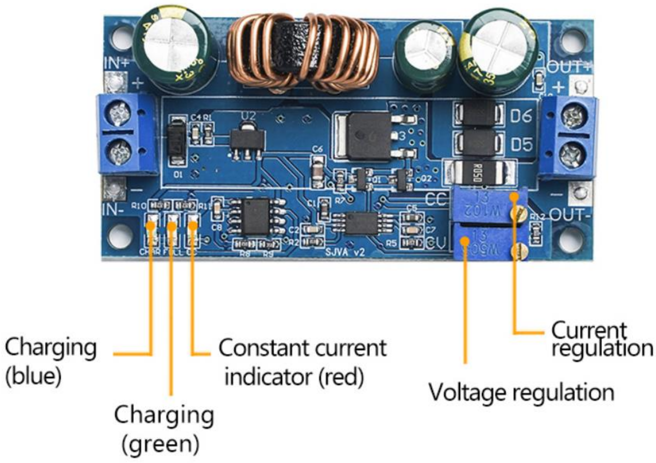

Fig. 7.23 Buck/Boost converter for battery charging [29]

This module is set to a maximum output current of 500 mA and an output voltage of 21 V. Thus, when the batteries are completely discharged, (corresponding to a voltage of 16 V) the first state of charge is a constant current of 500 mA until the batteries are charged to a sufficient voltage level. The current begins to drop and switches to a constant voltage state of 21 V until the battery pack is fully charged. LEDs directly from the module board, as can be seen in Figure 7.23, are used to indicate the charging status, namely blue (indicating charging), and green (indicating the full charge of the battery pack) LED. These LEDs are pulled to the rear panel of a multifunction speaker for a clear indication for the user.

Thanks to the appropriate BMS and Buck/Boost converter control, it is possible to have the speaker still connected to the source of DC power and use it as a mains-powered device. Thanks to this module, it is also possible to charge the multifunction speaker with a large voltage range, from 4.8 to 30 V (the output power of the source must be at least 11 W). The most suitable source of DC voltage for charging is very easily available chargers for laptops, which have an output voltage of about 20 V DC and output currents in the order of amperes. This makes it perfect for charging this device.

7.2.11 Loudspeakers

For the best sound, it is necessary to take up the entire audible bandwidth, i.e., about 20 Hz to 20 kHz. Two 8-ohm TVM ARN-100-10/8 midrange speakers with a frequency range of 100 - 6000 Hz, shown in Figure 7.24, and one 8-ohm TVM ARV-104-61/8 tweeter with a frequency range of 2500 - 18000 Hz, shown in Figure 7.26, were selected for this activity.

Fig. 7.24 TVM ARN-100-10/8 mid-range speaker [30]

These mid-bass speakers reproduce music well from the 100 Hz and basically, from 20 to 100 Hz the frequencies are slightly attenuated, as can be seen in Figure 7.25, but this fact does not matter much, because the program part includes equalizers that balance these levels and SPL is basically from 40 Hz to 10 kHz at the same level. (Tested only sensory.)

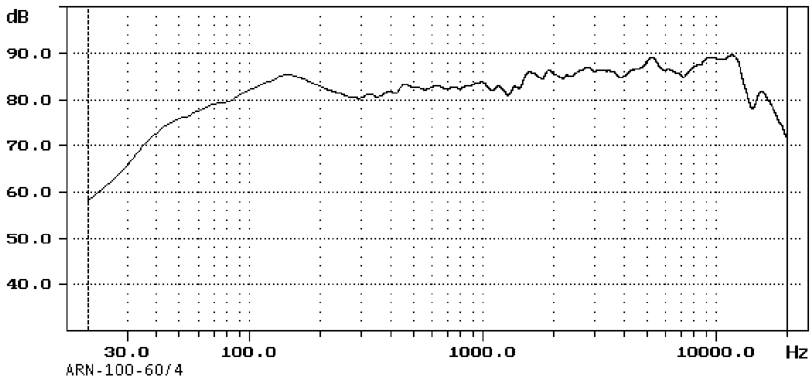

Fig. 7.25 Frequency response of TVM ARN-100-60/8 [30]

The frequency response of the TVM ARV-104-61/8 is shown in Figure 7.27, which meets the requirements for high-quality reproduction of high frequencies.

Fig. 7.26 TVM ARV-104-61/8 tweeter [31]

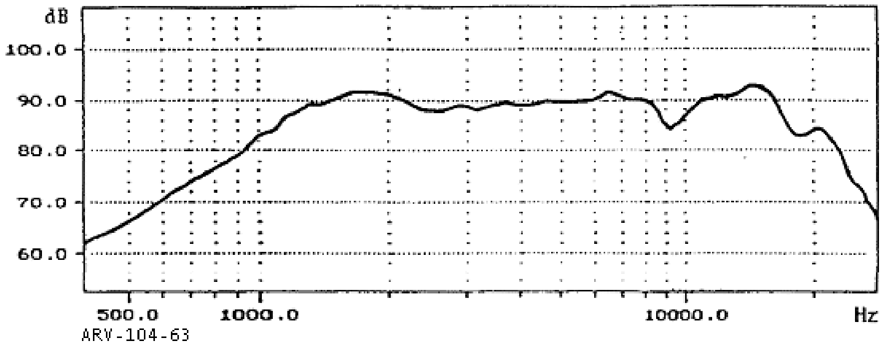

Fig. 7.27 Frequency response of TVM ARV-104-61/8 [31]

The small spike to 85 dB around 9 kHz, which can be seen in Figure 7.27, was compensated by equalization in the program.

The parameters from references [30], [31] are related to the 4-ohm versions of these speakers. However, they are almost the same as the 8-ohm versions used in this work, which unfortunately could not be found.

7.2.12 Crossover

For better separation of power and unnecessary and useful frequencies for each loudspeaker, a crossover is included in the system. This crossover is shown in Figure 7.28, with its equivalent scheme.

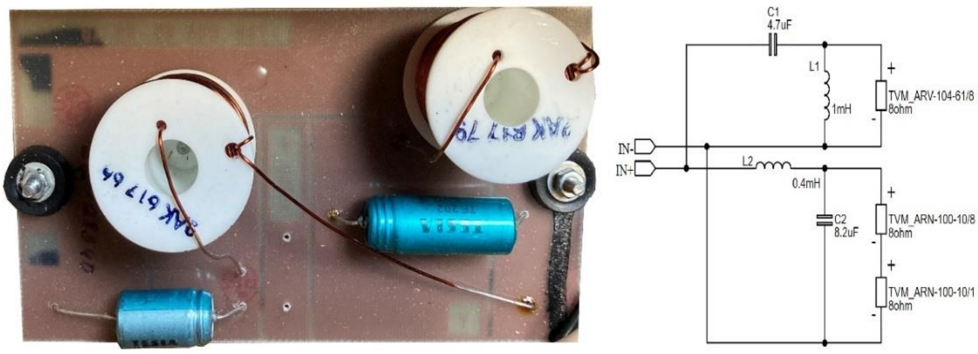

Fig. 7.28 Crossover used in multifunction speaker with scheme

An LC high pass filter with a cut-off frequency of 2.3 kHz. The cutoff frequency for an LC high pass filter is calculated using the following formula 7.1:

$$f_c = \frac{1}{2\pi\sqrt{LC}} = \frac{1}{2\cdot\pi\cdot 20\cdot\sqrt{1\cdot 10^{-3}\cdot 4.7\cdot 10^{-6}}} = 2321\ Hz, \tag{7.1}$$

And for mid-range speakers is used LC low pass filter with the cut off frequency of 2.8 kHz, calculated using the following formula 7.2:

$$f_c = \frac{1}{2\pi\sqrt{LC}} = \frac{1}{2\cdot\pi\cdot 20\cdot\sqrt{0.4\cdot 10^{-3}\cdot 8.2\cdot 10^{-6}}} = 2778\ Hz, \tag{7.2}$$

With this connection in Figure 7.28, the total load has a 16 ohms impedance. A variant with a total impedance of 8 ohms with TVM ARN-100-60/4 mid-range speakers was tested and the feeling of playback was the same as with the 8-ohm variant. Therefore, speakers with 8-ohm impedance were chosen, because the resulting current flowing into them was much smaller and thus ensure longer battery life.

7.3 Appearance design

Another integral part is the appearance design. The aim was focused on a clean and easy design, which is immediately understandable for the user, and it is useless to open the instructions for use at all. Just push a button and play.

7.3.1 Enclosure for multifunction speaker

A particleboard enclosure is used for the safe storage of all hardware parts. This enclosure is recycled from the old enclosure of TVM ARS-204-50/8 with bass reflex. The prepared enclosure is shown in Figure 7.29, from several perspectives.

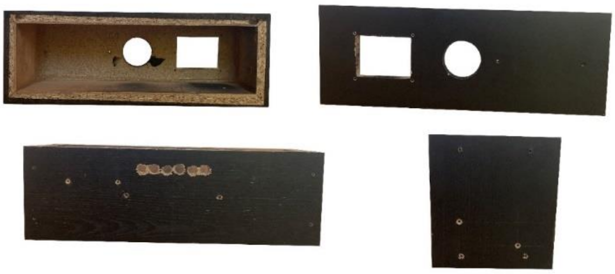

Fig. 7.29 Prepared enclosure for the multifunction speaker

A front cover and holder for the TVM ARV-104-61/8 and TVM ARN-100-10/8 speakers with a protective snap-on net are shown in Figure 7.30.

Fig. 7.30 Front cover

This enclosure was then covered with gray cloth. This cloth was recycled as cut pieces from IKEA curtains. A piece of this gray cloth is shown in Figure 7.31.

Fig. 7.31 Recycled gray cloth from IKEA ANNAKAJSA

7.3.2 Rear panel

The rear panel has been designed for easy access to features like an SD card, headphone output, and power input for charging and switching on/off the multifunction speaker. The 3D model can be seen in Figure 7.32.

Fig. 7.32 3D model of rear panel

The resulting rear panel printed by a 3D printer can be seen in Figure 7.33, which shows all its functions.

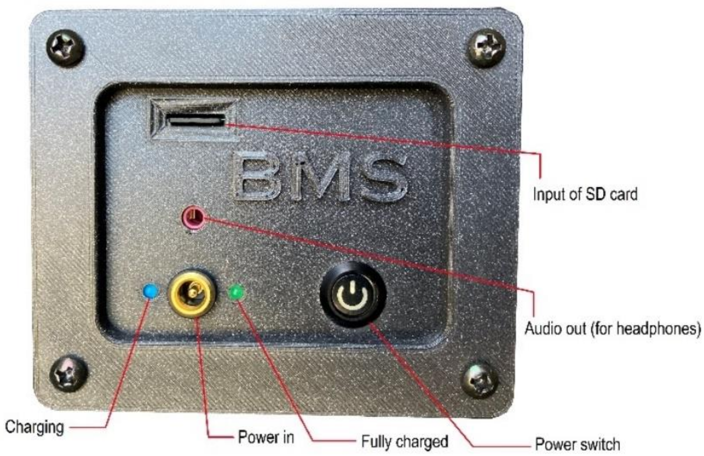

Fig. 7.33 Rear panel

The rear panel is designed for easy use. A cutout has been created for the SD card for better mounting. To indicate on or off status, the power switch has an integrated LED that lights up at startup. For charging status, there are 2 LEDs next to the power input. Subsequently, BMS is written on the free space, which was chosen as the name of this multifunctional speaker and this abbreviation means Baranek's Multifunction Speaker.

7.3.3 Top control panel

Another integral part is the top control panel with buttons and their functions marked with simple symbols such as play/pause, next, previous, etc. The 3D model of this panel is shown in Figure 7.34.

Fig. 7.34 3D model of top control panel

In this panel are buttons that have integrated LEDs. These LEDs are connected to the battery indicator circuit from Subchapter 7.2.7 and show the user an overview of how the battery pack is charged. Subsequently, the buttons perform the classic functions of a multifunction speaker. Functions, together with the percentage representation of the battery status, are shown in Figure 7.35, where is the installed resulting top control panel. These percentage values are set based on the discharge characteristics of the batteries used in the battery pack that is available at [32].

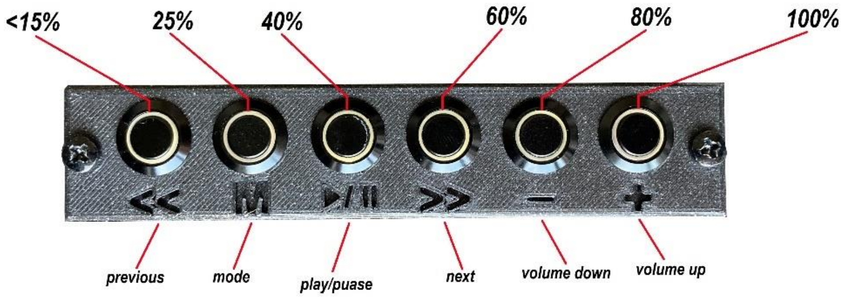

Fig. 7.35 Top control panel with explanations

7.3.4 Protective feet and Holder

To protect the entire multifunction speaker, protective feet have been designed. These feet are shown in Figure 7.36. These feet are suitable for any placement of the multifunction speaker. So, that it can stand both vertically and horizontally.

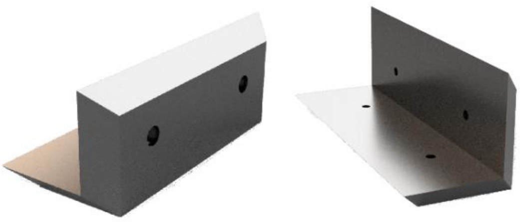

Fig. 7.36 Protective feet

For convenient transmission of the multifunction speaker, a holder is installed on the top of the enclosure, that fits perfectly into the entire design of the multifunction speaker. This holder is shown in Figure 7.37.

Fig. 7.37 Holder

7.3.5 Final result of BMS (Baranek’s Multifunction Speaker)

This subchapter reveals the complete multifunctional speaker of this work, hereinafter referred to as BMS (Baranek's Multifunction Speaker).

Figure 7.38 shows the inner part of the BMS with all the parts already discussed and installed. The advantage is that all parts are replaceable. Thus, for example, replacing the battery pack is not a problem (or anything else). The separability of the functional parts has been maintained for easy future BMS improvements. For better vibration resistance, all parts were installed on rubber or foam pads and glued.

Fig. 7.38 Inside of BMS

The icing on the cake was the addition of a symbol on the front with the speaker's name. The 3D model of this symbol is shown in Figure 7.39.

Fig. 7.39 BMS symbol

Finally, the outer part of the BMS is shown in Figure 7.39.

Fig. 7.40 The BMS

8. COMPARISON

The fully functional multifunction speaker BMS is ready for commercial use. But the question remains, how would this product perform on the market? Therefore, this chapter focus on comparing competing products that are the best-selling in the industry in terms of the product price, parameters, multifunctionality, appearance, etc.

8.1 Parameters of BMS

Let's summarize all the parameters that have already been mentioned in a clear table 8.1 and find out how the BMS is doing.

Table 8.1 Parameters of BMS

Name	Wi-Fi	BT	SD card	Jack output	Radio	Power	Battery life	IP Code	Weight	Price
BMS	✓	✓	✓	✓	Via internet	11 W	20 hrs	IPX0	5.7 kgs	3400 czk

All parameters have been tested in practice and are guaranteed. Rated power is in RMS value and the price is completely calculated and related to only one piece. If the power in RMS is taken at 10 % THD+N, the power is 16 W. The table shows the power when the output was without distortion.

Battery life very much depends on the used mode. The table shows the minimal playtime at medium volume in Wi-Fi radio and Bluetooth mode in hours. In SD card mode, the battery life is even greater. The BMS was tested at the lowest to middle volume, lasting over 30 hours on Wi-Fi radio and Bluetooth mode. On the middle to highest volume level was playtime of more than 20 hours. Another advantage is the clean appearance of the BMS, Figure 8.1.

A detailed overview of the parameters is in table 8.2 for an exact overview of what the product is capable of.

Fig. 8.1 Appearance of the BMS

Table 8.2 Detailed parameters of BMS

Audio Specification	
System type	Active
Output power	11 W
Frequency response	80 Hz ~ 18 kHz
Audio codec	MP3
Connections	
Wireless features	Bluetooth, Wi-Fi
Local playback	SD card
Output	3.5 mm jack
Bluetooth specification	
Bluetooth version	Bluetooth 4.2
Bluetooth profiles	A2DP, AVRCP
Bluetooth frequency	2402 ~ 2480 MHz
Bluetooth transmitter power	-12 ~ 9 dBm
Max connected devices	2
Wi-Fi specification	
Wi-Fi frequency band	2412 ~ 2484 MHz
Standards	IEEE 802.11b/g/n
SD card specification	
Type	microSD card
Supported formats	FAT12/16, FAT32, exFAT
Battery	
Playback time	20 hrs (medium volume)
Charging	DC-022B (5.5X2.1 mm)
Charging time	4.5 hrs for fully charged
Dimensions	
Dimensions	42.3 x 16.3 x 15.7 cm
Weight	5.7 kgs

8.2 Parameters of competition

To compare the parameters, the best-selling speakers were selected in a similar category as the BMS. These speakers are listed in Table 8.3 with their parameters as well as with the BMS parameters.

Table 8.3 Parameters of BMS and best-selling speakers from the e-shop alza.cz

Name	Wi-Fi	BT	SD card	Jack output	Radio	Power	Battery life	IP Code	Weight	Price
BMS	✓	✓	✓	✓	Via internet	11 W	20 hrs	IPX0	5.7 kg	3400 czk
JBL Extreme 3 [33]	✗	✓	✗	✓	✗	50 W	15 hrs	IPX7	1.83 kgs	7 108 czk
Bang & Olufsen Beolit 20 [34]	✗	✓	✗	✓	✗	70 W	8 hrs	✗	2.7 kgs	11 273 czk
Gogen SMILEE BPS 636 [35]	✗	✓	✗	✓	FM	40 W	6.5 hrs	✗	5.2 kgs	2 999 czk

The appearances of the best-selling speakers are in Figure 8.2.

Fig. 8.2 a) JBL Extreme 3 [33], b) Bang & Olufsen Beolit 20 [34], c) Gogen SMILEE BPS 636 [35]

As Table 8.3 shows, in terms of multifunctionality, battery life, and price, BMS is surely a better choice. On the contrary, it stands far behind weight and power. However, the BMS was tested face-to-face with a JBL speaker, JBL Charge 4, which states on the JBL website that this speaker has a power of 30 W RMS. In the tests, the BMS was as loud if not louder than this portable speaker, and it has a measured power of 11 W RMS. The same goes for battery life. The manufacturers do not state at what volume the speaker is used. Therefore, it is difficult to compare these parameters unless it is determined under what conditions the manufacturers state them.

Each of the speakers on the commercial market has its advantages and disadvantages, and the user chooses according to how and where the device will be used. It is probably clear that BMS will not be bought by a user who plans, for example, to go on a hike, where he also takes a portable speaker to play music. However, BMS is a very good alternative to home sound systems, as a portable speaker that can be moved to another location (such as a garden, workshop, living room, bathroom, etc.) or is statically in one place with the possibility of continuous mains supply.

As a result of this comparison, the BMS is certainly able to compete with commercial multifunction speakers. The price of BMS is only for this one piece, and if mass production were considered, the price would decrease rapidly and thus the final selling price.

9. FUTURE NEXT STEPS

This project is not finished and over. Many details need to finish and improve. Therefore, this chapter reveals future steps with this project to bring the BMS to the highest possible level among multifunctional speakers.

9.1 Modifications

The model of this project called BMS as shown in Figure 7.40, is undoubtedly a finished product. However, it is always possible to improve, and therefore the most important future adjustments that this project awaits are listed here.

9.1.1 Mid-range speakers

Among the fixtures that would certainly benefit are the replacement of midrange speakers. The used TVM ARN-100-10/8 mid-range speakers are excellent, and the sound is clear from them but since they have a frequency response from 100 Hz, they lack proper bass reproduction. Therefore, future Visaton KT 100 V mid-range speakers are selected for this purpose, which has a frequency response from 32 Hz up to 9500 Hz with a resonant frequency of 42 Hz. [37] Thanks to these features, bass reproduction could be much better, thus improving the speaker's characteristics. The Visaton KT 100 V speaker is shown in Figure 9.1.

Fig. 9.1 Visaton KT 100 V [37]

9.1.2 Power amplifier

As the results of the measurements in Subchapter 7.2.3, the power amplifier TPA3110D2 is either non-original or could be better designed (most probably non-original). Therefore, as soon as the original TPA3110D2 can be ordered from authorized Texas Instruments dealers, this IC will be replaced with the original.

However, the adjustments do not end here. For even better BMS results, it is planned to test other amplifiers and compare their parameters and choose the best one. A suitable replacement is a variant of a class D power amplifier again from the manufacturer Texas Instruments, namely TPA3128D2. This PA has a Wide Voltage Range: 4.5 V to 26 V, Very Low Idle Current: <23 mA for recommended LC filter configurations, Greater than 90 % Power Efficiency Combined With Low Idle Loss Greatly Eliminates Need for Heat Sink, Integrated Self-Protection Circuits Including Overvoltage, Undervoltage, Overtemperature, DC Detect, Short Circuit With Error Reporting, and especially the TPA3128D2 can be connected in PBTL mode enabling up to 60 W output power [38], which will help increase BMS output power and match the competition in Table 8.3. Of course, the power would not be 60 W but would be limited to a maximally up to 30 W.

9.1.3 Bluetooth remote control

The only major problem with BMS is the synchronized volume control on both devices. This bug is purely in the software part and there is surely a solution to it because the AVRC profile of Bluetooth technology enables this feature. Therefore, the greatest possible emphasis will be placed on solving this problem aiming to avoid the distribution of volume adjustment into two separate systems, so that the volume on both devices can be controlled synchronously.

9.2 Version 2.0

The next step in building this project is version 2.0. This means a completely new solution in terms of enclosure design, appearance, parameters, etc. The next version will focus on smaller dimensions and lightweight for easy mobility, fulfillment of all given functions in the first version without operational problems and adding others for even greater multifunctionality. For example, Home Assistant, Live input, 5 V USB output as a battery bank for mobile phones, etc...

The goal is to remove the development board ESP32-LyraT-Mini from the project and design its own solution that will replace the board's features. This reduces the entire hardware part weight and dimensional properties to the smallest possible.

10. CONCLUSION

The beginning of this work was an introduction to the basics of acoustics, loudspeakers, low-frequency amplifiers, and multifunctional speakers in general with connection methods. The goal is to replicate today's portable multifunction speakers with the addition of radio as an autonomous source of music, which is often lacking in today's commercial devices.

The prerequisite was to ensure that the multifunction speaker was able to perform various functions, such as the ability to receive audio signals via wireless technology such as Bluetooth or Wi-Fi, the ability to play SD card data locally, and battery power.
To meet these requirements, the ESP32-LyraT-Mini development board, the TPA3106D2 power amplifier, the 5S battery pack consisting of 18650 Li-Ion batteries, and other function blocks needed to operate, reduce consumption, and achieve maximum system efficiency were selected.

Using the components from the Espressif Audio development framework, it was possible to design and create the overall form of the software part of this project. The power amplifier was designed using an application diagram from the manufacturer and similarly, other function blocks were designed the same way.

The next part was the optimization and testing of the proposed parts of the project. Fatal errors appeared here, which had to be solved by software modifications, a completely new design of the power amplifier, and the removal of some functional blocks. It also included the addition of new function blocks, which are necessary for users such as battery indicator etc...

The result was a finished inner part of the multifunction speaker with the required parameters. After consideration of power consumption were chosen 2000 mAh 18650 Li-Ion batteries for the 5S battery pack. For the outer part, the old chipboard enclosure was recycled, along with the crossovers and loudspeakers for faithful reproduction of audio signals. The entire exterior of the multifunction speaker has been suitably designed. The resulting multifunction speaker is called BMS (Baranek's Multifunction Speaker). An example of BMS functionality can be found here: https://www.youtube.com/watch?v=J5AC4qzRAgA

The last steps were comparison with commercially sold competitors, specifically with the best-selling multifunction speakers, and a look at the future for this project. The results of the comparison are very good and BMS is certainly able to compete with commercial multifunction speakers. The other plans are exciting to improve this multifunctional speaker to the highest possible level.

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SYMBOLS AND ABBREVIATIONS

Abbreviations:

AC	Alternating Current
ADC	Analog to Digital Converter
AP	Access Point
API	Application Programming Interface
AVRCP	Audio Video Remote Control Profile
A2DP	Advanced Audio Distribution Profile
BJT	Bipolar Junction Transistor
BMS	Baranek’s Multifunction speaker
BMS	Battery Management System
BUT	Brno University of Technology
DAC	Digital to Analog Converter
DC	Direct Current
EMI	Electromagnetic Interference
EMC	Electromagnetic Compatibility
FEEC	Faculty of Electrical Engineering and Communications
FSM	Final State Machine
GND	Ground
Hi-Fi	High Fidelity
HTTP	Hypertext Transfer Protocol
I2C	Inter-Integrated Circuit
I2S	Inter-IC Sound
IC	Integrated Circuit
IEEE	Institute of Electrical and Electronics Engineers
IRAM	Internal Random-Access Memory
LAN	Local Area Network
LED	Light Emitting Diode
LF	Low-Frequency
MOSFET	Metal-oxide-semiconductor Field-effect-transistor
PA	Power Amplifier
PCB	Printed Circuit Board
PSRAM	Pseudo Static Random-Access Memory
PWM	Pulse Width Modulation
RF	Radio Frequency
RFI	Radio Frequency Interference
RMS	Root Mean Square
SD	Secure Digital
SDA	Secure Digital Association

SDHC	Secure Digital High Capacity
SDUC	Secure Digital Ultra Capacity
SDXC	Secure Digital eXtended Capacity
SNR	Signal to Noise Ratio
SPL	Sound Pressure Level
SSID	Service Set Identifier
THD	Total Harmonic Distortion
THD+N	Total Harmonic Distortion + Noise
TTL	Transistor-Transistor Logic
UHS	Ultra High Speed
VCC	Voltage Common Collector
WECA	Wireless Ethernet Compatibility Alliance
WEP	Wired Equivalent Privacy
Wi-Fi	Wireless Fidelity

Symbols:

v	velocity	*(m/s)*
t	temperature	$(^{o}\,C)$
p	pressure	(Pa)
ρ	density	(kg/m^3)
λ	lambda	(m)
f	frequency	*(Hz)*
P	power	(W)
B	bandwidth	*(Hz)*
THD	total harmonic distortion	*(%)*
V	voltage	*(V)*
Z	impedance	(Ω)
R	resistance	(Ω)
X	reactance	(Ω)
$THD+N$	total harmonic distortion + noise	*(%)*
q	electric charge	(mA)

LIST OF APPENDICES

Appendix A - Measured values

A.1 Measured of frequency response of TPA3106D2

f [Hz]	U_1 [V_{RMS}]	U_2 [V_{RMS}]	A_U [dB]
10	0,2	1,776	18,96826
20	0,2	1,95	19,78009
30	0,2	1,966	19,85107
50	0,2	2	20
70	0,2	2	20
100	0,2	2	20
200	0,2	2,02	20,08643
300	0,2	2,02	20,08643
500	0,2	2,02	20,08643
700	0,2	2,02	20,08643
1000	0,2	2,02	20,08643
2000	0,2	2	20
3000	0,2	1,998	19,99131
5000	0,2	1,97	19,86872
7000	0,2	1,93	19,69055
10000	0,2	1,856	19,35096
20000	0,2	1,492	17,45478
30000	0,2	1,082	14,66395
50000	0,2	0,564	9,004982
80000	0,2	0,264	2,411479

A.2 Measured of frequency response of TPA3110D2

f [Hz]	U_1 [V_{RMS}]	U_2 [V_{RMS}]	A_U [dB]
10	0,16	0,32	4,0824
20	0,29	0,58	9,24796
30	0,392	0,784	11,86572
50	0,527	1,054	14,43621
70	0,609	1,218	15,69235
100	0,67	1,34	16,5215
200	0,729	1,458	17,25455
300	0,741	1,482	17,39636
500	0,748	1,496	17,47803
700	0,75	1,5	17,50123
1000	0,751	1,502	17,5128
2000	0,753	1,506	17,5359
3000	0,754	1,508	17,54743
5000	0,756	1,512	17,57044
7000	0,757	1,514	17,58192
10000	0,753	1,506	17,5359
20000	0,687	1,374	16,73913
30000	0,551	1,102	14,82303
50000	0,339	0,678	10,60399
80000	0,191	0,382	5,620667

A.3 Measurement of maximum output power for limitation and efficiency measurement of TPA3106D2

Measurement of maximum output power for limitation			Efficiency measurement		
At THD+N = 2,25 % (without visible distortion), V_{BAT} = 21 V					
U_{IN} [V_{RMS}]	**U_{OUT} [V_{RMS}]**	**P_{OUT} [W_{RMS}]**	**U_{OUT} [V_{RMS}]**	**LOAD [Ω]**	**η [%]**
1,065	5,31	14,0981	10,62	8	86,624
At THD+N = 10 %, V_{BAT} = 21 V					
U_{IN} [V_{RMS}]	**U_{OUT} [V_{RMS}]**	**P_{OUT} [W_{RMS}]**			
1,168	5,77	16,6465			

A.4 Measurement of maximum output power for limitation and efficiency measurement of TPA3110D2

Measurement of maximum output power for limitation			Efficiency measurement		
At THD+N = 1 % (without visible distortion), V_{BAT} = 21 V					
U_{IN} [V_{RMS}]	**U_{OUT} [V_{RMS}]**	**P_{OUT} [W_{RMS}]**	**U_{OUT} [V_{RMS}]**	**LOAD [Ω]**	**η [%]**
1,23	4,71	11,0921	13	8	81,2563
At THD+N = 10 %, V_{BAT} = 21 V					
U_{IN} [V_{RMS}]	**U_{OUT} [V_{RMS}]**	**P_{OUT} [W_{RMS}]**			
1,67	5,68	16,1312			
At THD+N = 1,67 % (without visible distortion), V_{BAT} = 16 V					
U_{IN} [V_{RMS}]	**U_{OUT} [V_{RMS}]**	**P_{OUT} [W_{RMS}]**			
0,92	3,35	5,61125			
At THD+N = 10 %, V_{BAT} = 16 V					
U_{IN} [V_{RMS}]	**U_{OUT} [V_{RMS}]**	**P_{OUT} [W_{RMS}]**			
1,24	3,98	7,9202			

Appendix B - Price values

B.1 Prices of individual parts and end price of the BMS

Quantity	Part name	Description	Price [czk]
1	ESP32LyraTMini	Audio Integrated Circuit Development Tools Audio Integrated Circuit Development Tools A lightweight audio development board based on ESP32 WROVERB, which implements AEC, AGC, NS WWE (wake word engine) and other audio signal processing technologies.	488,11
1	TPA3110D2PWPR	Audio Amplifiers AC 15W Fltr-Free Class D Stereo Amp	57,25
1	BD9E104FJE2	Switching Voltage Regulators Switching Voltage Regulators 726V; 1A MOSFET S. Sync Buck DC / DC	22,4
3	LM358DR	Operational Amplifiers Operational Amplifiers Dual Op Amp	21,87
2	BC817-16LT1G	Bipolar transistors, BJT Bipolar transistors, BJT 500mA 50V NPN	8,32
1	UJ2-MIBH-G-SMT-TR	USB connectors USB connectors Micro B, USB 2.0, 1.8 A, Right Angle, Surface Mount (SMT), USB Receptacle	7,78

2	TB002-500-02BE	Fixed Terminal Blocks Fixed Terminal Blocks 2 24 Poles, Screw Type, Horizontal, 5.0 Pitch, 22 12 (AWG), Terminal Block Connector	10,6
2	SJ1-3515-SMT-TR-PI	Phone Connectors Phone Connectors 3.5 mm, Stereo, Right Angle, Surface Mount (SMT), 3 Conductors, 0 2 Internal Switches, Audio Jack Connector	48,2
7	JST PH 2.0 2 Pin	Clamp Plastic housing Male Male Pin head connector JST 2.0 2mm Socket Shell plug	3,7499
2	VE-221M0JTR-0605	Aluminum Electrolytic Capacitors - SMD 6.3 Volts 220uF 20% 6.3x5.3	2,7
2	EEU-FM1V221	Aluminum Electrolytic Capacitors - Radial Leaded Aluminum Electrolytic Capacitors - Radial Leaded 220UF 35V ELECT FM RADIAL	27,1
12	0603ZC105KAT2A	Multilayer Ceramic Capacitors MLCC - SMD, SMT Multilayer Ceramic Capacitors MLCC - SMD, SMT 10V 1uF X7R 0603 10%	39,84
3	12105G106ZAT4A	Multilayer Ceramic Capacitors MLCC - SMD, SMT Multilayer Ceramic Capacitors MLCC - SMD, SMT 50V 10uF Y5V 1210 -20%, + 80%	55,68

2	VES-101M1CTR-0605	Aluminum Electrolytic Capacitors - SMD 16 Volts 100uF 20% 6.3x5.3	3
2	06035A102FAT2A	Multilayer Ceramic Capacitors MLCC - SMD, SMT Multilayer Ceramic Capacitors MLCC - SMD, SMT 50V 1000pF C0G 0603 1%	14,08
2	C0603C104K5RAC3121	Multilayer Ceramic Capacitors MLCC - SMD, SMT Multilayer Ceramic Capacitors MLCC - SMD, SMT 50V 0.1uF X7R 0603 10% AEC-Q200	4,14
1	C0603C120K5HACTU	Multilayer Ceramic Capacitors MLCC - SMD, SMT Multilayer Ceramic Capacitors MLCC - SMD, SMT 50V 12pF X8R 0603 10%	0,536
1	C0603C391K5HACAUTO	Multilayer Ceramic Capacitors MLCC - SMD, SMT Multilayer Ceramic Capacitors MLCC - SMD, SMT 50V 390pF X8R 0603 10% AEC-Q200	1,96
2	GRM155R61H474ME11J	Multilayer ceramic capacitors MLCC - for surface mounting SMD, SMT	1,5
9	CR0603-FX-1002ELF	Thick Film Resistors - SMD Thick Film Resistors - SMD 10K 1% 1 / 10W	1,377
1	MR06X1004FTL	Thick Film Resistors - SMD Thick Film Resistors - SMD 0603 1M00 1% Lead Free	0,612

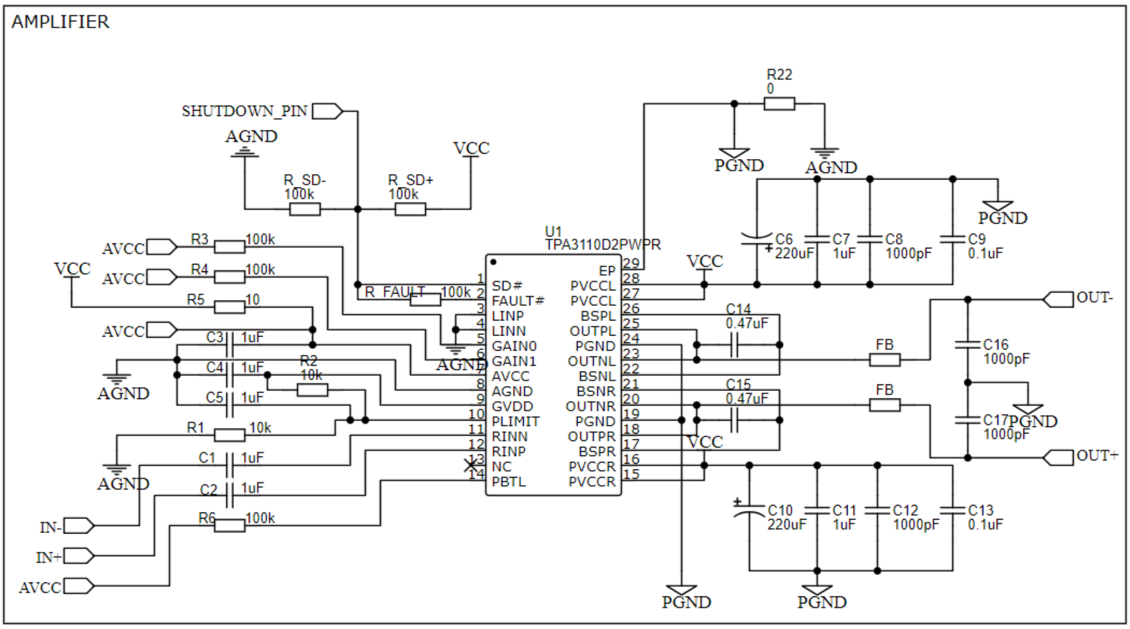
1	ERJ-PA3J155V	Thick Film Resistors - SMD Thick Film Resistors - SMD 0603 1.5Mohm 5% Anti Surge AEC-Q200	2,32
2	AC0603FR-0782KL	Thick Film Resistors - SMD Thick Film Resistors - SMD 82 kOhms 100 mW 0603 1% AEC-Q200	0,408
4	ERJ-3EKF1003V	Thick Film Resistors - SMD Thick Film Resistors - SMD 0603 100Kohms 1% AEC-Q200	1,836
1	CR0603-FX-4303ELF	Thick Film Resistors - SMD Thick Film Resistors - SMD 430K ohm 1%	0,76
1	CR0603-FX-4992ELF	Thick Film Resistors - SMD 49.9K ohm 1%	2,38
5	EVM3ESX50B16 1M	Trimmer resistor Potentiometer Trimpot SMD 3X3 Adjustable Variable resistor	10,2175
2	MPZ2012S101AT000	Ferrite Beads 100ohms 4A 20mOhms 0805 Ferrite Chip	5,7
1	SRP7050AA-6R8M	Fixed Inductors Fixed Inductors Ind., 7.3x6.6x4.8mm, 6.8uH +/-20%, 9A, Shd, SMD	31,44
1	Black Push Button Switch 12/16/19/22mm Waterproof illuminated Led Light Metal Flat Momentary Switches with power mark 5V 12V 24V	white, 16mm, 9-30V (12V), fixed self-locking	44,1
6	Black Push Button Switch 12/16/19/22mm Waterproof illuminated Led Light Metal Flat Momentary Switches with power mark 5V 12V 24V	White Circle, 12mm, 9-30V (12V), Momentary self-reset	162,06

10	18650 Lithium Cell Cylindrical Battery Case Holder Batteries Pack Bracket		11,085
1	Cabinet Handle Soft Cow leather Leather Dresser	Cabinet Handle Soft Cow leather Leather Dresser	76,32
1	Step Up Down Buck Boost Converter	CC CV Adjustable 3A 35W DC 5 -30V to DC 0.5 -30V Step Up Down Buck Boost Converter Power Supply Module Voltage Regulator	40,16
1	golden DC-022B 5.5x2.1mm 5.5 X 2.1 mm Female DC Power adapter dc jack connector DC022B DC power plug male 5.5*2.1mm	golden female	5,057
1	Bms 12v 16.8v 21v 3.7v 100a Li-ion Lmo Ternary Lithium Battery Protection Circuit Board Li-polymer Balance Charging	5S 21V 100A	58,24
1	Light Emitting Diode	3mm blue	0,23
1	Light Emitting Diode	3mm green	0,23
5	LCR18650-2000MAH	Rechargeable Li-Ion 18650 3.7V / 2000mAh 3C MOTOMA	395
2	TVM ARN-100-10/8	Subwoofer with shielded magnetic circuit for applications in small speaker systems where operation near TVs or monitors is expected.	998

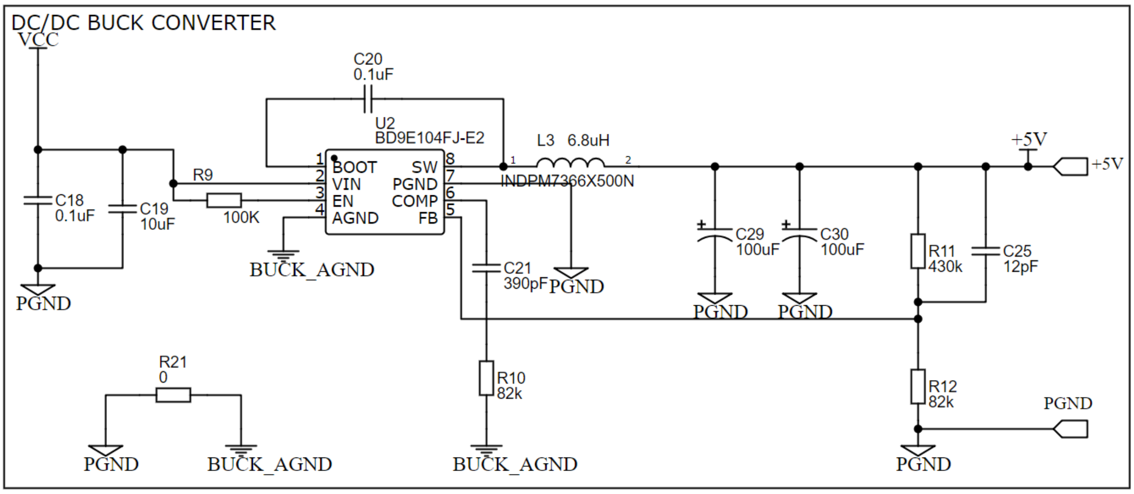
1	TVM ARV-104-61/8	Tweeter with textile hemispherical diaphragm with shielded magnetic circuit for application in TVs and speaker systems that are expected to operate near TVs or monitors.	499
4	Hex Female to Female M2 M2.5 M3 M4 M5 brass standoff spacer Hexagonal Stud Spacer Hollow Pillars	M3XL, 10mm	8,136
2	Hex Female to Female M2 M2.5 M3 M4 M5 brass standoff spacer Hexagonal Stud Spacer Hollow Pillars	M3XL, 40mm	9,553
6	Hex Female to Female M2 M2.5 M3 M4 M5 brass standoff spacer Hexagonal Stud Spacer Hollow Pillars	M3XL, 20mm	19,824
1	Crossover	approx.	80
1	Material for enclosure	chipboard, approx.	50
X	cabels, connectorss, 3D prints, cloth	approx.	50
Overall price (VAT included):			**3382,8614**

Appendix C - Schemes

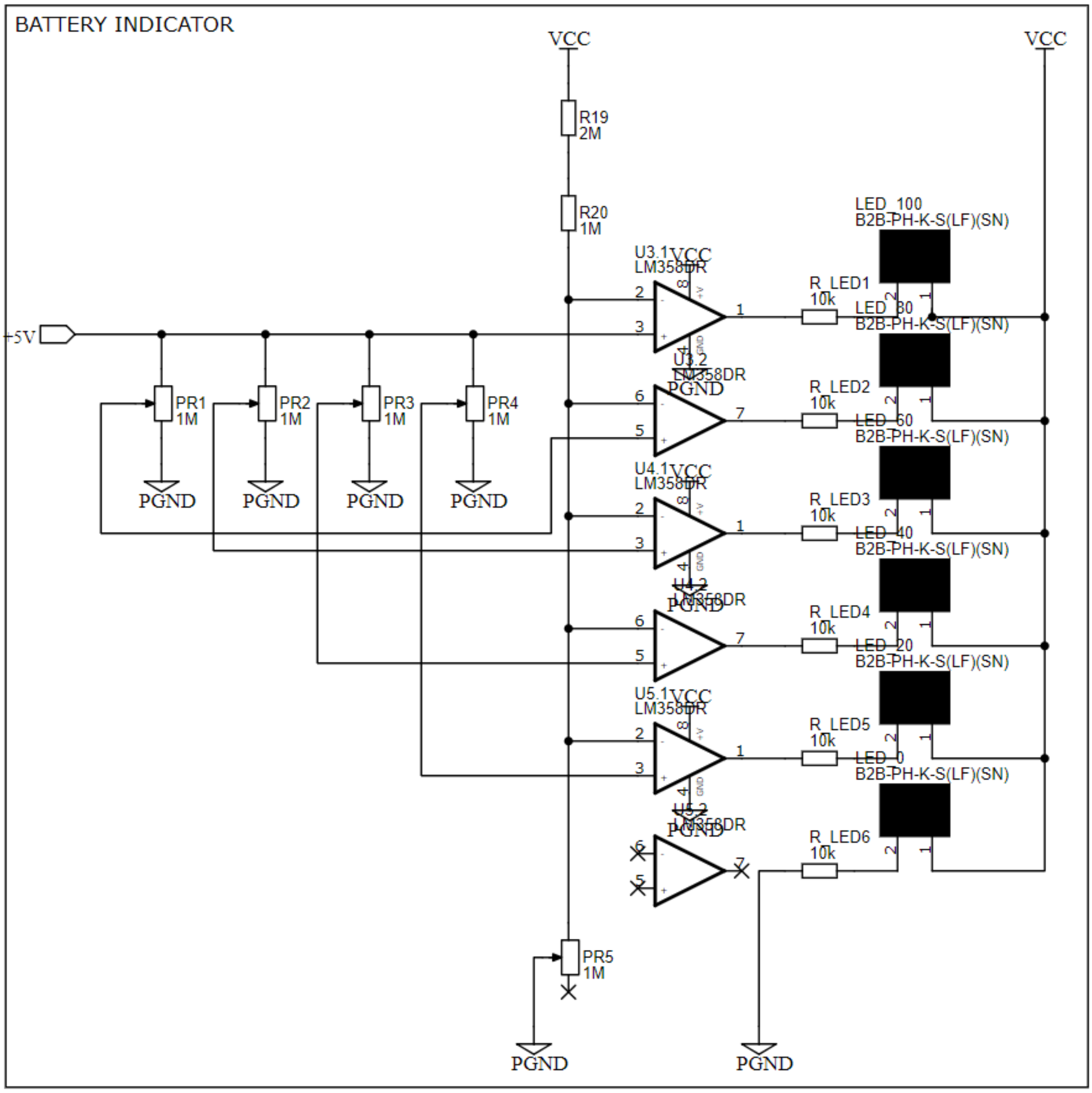
C.1 Power amplifier TPA3110D2



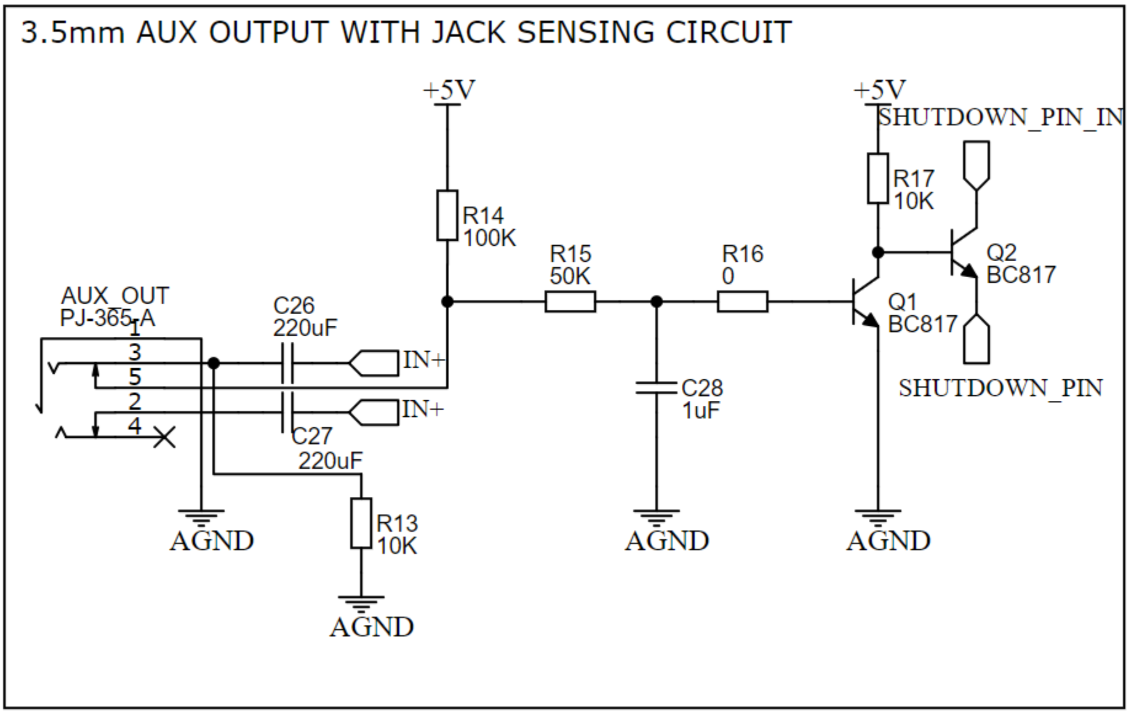
C.2 DC/DC buck converter



C.3 Battery indicator



C.4 Jack-Sensing circuit



C.5 Connectors